the record groove under actual playing conditions. His equipment consists of a modified "Trutrak" parallel tracking device. This has an arm mounted on a float in an oil bath. The arm is located longitudinally by means of magnetic repulsion. The frictional drag on the stylus moves the arm forward against the magnetic repulsion, and the value can be read off a scale. The system is easily calibrated by means of weights. The friction varies very little with speed and weight, and mean values on flat surfaces for the various styli are given in Table III.

It will be noted that the coefficient of friction increases with increasing ellipticity of the stylus. Over most of the range of loads of interest, styli work in the fully plastic range of the vinyl. The contact pressure is about 3 times the yield stress and is almost constant with load. Under a given load, the area of contact and the contact pressure will be similar for each indenter. The differences in the coefficient of friction are therefore a function of geometry. A stylus with a very small longitudinal radius ploughs out a groove of larger cross section than a spherical stylus, thus giving a higher coefficient of friction.

5. PARAMETERS FOR CALCULATION

It was considered by Hunt [15] that an acceleration of 1000g was a reasonable maximum requirement for tracking. This is for a lateral signal, corresponding to 707g for each groove wall. This gives a minimum trace radius at 10 kHz at the outer edge of the record of 36 μm.

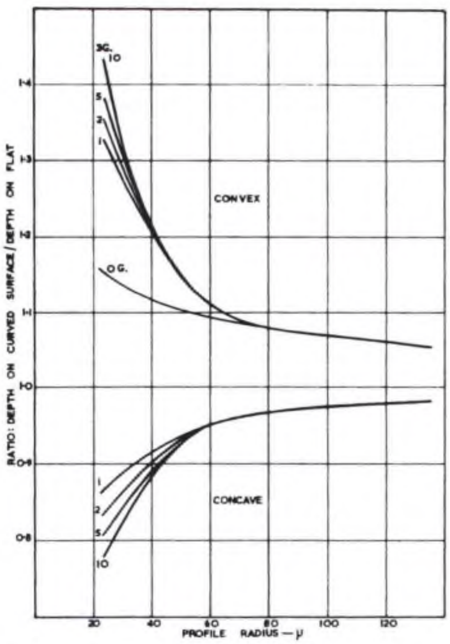


Fig. 5. Depth ratio versus profile radius, vinyl 12.7-μm radius stylus.

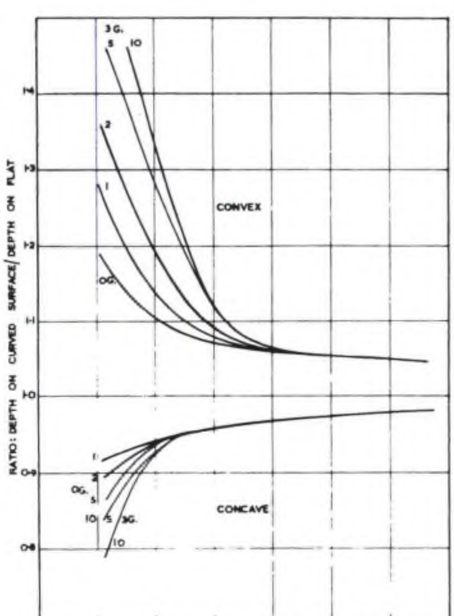


Fig. 6. Depth ratio versus profile radius, vinyl 18/9-μm radius stylus.

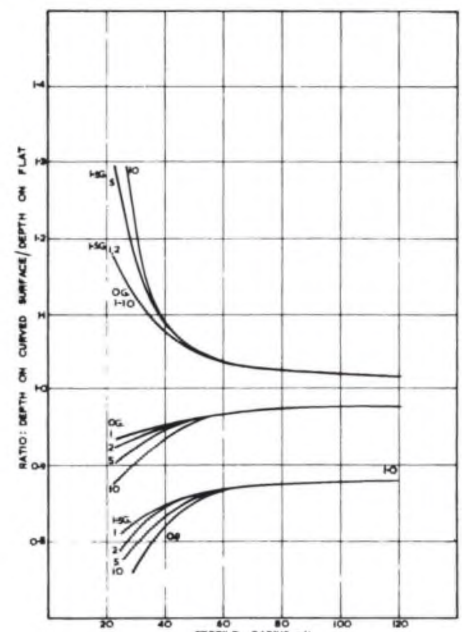


Fig. 7. Depth ratio versus profile radius, vinyl 20/5-μm radius stylus.

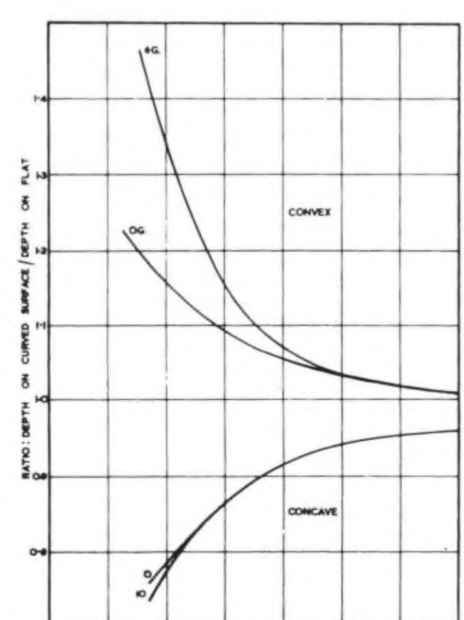


Fig. 8. Depth ratio versus profile radius, nylon 18-μm radius stylus.

Curvature overload occurs when the trace radius is as small as the stylus radius, but there may be serious damage to the groove before this point is reached. In slow-speed tests, digging in with the formation of a chip occurs at a trace radius of 18–25 μm. At the inner edge of the recorded area, the corresponding minimum trace radius would be less than 9 μm, so that the maximum playable acceleration will be much less than 707 g, except perhaps with the 20/5-μm stylus.

The worst case for distortion would be with the signal on one wall only. With equal signal on both walls in phase, the lateral case, the system is in push–pull, so that second-harmonic tracing distortion is canceled. In the vertical case the signals are out of phase. It is convenient

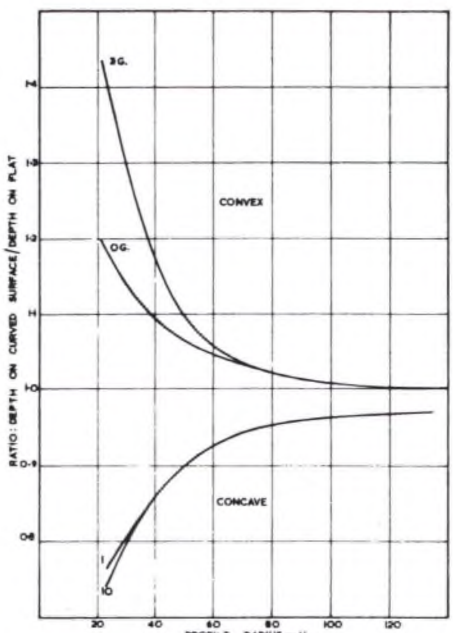


Fig. 9. Depth ratio versus profile radius, nylon 12.7-μm radius stylus.

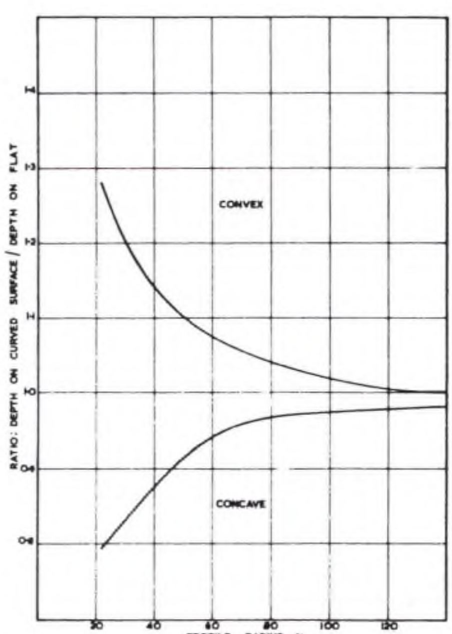


Fig. 10. Depth ratio versus profile radius, nylon 18/9-μm radius stylus.

to compare results by taking similar nominal accelerations. Thus by taking values of amplitude proportional to 1/(frequency)^2 (equivalent to a frequency response of -6 dB per octave), the maximum nominal acceleration and minimum trace radius are the same at all frequencies. Furthermore, the tracing distortion is the same at all frequencies. The velocity of the cut varies as 1/frequency, so that the resistive component of the load will be higher at the lower frequencies.

In comparing pickups, the tip mass in each case is 1/3000 of the playing weight, such that one third of the playing weight would be needed to track an acceleration of 1000 g. The compliance in each case is such as to need one third of the playing weight to track the maximum amplitude of 0.005 cm (at low frequencies). Resistance values were taken proportional to playing weight and give Q of 1.25–7.0 at the top resonance of the tip mass with groove compliance. Values of parameters are given in Table III.

6. TOP RESONANCE

The compliance of the groove at a given playing weight is given by the tangent modulus or slope of the load–penetration curve, and values have been used to calculate the top resonances (Table IV). Some of the resonances are lower in frequency than might be expected and decrease with repeated playing. A check was made on a gliding tone record, DGG 1099111, using a Shure M55E pickup with 0.0007/0.0002-inch stylus at 3-gram playing weight. The top resonance was at 15 kHz and altered very little with repeated playing. It has been argued previously [16] that although considerable plastic deformation takes place, the top resonance may be governed by elastic properties. On loading, the material follows the load–penetration curve. At any point it has deformed and/or work-hardened to support that load. On unloading it is elastic up to the particular load and springs back according to the elastic

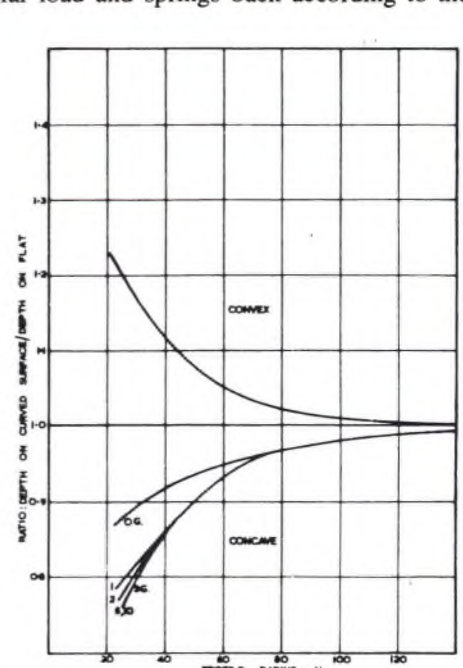


Fig. 11. Depth ratio versus profile radius, nylon 20/5-μm radius stylus.

Table III. Parameters for calculations.

Frequency [kHz]	10		7.07		5		3.54	
Record velocity [cm/s]	50	25	50	25	50	25	50	25
Amplitude [μm]	1.75 0.88 0.44 —	— 0.88 0.44 0.22	3.51 1.75 0.88 —	— 1.75 0.88 0.44	7.01 3.51 1.75 —	— 3.51 1.75 0.88	14.03 7.01 3.51 —	— 7.01 3.51 1.75
Velocity of cut [cm/s]	11.0 5.5 2.8 —	— 5.5 2.8 1.4	15.6 7.8 3.9 —	— 7.8 3.9 2.0	22.0 11.0 5.5 —	— 11.0 5.5 2.8	31.2 15.6 7.8 —	— 15.6 7.8 3.9
Acceleration g [cm/s^2]	707 354 177 —	— 354 177 88.5	707 354 177 —	— 354 177 88.5	707 354 177 —	— 354 177 88.5	707 354 177 —	— 354 177 88.5
Minimum trace radius [μm]	36 72 144 —	— 18 36 72	36 72 144 —	— 18 36 72	36 72 144 —	— 18 36 72	36 72 144 —	— 18 36 72
Stylus radius [μm]	18		12.7		18/9		20/5	
2nd-harmonic tracing distortion [%] (all frequencies)	25 12.5 6.3 —	— 50 25 12.5	17.7 8.8 4.4 —	— 35.4 17.7 8.8	12.5 6.3 3.1 —	— 25 12.5 6.3	6.9 3.5 1.7 —	— 13.9 6.9 3.5
3rd-harmonic tracing distortion [%] (all frequencies)	9.4 2.4 0.6 —	— 37.5 9.4 2.4	4.7 1.2 0.3 —	— 18.8 4.7 1.2	2.4 0.6 0.2 —	— 9.4 2.4 0.6	0.7 0.2 0.04 —	— 2.9 0.7 0.2
Coefficient of friction								
Vinyl	0.24		0.24		0.31		0.36	
Nylon	0.16		0.16		0.17		0.21	
Pickups	Playing Weight [gram]		Compliance [cm/dyn]		Effective Tip Mass [mg]		Resistance [mechanical ohm]	
	3.0 1.5 0.75		5×10^{-6} 10×10^{-6} 20×10^{-6}		1.0 0.5 0.25		44 22 11	
Number of playings 1, 2, 5, 10								

condition, and does not follow the loading curve. Using the elastic condition, good agreement is obtained with actual values of top resonance with spherical styli on certain pickups where tip mass is accurately known, but agreement with biradial styli is less satisfactory. It is not clear on some pickups just where the top resonance is, as there may be other peaks in some designs due to the presence of other compliances. Also there may be an electrical peak and cutoff. The inductance in many designs is typically 0.5 H, and the self-capacity of the cable may be 120 pF per channel. This gives a low-pass filter cutting off above 20 kHz. Higher shunt or cable capacities are recommended by the manufacturers in some cases. Further work is needed to determine the exact nature of the groove compliance. The tangent modulus is used in the present calculations.

7. CALCULATION OF DISTORTION

The method of calculation is iterative, starting with the tracing distortion curve and successively producing modified deformed curves until convergence to the true deformation has been achieved. A flow diagram of the process is shown in Fig. 12, and the action of the boxes is amplified below.

Box 1. The tracing distortion curve is obtained purely from the geometry of the system in the usual way (assuming no deformation) and is calculated at 1° intervals from 0° to 360°, based on an original cosine curve, this being easier to handle than a sine curve.

Box 2. Only the first three harmonics have been considered, since

1) higher harmonics will be small and most will be well above the audio range and the working range of the pickup,

2) very high accuracy of results would be necessary to detect small amounts of higher harmonics—the amplitude of each harmonic has to be multiplied by its own number, as the pickup has a velocity characteristic

3) the higher harmonics are likely to make convergence more difficult. The Fourier coefficients were determined using a standard algorithm [17].

Box 3. In order to ensure convergence of the process, the difference between the newly calculated and the previous Fourier coefficients for the kth harmonic, expressed as a complex number, is modified by a factor $(1 - mC\omega^2) + jCr\omega$, where C is an average value for the record compliance, m the effective mass at the stylus tip, r the mechanical resistance of the pickup, and $\omega = 2\pi kf$, where k (= 1, 2, 3) is the harmonic number and f the frequency.[1]

Box 4. Convergence is achieved when the percentage distortions from the original cosine curve of the two latest deformed curves differ by less than 1% in the first two harmonics and by less than ½% in the third harmonic.

Box 5. Ordinates for the new deformed curve are calculated at 1° intervals by first obtaining a value for the dynamic load, which is a function of the static load, acceleration, stiffness, and resistance and then calculating the deformation from the tracing distortion curve, which is a function of load, friction, curvature, and slope.

This has been written as a Fortran program and run on the ICL 1904S* machine at Bradford University. Results are plotted in Fig. 13 for vinyl and Fig. 14 for nylon. The theoretical tracing distortion is indicated by means of horizontal bars. The change in level of fundamental from the recorded amplitude is expressed in decibels. The second and third harmonics are expressed as a percentage, based on the playback level of the fundamental, rather than the recorded level. The profile of a typical groove under load is shown in Fig. 15.

Table IV. Calculated top resonances of pickups [kHz].

	Stylus Radius			
	18 μm	12.7 μm	18/9 μm	20/5 μm
Vinyl 3.0 grams				
1 playing	20.0	14.3	23.1	14.9
2 playings	17.7	12.1	20.1	12.3
5 playings	15.3	10.2	14.8	10.5
10 playings	13.5	8.7	13.0	8.9
Vinyl 1.5 grams				
1 playing	33.3	23.8	33.5	23.0
2 playings	30.5	20.9	29.7	19.8
5 playings	27.3	18.3	26.5	17.3
10 playings	25.1	16.3	23.9	15.4
Vinyl 0.75 gram				
1 playing	55.1	39.3	47.8	35.5
2 playings	52.4	36.2	43.7	31.7
5 playings	49.0	32.8	40.6	30.0
10 playings	46.7	30.3	38.0	26.6
Nylon 3.0 grams				
1 playing	19.4	16.2	17.3	14.8
2 playings	18.2	15.3	16.3	14.1
5 playings	17.2	14.5	15.5	13.4
10 playings	16.4	13.8	14.8	12.8
Nylon 1.5 grams				
1 playing	27.4	23.0	24.5	21.0
2 playings	25.7	21.6	23.1	19.9
5 playings	24.3	20.5	21.9	18.9
10 playings	23.2	19.6	21.0	18.2
Nylon 0.75 gram				
1 playing	38.8	32.5	34.6	29.7
2 playings	36.4	30.6	32.6	28.2
5 playings	34.4	29.0	31.0	26.7
10 playings	32.9	27.7	29.7	25.7

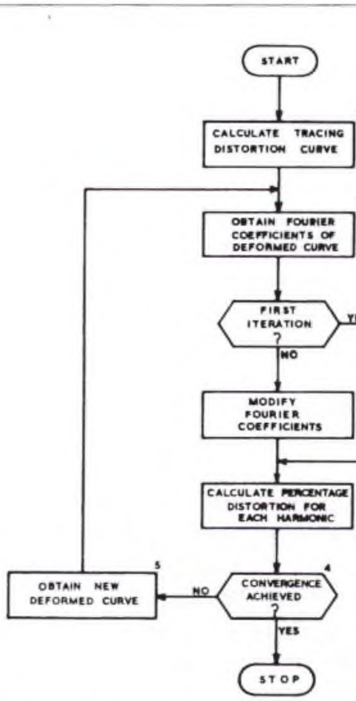


Fig. 12. Flow diagram of calculations.

8. DISCUSSION OF RESULTS

In general, with vinyl the net second-harmonic distortion is less than tracing distortion at the lower frequencies, and higher at the higher frequencies. This might be a function of the increasing velocity and resistance component at the lower frequencies. This was checked by reducing the resistance to zero in a suitable case, that is, where fundamental, second, and third harmonics are well below the top resonance, which would be undamped. Both second and third harmonic distortion were increased by reducing the resistance, but the trend of increasing distortion with increasing frequency was still present (Fig. 16).

The effect of repeated playing is usually to reduce the distortion. With a playing weight as low as 0.75 gram the effect of repeated playing is much less; deformation is fairly small, and the net distortion approaches the tracing distortion in most cases. The effect of biradial styli is to reduce the distortion in most but not all cases. The effect of friction is shown in a typical case recalculated with zero friction (Fig. 17). Friction reduces the second harmonic but increases the third harmonic.

The effect of increasing the playing weight, other pickup parameters remaining unaltered, is shown in a typical case in Fig. 18. Woodward and Werner [18] found that increasing the playing weight (above that needed for proper tracking) decreased the intermodulation distortion, measured using two tones close together. In this case, the second harmonic has been reduced, but the third harmonic has been increased.

Attempts have been made to reduce distortion on playback by predistorting the signal in the inverse of the tracing distortion [19–22]. This ignores deformation and any change on repeated playback. It seems unlikely that

[1] Suggested by Dr. P. G. Craven.

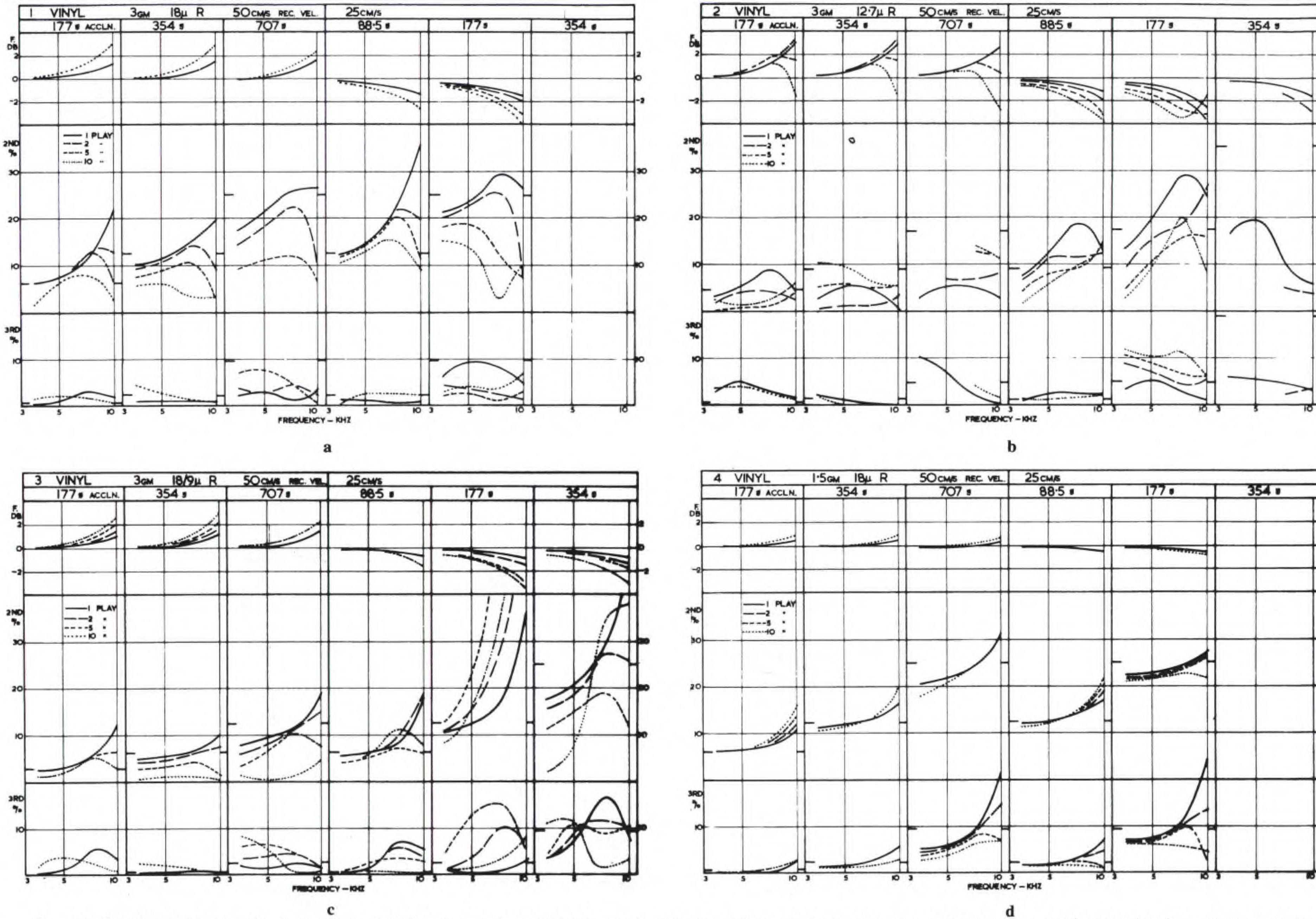


Fig. 13. Calculated change in fundamental level (dB) and second- and third-harmonic distortion (%) on playback of vinyl records at various nominal accelerations, playing weights, stylus radii, record velocities and number of playings.

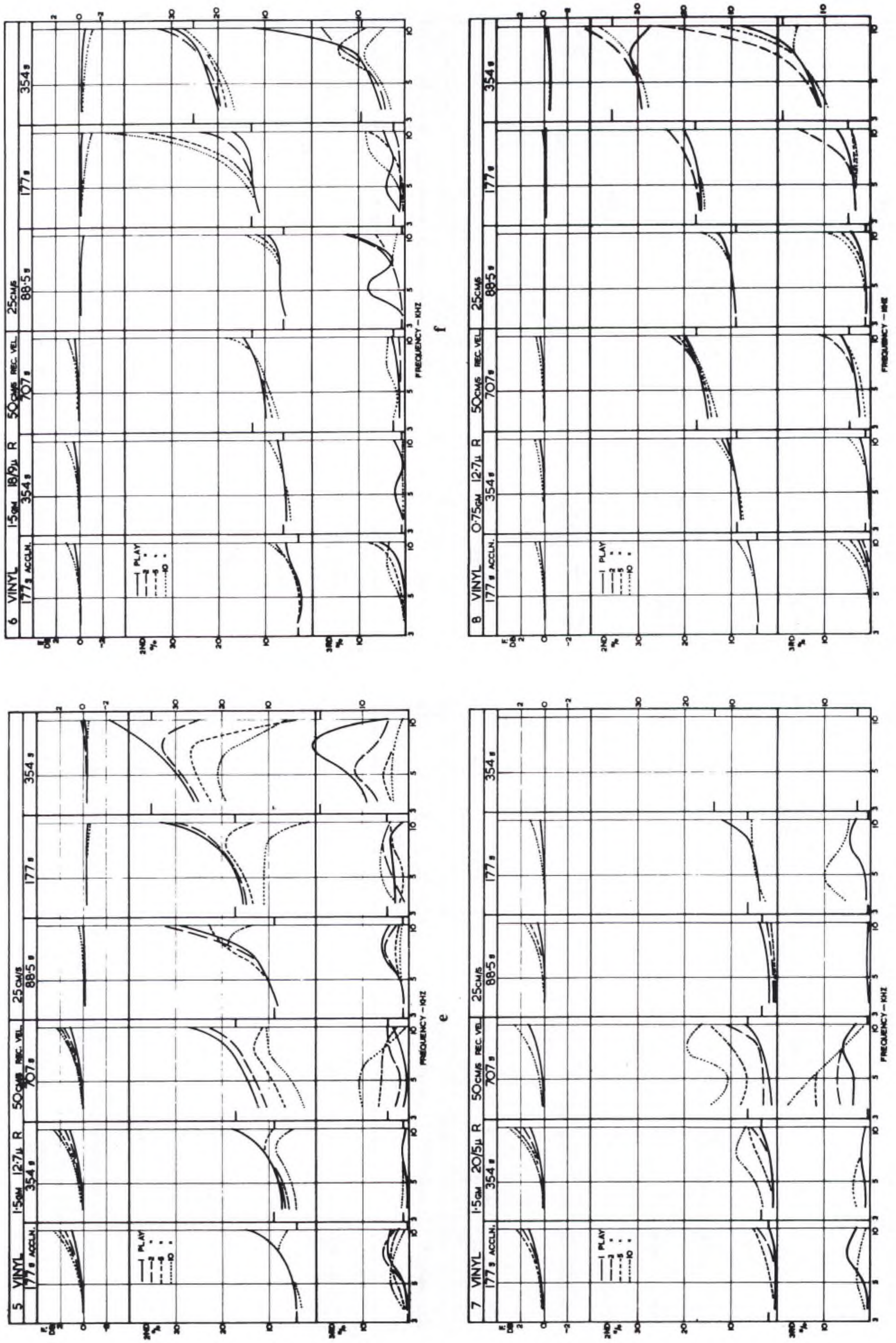


Fig. 13 continued

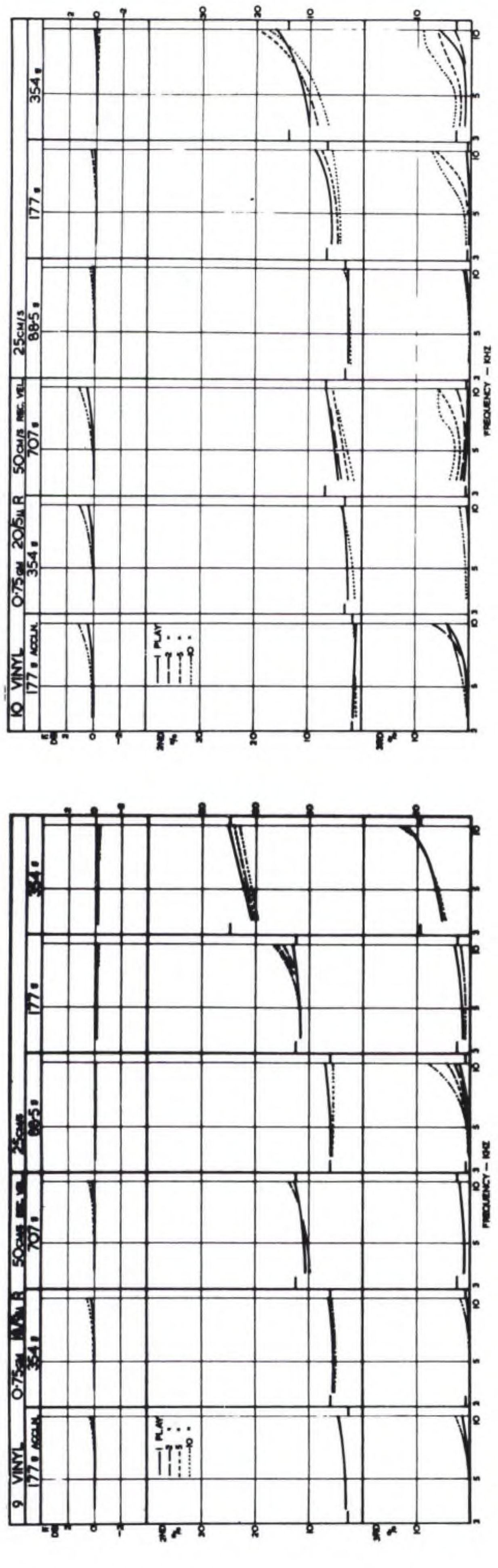


Fig. 13 continued

this could be very successful with existing pickups although Cooper [23] has argued that such predistortion will itself give less deformation distortion. If a future generation of pickups could be produced with a (genuine) playing weight of ¾ gram or less, predistorting for tracing distortion only would be reasonably accurate, and the change with repeated playing would be small. With this playing weight, a biradial stylus would be used to give minimum tracing distortion in any case, as there would be no danger of excessive deformation or frictional damage to the groove.

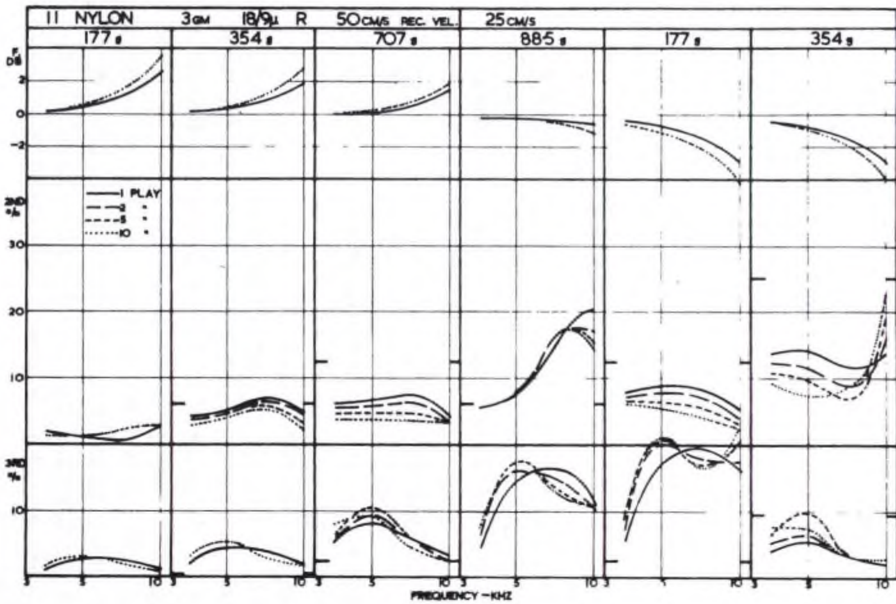
Nylon shows less change with repeated playing than vinyl, no doubt due to the work-hardening. Also, the second harmonic is lower than on vinyl under the same conditions, but the third harmonic can reach high values near the center of the record, although much of this will be inaudible. Whether using a material with a high rate of work-hardening is advantageous or not will depend on the particular case.

Comparison with shellac or harder materials would be of interest. The effect of a harder material was obtained by halving the compliance of nylon in a typical case. This gave less distortion at higher frequencies but more distortion at lower frequencies (Fig. 19).

9. PLAYBACK LOSS

It is well known that a high-frequency signal recorded near the center of the disc gives a lower output from the pickup than a similar signal recorded near the outside diameter of the disc. Hunt [15] has called this difference translation loss, caused by the curvature of the trace, such that the stylus would deform a convex curve more than a concave one, resulting in a net loss of signal. However, the present calculations clearly show that in addition to the treble droop at the inner diameter, relative to the mean level, there is a treble rise near the outer diameter of the record. A check test at an intermediate diameter, at 37.5-cm/s velocity, gave intermediate values. Omitting the curvature factors in a typical case (vinyl, 3 grams, 12.7-μm radius, $177g$, 1 play) gave a rise of 4 dB at 10 kHz at both outer and inner diameters of the disc. The rise at the outside is due to acceleration forces. These act against the playing weight on the hills and with the playing weight in the hollows. The load and deformation are thus reduced on the crests (tending to preserve them) and are increased in the hollows (which are better able to withstand heavy loads). This increases the signal level obtained. The same applies to the inner diameter of the record, but it is obscured by the greater loss due to the greater curvature.

Another term which is sometimes used is scanning loss. This is the analogy with the gap-length effect in magnetic recording, the slit-width effect in the photoelectric playback of film sound tracks, and the aperture-loss effect in optical scanning for facsimile transmission. In these cases, the effect is purely geometrical, occurs in isolation, and is predictable. In record playback, the effect of the finite area of contact is all part of the deformation process, and there seems little point in trying to separate out some hypothetical component.



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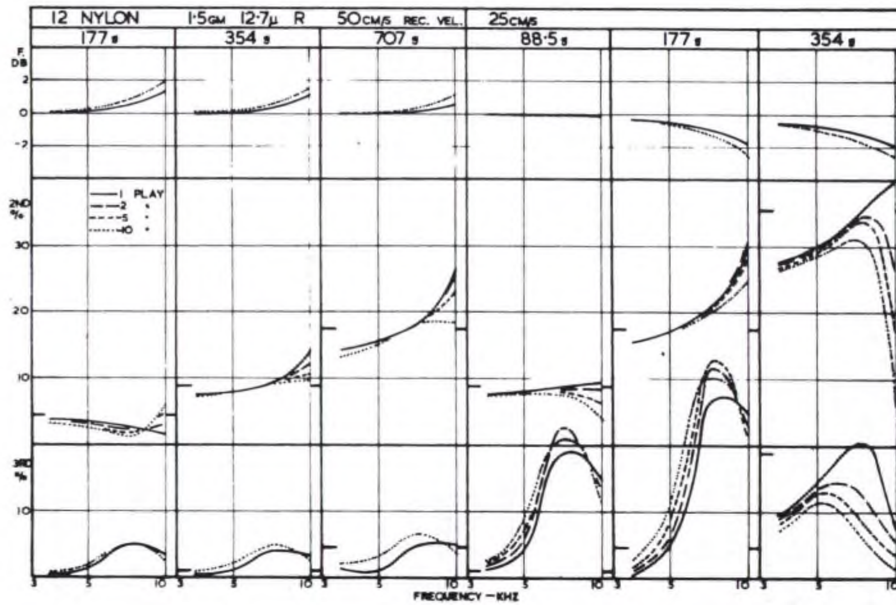

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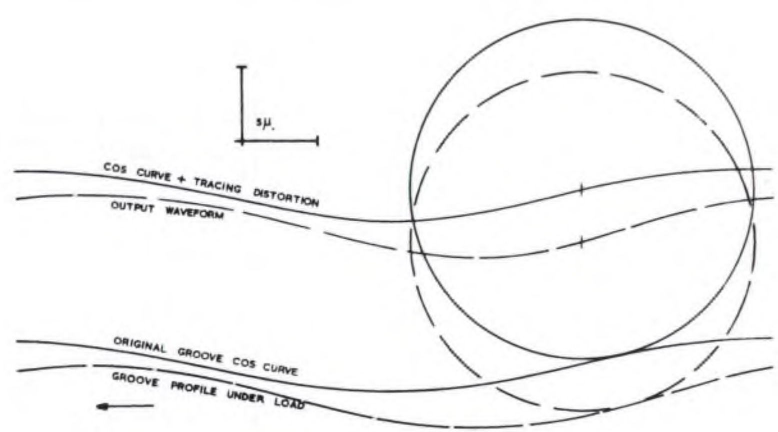

Fig. 15. Profile of typical groove under load.

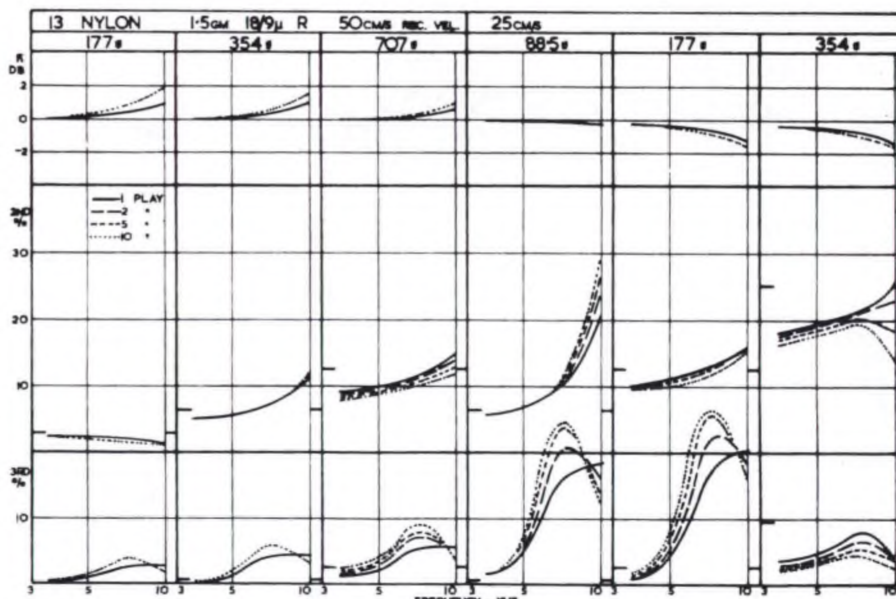

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Fig. 14. Calculated change in fundamental level (dB) and second- and third-harmonic distortion (%) on playback of nylon records at various nominal accelerations, playing weights, stylus radii, record velocities and number of playings.

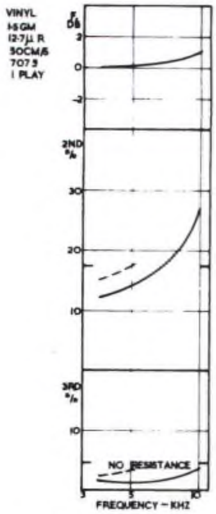

Fig. 16. Effect of resistance.

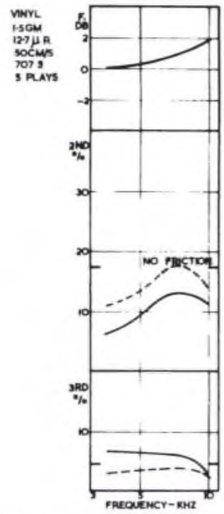

Fig. 17. Effect of friction.

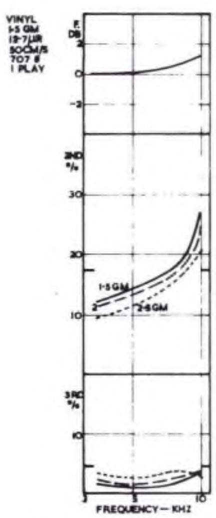

Fig. 18. Effect of increasing playing weight.

Fig. 19. Effect of harder material.

10. FURTHER WORK

The need to elucidate the nature of groove compliance has already been stated. Comparison with practical test data is obviously called for. Predistortion and intermodulation distortion calculations of difference tones on two-tone signals are desirable. Similar work on metal would permit the effect of monitoring of the metal mothers to be determined.

ACKNOWLEDGMENT

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Gerald Garside received the B.Sc. degree in mathematics from Manchester University, England, in 1963 and worked as a graduate apprentice and systems analyst with the GKN group of companies in the U.K. Moving to the newly created University of Bradford in 1966, he held a research assistantship in mathematics, gained an M.Sc. degree in computational statistics in 1969, and then took up his present appointment as a lecturer in computer science.

MORE ABOUT "GROOVE DEFORMATION AND DISTORTION IN RECORDS"[1]

My attention has been drawn to the work of Otto Kornei.[2] I have long known of this work as it has been modified and used by Miller[3] and Kantrowitz.[4] I did not include it in the references in the above paper as, like some other work, it deals with the elastic case only. However, on reading the paper by Kornei, I find that it includes a number of interesting observations, and playback loss and translation loss are clearly defined.

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Bingley, W. Yorkshire, U.K.

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disk recording systems

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Binaural Disc Recording

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A synopsis of major factors involved with the recording on disc of binaural sound. Notes are given on radial playback error, recording techniques, testing and alignment methods, and compatibility with existing standards. Placement of loudspeakers for best effect is discussed, as well as the influence of binaural reproduction on quality, loudness, and scope of records.

IN THE field of records and recording it is time to take stock of the situation and to plan for the future. What has been accomplished thus far? First, a large repertory has been recorded on 78-rpm discs, but these are of indifferent quality. Second, other speeds have been introduced, including the 33⅓-rpm records with a "long play" per side. Third, a very large repertory has been recorded at 33⅓ rpm but the time is not far away when a record producer will have to choose between selections which have already been recorded once or twice and selections which are not popular or are not good music. Last, techniques have been developed for making records of extremely wide range and low distortion, although this is not often accomplished in practice.

The medium must be improved, but this can be accomplished only by making a new creation out of each selection through added dimension, direction, and perspective. Binaural records are the solution to the problem because the binaural medium is practical, economical, and irresistibly exciting in records, and, unlike color television, it is totally compatible.

The method to be described here for producing binaural records is compatible. It is a practical working basis for making binaural records at no increase in cost over regular records. There are various ways in which two channels can be produced on records. Before this system was chosen, however, several were examined to determine which would be most practical. These other systems are briefly described as follows:

1. *Single-sideband carrier system.* For practical applications, this system has two faults. The frequency range is limited to less than half that of which the record groove is capable on the regular basis; also, the record cannot be played on a monaural basis by those who wish to use it during the period when their equipment is being converted for binaural playback.

2. *Vertical-lateral modulation of the same groove.* There are many things wrong with a practical application of this method. For example, cost of a playback arm and cartridge would be prohibitive for wide use—after all, present cartridges cannot all be discarded at once. The recording cutter would be an expensive, fantastically delicate monstrosity, and, in the final result, crosstalk between channels would take place at certain frequencies.

3. *Each channel pressed on opposite sides of a record.* A plant engineer knows that it is practically impossible to align stampers rotationally in the press to within 0.010 in. in arc referred to the outside grooves. Also, if one side were off-center with respect to the other side, a new playback machine would have to be built.

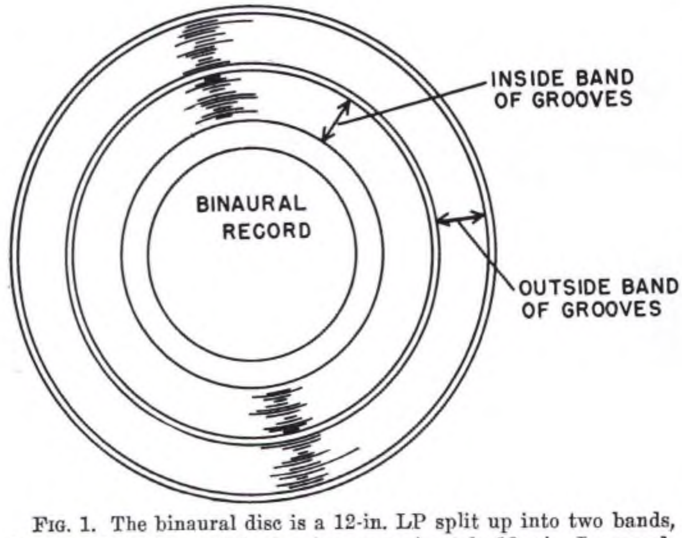

FIG. 1. The binaural disc is a 12-in. LP split up into two bands, inside and outside, which play for approximately 12 min. In recording, the two styli travel along a radius, spaced 1 11/16 in. apart.

4. *Interleaving of grooves on the same side of the record.* With this method, a new type of cartridge would be required, having two needles close together. First, nobody would be willing to record this way. The tracks have to be recorded one at a time on synchronized machines, a very expensive process. Also, this method has the same unfortunate characteristic as LP's; there is a wide range of operating diameters. It is still required to go all the way from outside to inside, unnecessarily.

There are other possible methods, but for total compatibility and low cost the two-band method is the best. A 12-in. LP disc is split up into an inside band and an outside band (Fig. 1). Each band in itself does not have as great a change in diameter from outside to inside as a regular LP; therefore the translation loss problem is minimized. As a matter of fact, the inside of an LP disc would be acceptable as a medium, except for the drastic attempts made by some to pre-equalize it during recording to match pre-emphasis on the outside.

The recording styli are spaced 1 11/16 in. apart on a radius, just as with standard records. Playback styli are spaced similarly, and there is a certain geometry of arm, pivot point, and offset angle which allows only about ± 0.010 in. of error in departure from a mutual radius of the two playback points as they travel along the bands of grooves. If the record is slightly stretched or warped, there is a little lateral lost motion in one or both of the cartridge mountings, in order to let them find the right groove and stay there. The rattle in this lost motion is taken up completely with a viscous damping grease. There is a lateral adjustment to permit accuracy in the 1 11/16-in. spacing, and a longitudinal adjustment to synchronize the two playback points on the same radius. There is a test record to allow this synchronization to be performed, and also to serve as a standard for alignment of cutters on the recording lathe.

The system permits about 12 min. per side on an LP without the need for extremely fine grooves. It is free from synchronization bugs caused by eccentric pressings, or by unsymmetrically stretched or warped metal parts or pressings. Only simple modifications of existing recording equipment are required. It is necessary only to produce two individually sprung cutters on the carriage, spaced 1 11/16 in. apart.

TRANSLATION LOSS

The subject of translation loss must be considered from a fresh point of view. In simple terms, translation loss is the difference in response and performance between outside and inside. But with this method we do not go more than halfway from the outside to inside of the disc for one band. There is no valid reason for having the frequency response of one playback channel identical with the other. For example, we could leave off the pre-emphasis on the inside band, make it "flat," and then play it back "flat." The spectrum of surface noise at the smaller diameters for the same record compound is narrower; therefore, there will not be any great difference in surface noise between the two bands. We use only partial compensation for translation loss, not only because full compensation is undesirable, impossible, and distortive, but also because the time is approaching when ½-mil radii will be used for playback. With ½-mil points, the mildest compensation will be enough.

SYNCHRONIZATION

Binaural sound cannot be compared to stereo sight. We will not wear earphones any more than we will wear stereo spectacles constantly at the motion pictures. Instead, we are going to have loudspeakers in a living-room and sit comfortably and unencumbered. Thus, the whole approach to microphone spacing technique and allowable synchronization error changes. By using a 6-in. (human ear) spacing of microphones and by wearing earphones for playback, we could get phase effects at 1,000 cycles, but only if the selection were played back precisely the same way it was recorded. The relative position of two heads, even in tape, is not mechanically stable enough to keep the listener within a half wavelength at 1,000 cycles. This point is unimportant, however. The listener may sit or stand in a random position in a living-room with two loudspeakers spaced perhaps 12 or 15 ft apart. Also, the living-room acoustics are added to the recorded acoustics. The listener may be closer by 1 or 2 ft to one speaker. This does not impair the binaural effect. It does mean that, for each recording, we shall have to study the philosophical question of whether we want to provide the listener with the original acoustics of the concert hall or studio, or whether instead we want to produce the effect of transporting the artist to the living-room. Here

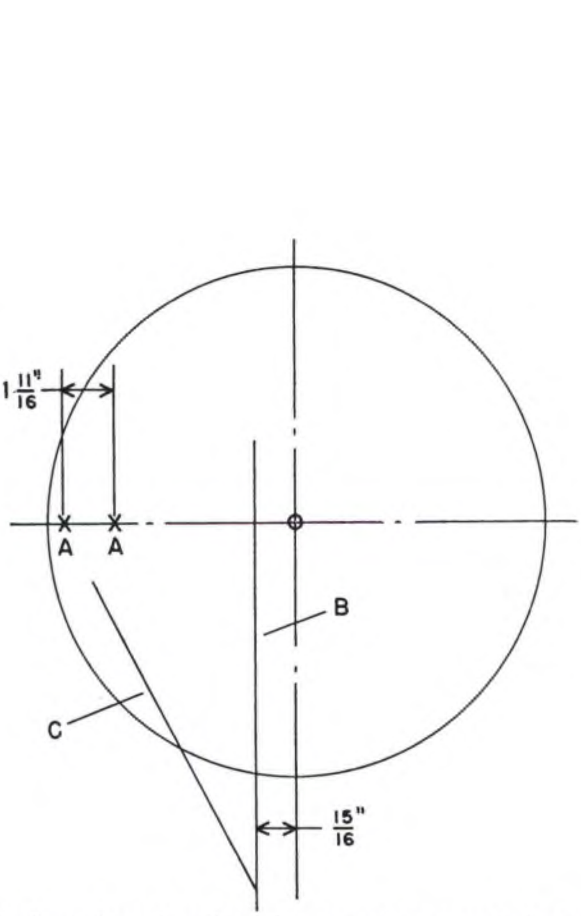

FIG. 2. *A–A* are the points of contact of the two playback styli at the start of the record. Line *B* is the locus of points along which the pivot point of the arm *C* must be located. The angle *BC* is the cartridge offset angle.

is an important artistic point that was disregarded in the production of regular records but which is an exciting new facet of binaural records.

The combined rms synchronization error from beginning to end of the entire process of recording a binaural disc is about one wavelength at 1,000 cycles. In terms of phase delay, this is equivalent to sitting in the tenth row of a concert hall and leaning over or turning the head.

The mechanical playback error caused by the fact that the two playback points are not always on the same radius is included in Fig. 2, which shows the geometry of the playback.

At the outside of the record, the error is, let us say, positive; in the middle, negative; and at the inside, positive again. Therefore, we may say that the error describes a "cycle" and passes through two points at which the error is zero. The geometry of any length of arm can be constructed by referring to Fig. 2. Draw x and y coordinates through the center of the record. The x coordinate is the radius on which the two playback points approximately lie at the beginning, or outside, of the binaural record. Depending on the length of the arm, the pivot point of the arm will be located somewhere along a locus of points represented by $x = -15/16$ in. The offset angle of the cartridges in the arm will be equal to the acute angle of the triangle formed by the coordinates and the center line of the arm at the starting position.

To sum up the subject of synchronization, we may observe that it is desirable to have the "ears" or channels in perfect step. With radially driven playback cartridges, this would be automatic. However, over the short arc of swing of the arm in traversing the narrow band of grooves, set up according to the geometry given here, there is a completely inconsequential maximum error; in addition, the tracking-angle error of the cartridges themselves is lower than ever before.

COSTS

Producing records by this method will cost no more than under other systems. There will be a nominal amount of conversion of existing disc-recording equipment, and synchronized dual-track tape machines will have to be installed to record originals. But the expense of copyrighted music and of musicians will be less per side because the time is less. Musicians will be happy, because no recording originally made by present methods can be "converted" by any mysterious secret process to a binaural performance. The practice of hiring a hall and playing old records through a mammoth loudspeaker into two binaural microphones located in the first aisle is completely false; the result is different from the original only in that it sounds worse. There is no synthetic substitute for binaural recording.

The added cost of playback equipment is of little consequence. Every audiophile who has a spare cartridge, an amplifier, and a speaker requires, in addition, only an arm and a little work. The cost of a binaural phonograph as compared with a regular phonograph of the same quality of construction referred to list price has been estimated at between \$30 and \$40 more. And in equipment under the \$100 level, the binaural improvement is so tremendous that purchase becomes imperative for people who are at all ear-conscious. The cost of a binaural amplifier will be very little more than the equivalent push-pull throughout amplifier of the same construction. For twenty years in engineering circles binaural recording has been considered an impractical medium for mass consumption. Many of us can remember the original Bell Laboratories' "audio perspective" demonstrations at the Institute of Radio Engineers. Now, with 33⅓-rpm speed well established, three-dimensional records are a practical medium because the cost is practical.

The field of audio is growing rapidly. New technical developments are arriving, and many more people are becoming interested. A large percentage of new arrivals to the audio scene are discriminating television watchers who have been driven away from most television programs. These people will be discriminating listeners as well as discriminating watchers. Therefore the development and direction of our technical growth must harmonize with the requirements, tastes, and interests of these newcomers, for it cannot otherwise survive.

Monogroove Stereophonic Disk Recording*

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The day seems to be fast approaching when the high-fidelity enthusiast will demand the additional advantages of stereophonic reproduction. "Stereo" radio broadcasts and special recordings have been made available which readily acquaint him with its benefits. In this paper a new type of "compatible" disk record is described which contains simultaneous recordings for two-channel stereophonic applications, yet the economy of running time is as great as that of the customary single-channel disk.

THE CRITICAL listener, given the benefits of the full advancement of the art, can be heard to say, today, that his recording and reproduction system is so good he is constantly annoyed by distracting sounds coming from his loudspeaker. Along with the splendid over-all quality of wide-band low-distortion musical reproduction, he must be content to hear the rumble of auditorium air-conditioning systems, the squeak and creak of orchestra chairs, and the occasional sneeze and cough of musician or audience. Our critical listener, attuned to the splendid reproduction of his favorite music, will insist that these sounds are of diabolical origin, marring, dulling, and scratching an otherwise beautifully polished performance. But is he totally at fault in criticizing the human frailties of his musician or audience brother in such alarming terms?

Not at all! Poor listener, a sparklingly transparent medium has been opened to him by wide-range low-distortion reproduction. Consequently, he hears all sources of sound—those he wishes to hear and those, as well, which are thoroughly distracting. He cannot "tune out" those he does not want to hear. Yet, seat him in the concert hall and he expresses no difficulty in hearing the music to its fullest enjoyment, unaffected by disturbing sounds.

The reason? Simple. It is one of those differences brought about by multiple-channel or "stereophonic" sound, compared to single-channel or "monaural" reproduction. In the concert hall his attention is directed toward the orchestra. He listens directionally to such a degree that irrelevant distracting noises do not annoy him in any way. If his wide-range home sound system were stereophonic, he would be annoyed to a far less degree by irrelevant sounds.

This advantage of stereophonic reproduction is one which has not been stressed to any great extent up to the present time. Followers of the art are acquainted with the development of the multiple-channel sound system and its obvious advantages—breadth, depth, and even an apparently vertical dimension—in short, the addition of a truly third-dimensional quality to the sound.

It is not the purpose of this paper to review these and other advantages of stereophonic sound over the single-channel system. This material has been covered thoroughly elsewhere.[1]

The literature has made it quite clear that a three-channel stereophonic system is almost completely ideal. However, the greatest step forward in stereophonic quality arises when a two-channel system takes over from the monaural. Going from two to three shows a definite improvement, of course, but so much enjoyment can be had by the addition of a second channel to an existing single channel that this becomes a logical, economical step for the home enthusiast.

Sources of the two-channel program material are limited at the present time to occasional two-channel broadcasts by such stations as KFAC and WQXR and to special disk records containing two separate grooves. These have been a stimulus to the listener to set up two-channel stereophonic equipment, and much credit is due the people who have made these broadcasts and records available.

This paper describes a two-channel disk recording system which is under development by Bing Crosby Enterprises, Inc., at the present time. It is capable of providing two distinct channels of stereophonic sound in a single disk record groove. This it does by virtue of two-way stylus motion.

Although virtually obsolete again today (its second time

* Presented at the Second Annual West Coast Convention of the Audio Engineering Society, Los Angeles, February 4-6, 1954.

[1] "Symposium on Reproduction in Auditory Perspective," *Electrical Eng.* (January, 1934).

round), vertical or "hill-and-dale" recording has long since proved itself to be a high-quality method of disk recording.[2] Lateral recording, of course, is the accepted standard today, be it in 78-rpm standard grooves or on LP. There is no fundamental reason why a vertical record and a lateral record cannot be simultaneously cut by one stylus. One great advantage of such a method will be found in the positive retention of phase relations between the two signals. There exist, in the professional field, two pickups of well-known design which are capable of reproducing either a vertical or a lateral record. These are the RCA 4875 and the Western Electric 9-A. A switch is normally provided to change operation to vertical or lateral, but the coils of the vertical and lateral units are capable of being wired in bridge circuitry in such a manner that the vertical motion of the stylus may always generate a signal at the input of one amplifier while lateral motion may generate a voltage which appears only at the input of a second channel amplifier. Thus, without special design, pickups exist which can be adapted for present-day demonstration purposes. Obviously, many of the existing commercial pickups for home use could be replaced by inexpensive modified versions which could generate signals in this manner.

The problem of playing a "VL" (vertical-lateral) record is therefore almost nonexistent. The main problem is to provide some means of cutting the disk.

This has been overcome in an experimental recorder of somewhat limited frequency response in the manner shown in Fig. 1. Three coils are wound about the legs of a T-shaped armature structure which is supported in magnetic fields at the point indicated by the black dot at the center of the T. This support provides two-way compliance as a pivot so that the cutting stylus can move laterally or vertically. If an alternating current is applied to the two series coils marked V, it will be observed that a vertical motion is imparted to the stylus point. Similarly, an alternating current applied to coil H will drive the stylus laterally, all motions being in the nature of rotation about the pivot centrally located in the T armature.

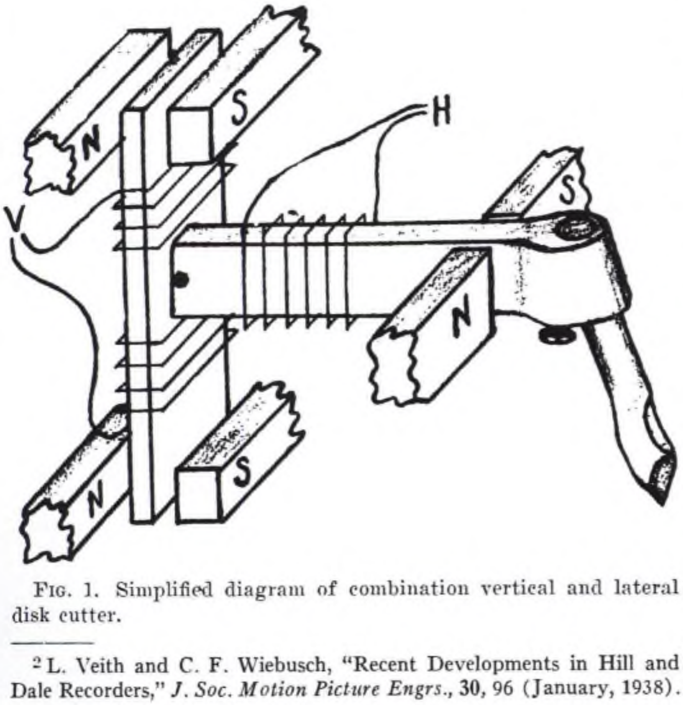

FIG. 1. Simplified diagram of combination vertical and lateral disk cutter.

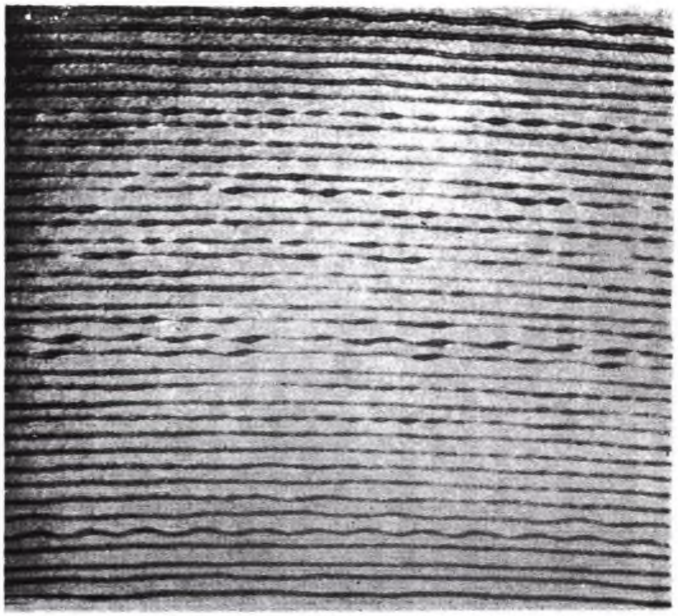
FIG. 2. Magnified grooves of disk recorded with the cutter shown in Fig. 1.

Figure 2 is a reproduction of magnified record grooves showing the effect of a stereophonic sound source. In this case a playlet was being read by several people on the stage. At the outermost grooves it will be observed that the recording is almost completely lateral, indicating pickup essentially by microphone No. 1. This is soon followed by some material almost exclusively vertically recorded, picked up mostly by microphone No. 2. Then comes some interesting groove material comprising both vertical and lateral recording. After this there is a section of low-frequency lateral tones mixed with some much higher-frequency vertical tones, since two people, at either side of the stage, were talking at the same time. Lastly, there is an almost purely lateral recording, ending this photographed sample recording.

Crosstalk might seem to be an objectionable feature of the system. Actually, undistorted crosstalk turns out to be no problem at all, since in effect it only reduces the apparent separation of the loudspeakers in the reproducing system. Any such crosstalk can be reduced in effect, if it ever should become loud enough to become objectionable, by separating the speakers very slightly. Actually, it has never been so loud in tests made so far as to warrant this.

[2] L. Veith and C. F. Wiebusch, "Recent Developments in Hill and Dale Recorders," *J. Soc. Motion Picture Engrs.*, **30**, 96 (January, 1938).

Generally, a vertical record will not crosstalk with high distortion into the lateral amplifier. On the other hand, the highest distortion produced in the system occurs because of pinch effect of high-amplitude lateral recordings causing vertical motion of the stylus.[3] This is strongly second harmonic and under certain conditions can be slightly noticeable. Since it is reproduced from the vertical loudspeaker while a loud sound is emanating from the lateral speaker, there is fortunately a great deal of masking of the audible distortion.

No percentages of crosstalk or distortion are given herewith, since they would be quite difficult to interpret in the face of the rather limited response and inherent distortion of this experimental cutter.

[3] J. A. Pierce and F. V. Hunt, "Distortion of Sound Reproduction from Phonograph Records," *J. Soc. Motion Picture Engrs.*, **31**, 157 (August. 1938).

Work is continuing on cutter development at the present time. Reduction of recorded harmonic distortion and extension of frequency response are the primary objectives of further research on the cutter.

Commercially, this system enables a disk to run the normal playing time, since the included vertical track requires no additional groove spread. Before he installs his VL pickup and second amplifier and speaker, the high-fidelity enthusiast can play VL records on his present lateral pickup provided that it has reasonable vertical compliance. The Weathers and GE pickups work nicely in this manner.

Thus, the VL system of stereophonic sound is automatically (to borrow a term from the color television people) a compatible one. From what work has been done to date, it promises, as an alternate to two- or three-track tape systems, to point the way to the future large-scale introduction of "stereo" sound in the home.

Recent Developments In Stereo Disc Recording*

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Westrex Corporation, Hollywood, California

The paper describes an analog computer analysis made to determine the cause of resonances in the upper high-frequency response of the Westrex StereoDisk cutter, and the mechanical modifications which were made in accordance with the solution suggested by the computer. The result is a smooth response to at least 15 kc verified by actual recordings. The paper also describes the design and application of an electromagnetic device which automatically controls depth of the groove in accordance with the pitch, being actuated from the pitch control mechanism of the Scully recorder. Included also is a description of a study of contemporary stereo disc pickups with regard to frequency response and crosstalk between channels.

INTRODUCTION

THE RECORDING characteristics of the Westrex StereoDisk system have been described in some detail in a recent paper by Davis and Frayne[1] and in general have met with the approval of the disc recording industry. The 3A cutter described in that paper was of essentially the same design as the original engineering model on which all the early demonstration stereo recordings were made. As soon as the original rush demand for stereo cutters had been met, studies were initiated to improve the performance and eliminate any deficiencies in the original design. It had become apparent, for example, that the original 3A and its production successor, the 3B, had characteristic resonant peaks and valleys in the upper audio spectrum. A by-product of these resonances was an undesirable amount of crosstalk in the recorded groove.

RECENT ADVANCES IN RECORDER DESIGN

Before attempting to discuss the approach employed to eliminate these undesirable characteristics, the basic design of the early recorder will be reviewed. The recorder shown in Fig. 1 contains two coil assemblies, one associated with each channel. Each comprises a drive coil and a feedback coil located in annular gaps in separate pole pieces. Vee-shaped beryllium copper coil support springs hold and position the assemblies, and by means of these springs the assemblies are constrained to have no motion other than that parallel to their axis. This motion is transmitted to the tubular stylus support member by means of wire links braced with magnesium sleeves. The magnetic gaps of the drive and the feedback coils are arranged in a series parallel fashion, and magnetic flux is provided to the system by a single magnet. The arrangement of magnetic paths ensures equal flux densities in the corresponding gaps. The shaded areas between the magnetic gaps indicate copper plugs or shields which reduce the inductive crosstalk from the drive coils to the feedback coils.

In the complex mechanical assembly such as that of this cutter, design problems are accentuated by the tendency of

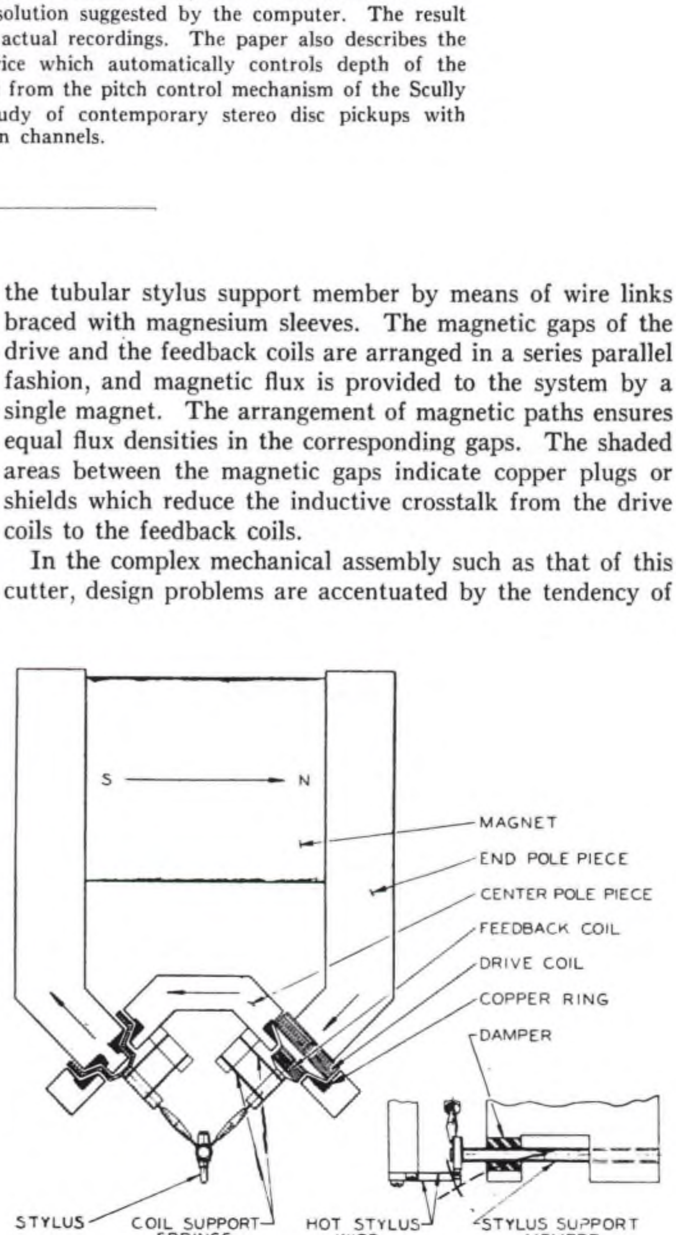

Fig. 1. Simplified illustration of 3A StereoDisk recorder.

* Presented February 18, 1959 at the Sixth Annual Western Convention of the Audio Engineering Society, Los Angeles, California.

[1] C. C. Davis and J. G. Frayne, Proc. Inst. Radio Engrs. 46, 1686 (1958).

each mechanical element to vibrate in multiple modes. These vibrations result in the well-known effect of resonances in the frequency response which, with earlier Westrex StereoDisk recorders, tended to occur in the frequency range between 10 and 20 kc. Since it is difficult to equalize the effects, the main problem was how to revise the mechanical system so that the frequencies of these resonant phenomena could be substantially raised beyond the normal recording range to the neighborhood of 20 kc.

In a preliminary analysis of the problem, it was decided that although a number of factors were contributing, the main cause was to be found in the mechanical characteristics of the moving assembly; that is, the two coil assemblies, the linkages, the stylus and its support tube.

To determine the cause and locate the origin of these effects, it was decided to carry out an investigation employing an electrical analog of the system rather than attempt to use empirical methods. A comprehensive study was made along these lines for Westrex by Computer Engineering Associates, Inc. of Pasadena, and the appropriate electrical analog was determined. While it is well beyond the scope of this paper to discuss and analyze the actual configuration which was employed, it should be mentioned to emphasize the complexity of the undertaking that the equivalent electrical circuit contained upwards of sixty components. A computer representation was effected by applying an oscillator to the input of the circuit and connecting to the output an oscilloscope or an automatic plotting device. Tests were first carried out to verify that the analog configuration was a true representation of the mechanical structure it was presumed to represent. Variations were then introduced electrically into the design, and the frequency characteristics recorded for each such variation. A summary of the three main areas of investigation and changes made as a result of the findings are detailed below.

DRIVE COIL ASSEMBLY

It was definitely established at the outset of the investigation that the drive coil assembly made no adverse contribution to the frequency response of the recorder, except for some minor effects in the unlikely event of a nonuniform magnetic field in the air gap. It was decided, therefore, that no changes were necessary in the original design of the assembly to eliminate the abnormalities in frequency response and crosstalk characteristics in the upper audio spectrum.

WIRE LINKS

The bending and axial modes of the linkages were investigated and with the initial configuration, normal modes involving predominantly lateral motion of the wire were apparent at frequencies of 21.3–21.7 kc. It is known that the linkages exhibit frequency characteristics which are in the same range as other effects but it is apparent that they involve mechanical energy which is substantially less than that associated with other portions of the recorder, and that,

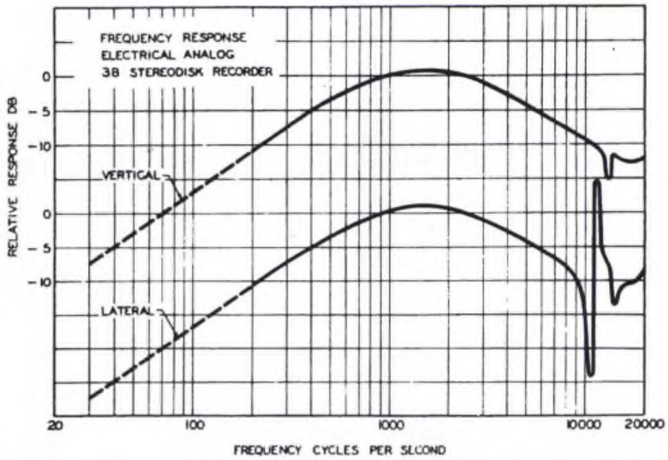

FIG. 2. Frequency response of 3B StereoDisk recorder obtained with electrical analog.

therefore, there is no contribution which is detrimental to the frequency response. On the basis of the analysis of the linkages it was decided that the only change necessary was that of increasing the diameter of the wire to decrease the axial flexibility.

STYLUS SUPPORT TUBE

In the analysis of the effect of the support tube, bending, torsion, and tension modes were investigated separately, and it was found that the lowest resonance frequency due to bending modes other than the principal one at between 1 and 2 kc was 13.2 kc in vertical motion and 14.0 kc in lateral motion. This motion was attributable to an antiresonance of the clamped-free tube. It was found that the torsion mode produced a serious resonance in the lateral motion at approximately 11 kc, while no effect was found in the vertical mode of vibration of the recorder. The electrical analog response for both vertical and lateral modes of motion is shown graphically in Fig. 2. It will be noted that neither of these curves shows the flat response expected of a constant-velocity cutter even though the analog included representation of the known amount of feedback used under actual recording conditions. It is quite probable that some resistive elements, including the load imposed by cutting the acetate and not included in the analog, account for the lack of a flat response in the area from 1 to 10 kc found in actual frequency recordings.

It was a simple matter to alter at will on the computer the constants of the stylus support tube assembly to move the unwanted resonances well beyond the customary upper limit of 15 kc to the neighborhood of 20 kc. From these experiments, it was found necessary for the tube to be of greatly increased stiffness in both bending and torsion and to provide a mounting for it that would prevent torsional motion but allow lateral and vertical motion of the stylus. These requirements were met by doubling the diameter of the tube, retaining the same wall thickness, and reducing its length to ⅔ of the original. The new mounting at the

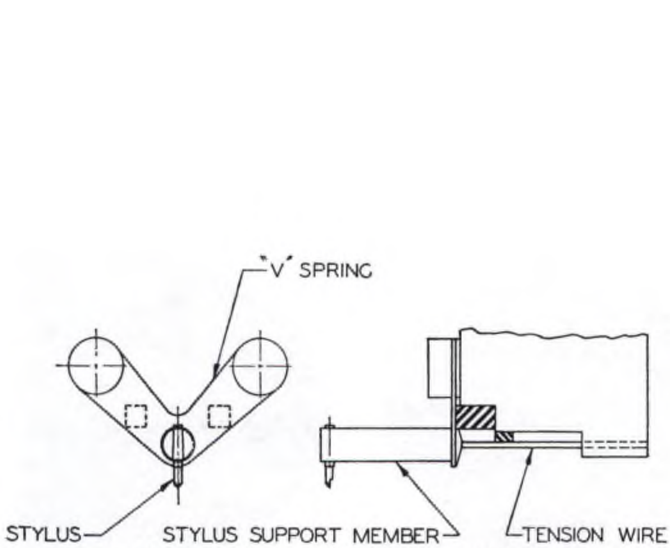

FIG. 3. Simplified illustration of 3C StereoDisk recorder.

end of the tube opposite the stylus was made in the form of a V-shaped spring that was very rigid in torsion but compliant in bending and twisting. An axial tension wire was located behind the V spring to prevent longitudinal motion of the tube. These changes are shown in the simplified illustration of Fig. 3. With the new beam length of ⅔ the original, the angle of stylus motion relative to the vertical with the coils driven in phase was retained at its original angle of approximately 23 deg from 30 cy to at least 8 kc. This represents an improvement over the actual performance of the 3B design in which the vertical angle varied from 23 deg at low frequencies to the 30–35 deg region at upper frequencies.

It should be emphasized that the main deficiency of the original design lay in the stylus support member and its clamping since the requirement was for a beam which possessed moderately low stiffness in the bending mode but high stiffness in the torsional mode. It was impossible to obtain simultaneously these two conflicting requirements with the earlier suspension method. The analog frequency response for the new support tube is graphically shown in Fig. 4 in which the disturbing resonances of Fig. 2 have been pushed beyond the 20-kc limit. The physical embodiment of the changes suggested by the computer is shown in Fig. 5 in which the new shorter and wider support tube and

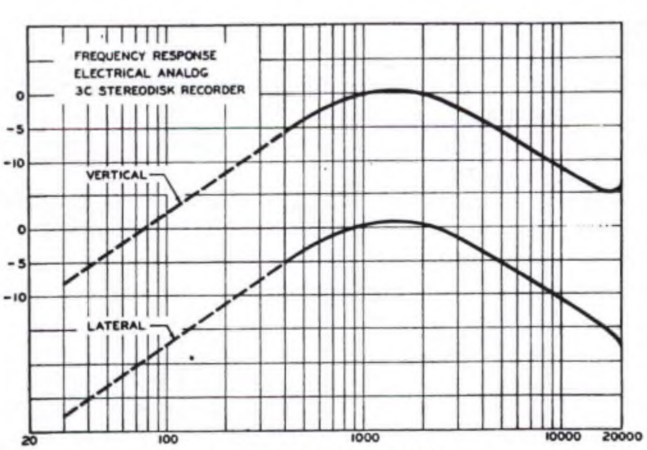

FIG. 4. Frequency response of 3C StereoDisk recorder obtained with electrical analog.

the associated V-spring support and tension wire are plainly indicated.

RECORDING CHARACTERISTICS

The response curves discussed above were obtained from the analog computer for the conditions indicated. The next step was to determine how the proposed modification affected the actual recording characteristic. For comparison purposes, the unequalized frequency response of a single channel of a typical Westrex 3B recorder is shown in Fig. 6 and this shows the combined effect of both lateral and vertical resonances. In this instance the characteristic was obtained by using a recording speed of 78 rpm for all frequencies and using a reproducing speed of 78 rpm below 1 kc and 33⅓ rpm for frequencies above 1 kc. This was done in order to eliminate reproducer errors at the high frequencies. The pronounced resonance effect around 12 kc will be noted.

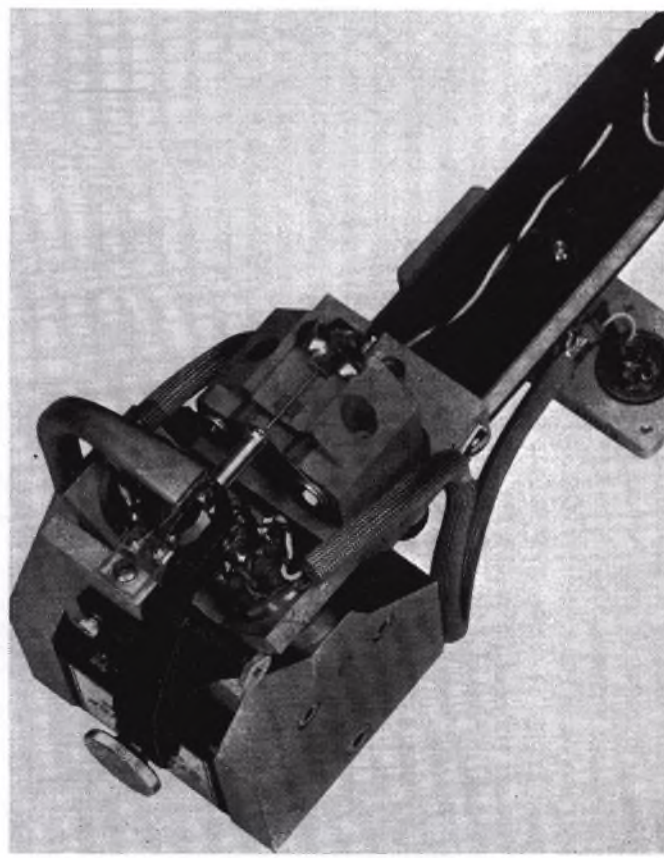
FIG. 5. Bottom view of 3C StereoDisk recorder.

Figure 7 shows the frequency response of a single channel of a typical 3C recorder showing the combined effects of lateral and vertical motion. This shows the dramatically improved response which has been obtained by incorporation of the changes discussed above. The recordings were made again at the same recording and reproducing speeds

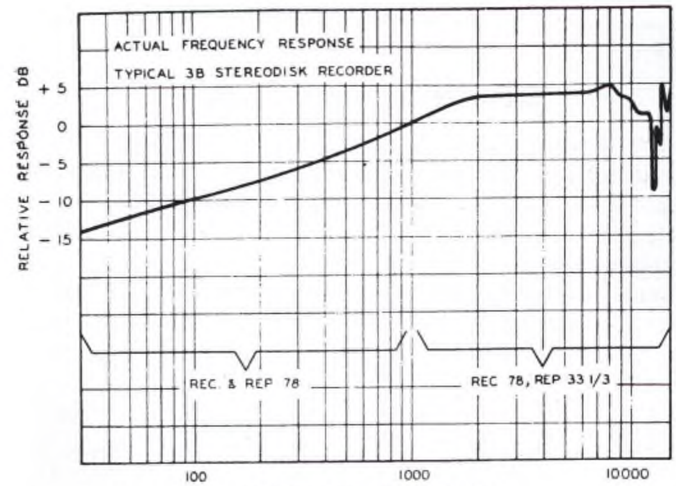

FIG. 6. Frequency response for 3B StereoDisk recorder (vertical and lateral channels combined).

as those of Fig. 6. The broken portion of the upper curve shows the equalized response in the region where the feedback is largely inoperative and resort is made to use of a passive equalization network. A by-product of the new design was an increase in sensitivity of 4–6 db at 1 kc. This may be attributed to several factors; namely, increased compliance of the support tube and elimination of viscous damping of the latter.

In the course of the computer analysis, it was noticed that the Lissajous figure on the oscilloscope showing relation between output and input changed from a circle to a straight line at the resonant frequency, immediately shifted phase by 180 deg, and then resumed the circular shape as the frequency was increased beyond resonance. This indicated that severe crosstalk from one side of the stereo record to the opposite side would occur at a resonance point. Pre-

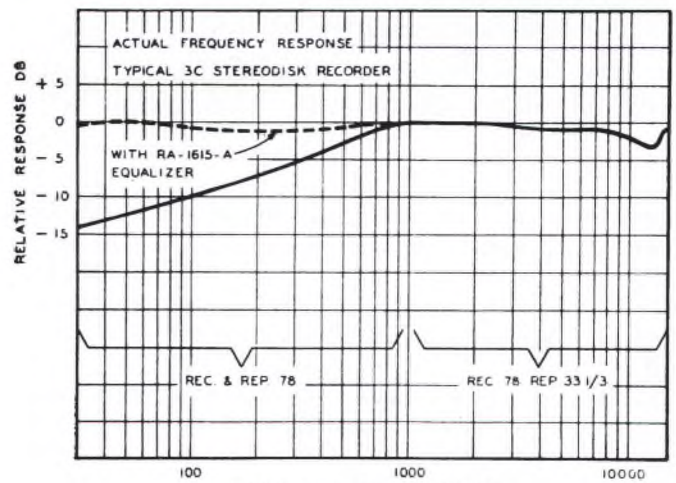

FIG. 7. Frequency response for 3C StereoDisk recorder (vertical and lateral channels combined).

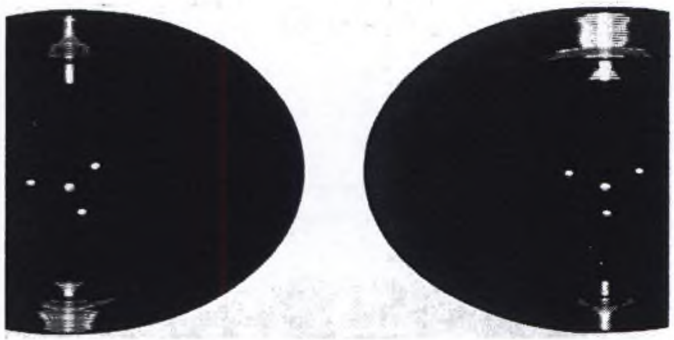

FIG. 8. Light patterns showing frequency response and crosstalk for left and right channels of a 3B recorder.

vious measurements with a stereo pickup had indicated crosstalk values as high as – 5 db in this region, but it was suspected that the trouble lay mainly in the operation of the pickup. Since it now appeared that the recorder was also at fault, it was decided to make a study of the amount of crosstalk actually recorded on the disc. The optical method of measuring groove modulation[2] was used to obtain the photographs shown in Figs. 8 and 9. The technique employed a sharply collimated light beam incident on the

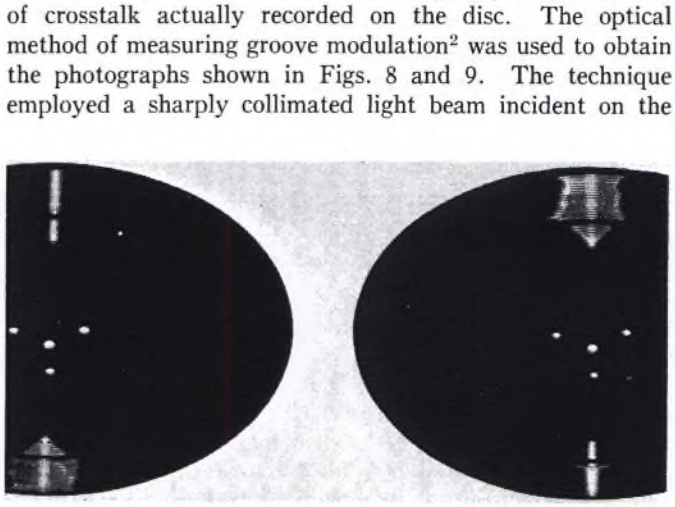

FIG. 9. Light patterns showing frequency response and crosstalk for left and right channels of a 3C recorder.

record surface at about 45 deg, the camera being mounted at essentially the same angle. This permitted simultaneous viewing of the signal on, say, the convex side of the groove and the resulting crosstalk on the concave wall. The opposite condition would result from impressing the signal on the opposite channel.

Figure 8 shows on the left the signal at the bottom and the resulting crosstalk for a 3B recorder (unmodified) at the top of the picture. Starting at the bottom, the first band indicates 1 kc, and the succeeding bands correspond to every integral value of kc up to 15 followed by 1 kc for comparison purposes. Following the break in the pattern, recordings of 30 cy to 1 kc are shown. The dip at 12 kc and the sharply rising response up to 15 kc are plainly shown. The increase in crosstalk for this same region is

[2] G. Buchmann and E. Meyer, Elek. Nachr.-Tech. 7, 147 (1930).

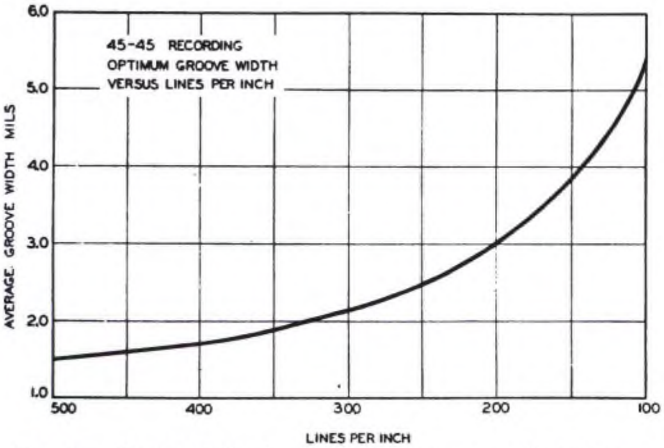

FIG. 10. Characteristic of relationship between optimum groove width and pitch for 45–45 recording.

plainly visible at the top of the photograph. The corresponding case for the right channel is shown at the right of Fig. 8. The dip at 12 kc is even more pronounced here.

Figure 9 gives the same series of photographs for the (modified) 3C recorder. The signal response shown at the lower left is much more uniform, and the crosstalk is greatly reduced. The same information for the right channel is shown on the right-hand side of the figure. Due to an operating error, the order of the low-frequency recordings is reversed for the 3C.

While both sets of photographs give only qualitative information on the amount of crosstalk present at the high frequencies for both 3B and 3C recorders, actual measurements by the Buchmann and Meyer method indicated crosstalk had been reduced about 6 db over the entire spectrum, thus confirming the photographic evidence of the dramatic improvement in crosstalk performance of the (modified) 3C recorder. The end result was a remarkable confirmation of the computer prognosis.

GROOVE DEPTH CONTROL

The practice of altering the pitch (number of lines per inch) in standard lateral recording is now universally used as a means of obtaining maximum playing time on a phonograph record. The change in pitch is usually effected by a servomotor, which is controlled by the rectified and amplified signal from an added magnetic head spaced about 15 in. ahead of the normal reproduce head in the tape machine supplying stereo signals to the disc recorder. The advent of stereo recording involving vertical as well as lateral movement of the cutting stylus demands control of depth of cut as well as pitch in order to take full advantage of the latter.

Figure 10 has been prepared to illustrate the relationship which exists between pitch and optimum groove width for 45–45 disc recording. It was prepared by assuming a minimum groove width of 1 mil, which is theoretically permissible with a reproducing stylus radius of 0.7 mil. Assuming also that adjacent grooves just touch at maximum amplitude, and that horizontal and vertical components are equal we have,

$$W_{av} = (W_{min} + W_{max})/2 = (1 + W_{max})/2.$$

But for a fixed value of P (number of lines per inch) the maximum possible groove width is given by $W_{max} = 1000/P$.

Therefore

$$W_{av} = (1 + 1000/P)/2,$$

where W_{min} = minimum groove width, W_{max} = maximum groove width, and W_{av} = average groove width.

The Westrex RA-1630-A Depth Control provides the relationship given in Fig. 10 with provision for manual control to provide a range of control of minimum groove width of $\pm$ 0.7 mil at 400 lines per inch. The depth control unit mounted on the recorder (see Fig. 11) contains a solenoid which is actuated by the current from a specially designed power supply, the value of which is controlled by a special potentiometer mounted on the pitch control unit of the Scully recorder. Thus, any change in pitch control automatically alters the solenoid current which in turn alters the setting of the recorder advance ball assembly.

In setting up the recorder, the pitch control on the Scully

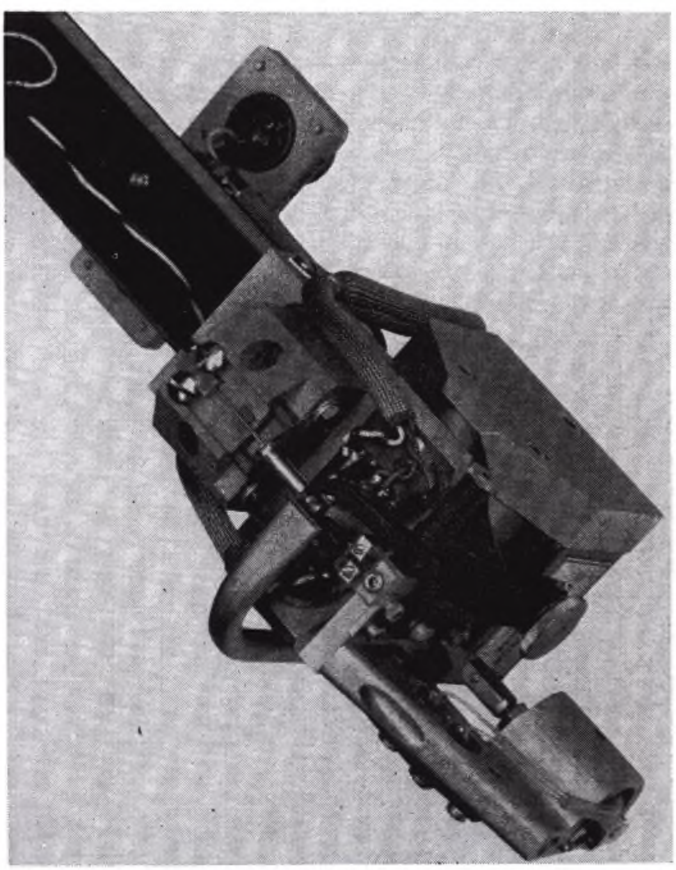
FIG. 11. Bottom view of 3C StereoDisk recorder equipped with depth control solenoid.

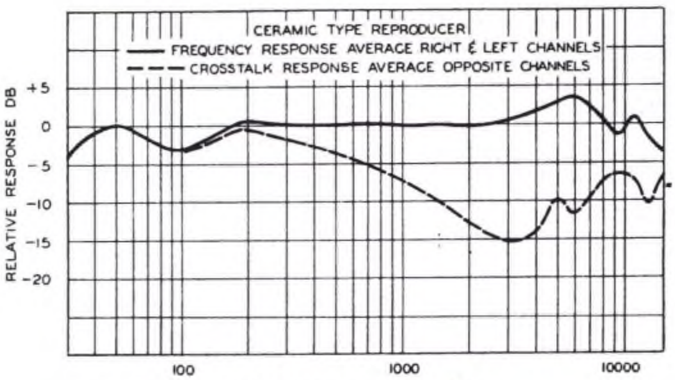

FIG. 12. Frequency response of typical ceramic-type reproducer.

recorder is set at 100 lines per inch which results in maximum current through the solenoid, and the advance ball assembly is adjusted to give a groove width of between 5.5 and 5.7 mils. The pitch control is then set at 400 lines per inch and a resistance in the power supply is adjusted to provide the required minimum groove width which is nominally 1.7 mils to meet the characterstic of Fig. 10. It should be noted that between 100 and 200 lines per inch, the adjustable resistor in the power supply has little effect since the pitch control rheostat is in parallel with it and at a low value of resistance. It should be mentioned that the attack and release times of the depth control are extremely fast and in the order of 25 msec for both attack and release.

A proposed alternate method of groove control is to use the difference signal of the two stereo tracks on the tape to provide control of the depth of cut and use the sum of the signals to provide pitch control. In the simple method as described, one magnetic head scans both stereo tracks on the tape and reproduces the sum of the signals which in turn actuates both pitch and depth controls.

REPRODUCERS

A paper on recent developments in stereo disc recording would not be complete without reference to some progress

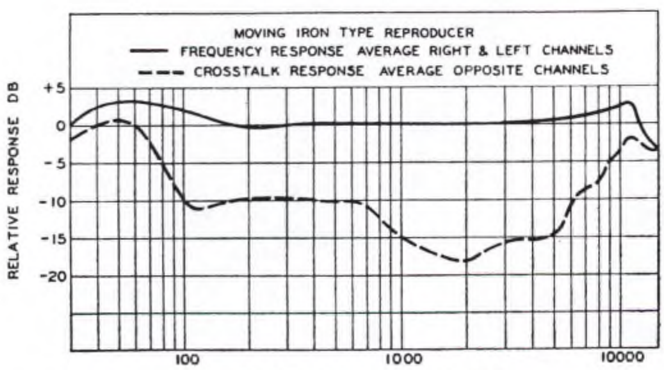

FIG. 13. Frequency response of typical variable-reluctance-type reproducer (moving iron).

in stereo disc reproducers. Since the original paper by Davis and Frayne,[1] many types have reached the market, embodying, as they do, every principle previously used in standard pickups. Due to failure to set any standard for vertical tracking angle, the result has been that there is a wide variation in all designs now on the market. It will be noted that the vertical cutting angle of the Westrex stereo record is 23 deg from a true vertical, and failure to realize the same angle in the reproducer may result in serious harmonic distortion.[1]

An investigation of the various stereophonic reproducers now on the general market showed an average of 17 deg for the vertical tracking angle. This angle was measured while setting the mounting surface of the cartridge parallel to the imaginary surface of a record, but other factors have to be considered in evaluating the tracking angle which would be obtained in practice. As an example, the cartridge might be oriented with the mounting surface not parallel to the disc surface but at some angle in the order of 3–5 deg so that adequate clearance would be provided between the cartridge and the disc surface. On record changers, this clearance would have to prevail with a full stack of records, and the average angle between the cartridge mounting surface and the disc would be still greater by 3 or 4 deg. The effective tracking angle would be greater, therefore, by vari-

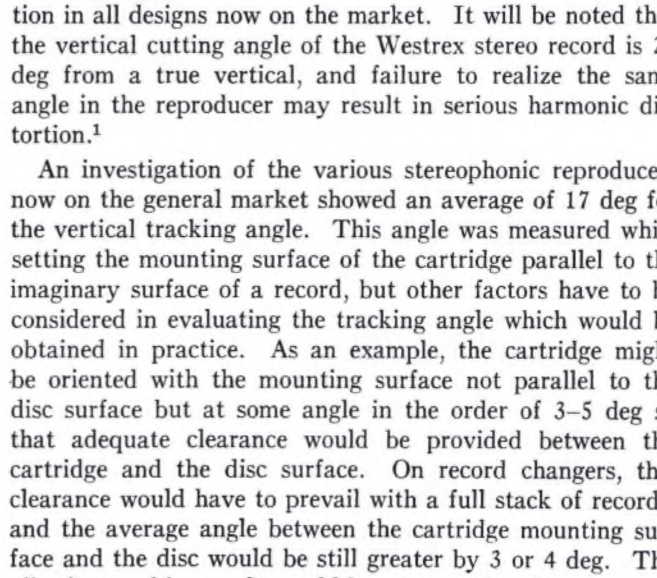

FIG. 14. Frequency response of typical-variable-reluctance-type reproducer (moving magnet).

ous amounts up to perhaps 8 deg than the average figure quoted above, say, up to a total of about 23 deg. Except for this feature, design of commercial pickups seems to be in general agreement with the EIA specifications for stylus radius, compliance, vertical force on the groove, and effective mass.

The performance of commercial pickups vary widely depending on the design selected. Figure 12 shows the response characteristics of a ceramic-type pickup. The frequency response is reasonably uniform from 30 cy to 15 kc. However, the amount of crosstalk from one channel to the opposite as indicated by the broken curve is very excessive up to approximately 1 kc and is never less than 15 db below the actual signal on the other channel. Figure 13 shows the same characteristics for a moving iron (variable reluctance) type pickup. The frequency characteristic is quite satisfactory from 30 cy to 15 kc. The crosstalk is fairly acceptable at 1–5 kc but measures only – 10 db with respect to the signal from 100 to 700 cy. Figure 14 shows the frequency and crosstalk performance of a moving magnet type pickup. This pickup shows the least crosstalk of any of the various types tested amounting to approximately – 20 db over a wide audio band. All three types exhibit poor crosstalk performance in the upper audio region, but this does not appear to interfere with good stereo listening. The presence of high-level crosstalk at very low frequencies also does not appear to impair the stereo illusion.

CONCLUSIONS

The acceptance of stereo disc recording by the public has been nothing short of phenomenal considering the short interval elapsing since its limited introduction a little over a year ago. The stage for stereo had undoubtedly been set by the pioneers in stereo tape making the introduction of the stereo disc inevitable to meet a pent-up demand for this type of reproduction in the home. The cooperation of recording equipment suppliers, record companies, and phonograph manufacturers in setting up standards at an early date on an international basis was perhaps the major factor in bringing this new method of recording so quickly into homes all around the world.

THE AUTHORS


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Robert R. Davis

Dr. John G. Frayne has been associated with the motion picture and recording industries for over 30 years, having worked with Electrical Research Products and its successor, the Westrex Corporation, during that period. He holds a Ph.D. degree from the University of Minnesota, and after teaching at different universities and colleges for a number of years, he came to Cal. Tech. as a National Research Fellow in 1928. The following year he joined the above company. He was associated with the developments in optical recording, such as noise reduction, push-pull recording, intermodulation test techniques, and later he was active in the development of the Westrex magnetic recording systems for motion pictures. He has been active with the SMPTE for a number of years, having served as President of that Society in the years 1955–56.

Mr. Robert R. Davis has been with Westrex Corporation for the past five years, during which time he has been active in the design of stereophonic disc reproducers and recorders. Previous employment with Consolidated Electrodynamics Corporation of Pasadena, and Beckman Instruments, Inc., of Fullerton, California, has given him a background in the design of scientific electromechanical instruments. He holds a B.S. degree in applied physics from California Institute of Technology. He is an Associate Member of the SMPTE and a member of the AES.

Practical Aspects of High-Fidelity Disk Recording. Part 1

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Several aspects of mechanical recording are analyzed, based on the author's observations, who, while dealing with the subject, has avoided as much as possible the use of electromechanical resources to control the effects.

The collection of topics presented does not cover the whole subject but merely represents one way among several of how to achieve good results in disk recording.

INTRODUCTION

THE development of certain branches of sound recording and reproduction has been fantastic, while in other branches development has remained almost stationary since the introduction of the electrical recording process.

The gap in this development is the mechanical recording process so well known to all disk-recording people, but seldom clearly understood.

In every recording company the problems resulting from this gap are faced daily, and the solutions generally applied are a product of experience acquired through many years of practice.

MECHANICAL RECORDING

Mechanical recording is one form of registration made by the removal of a quantity of material from a soft base in accordance with the produced movement. In the case of disk recording, the producer of the movement is the cutterhead, the movement is then transferred to the record by the stylus, that is the transference agent, and the record is the base where the movement is registered.

Several types of excellent cutterheads are known now, which work on a great variety of principles, and as the electromechanical transference is not the important point, the transference of mechanical motion will be discussed.

SHANKS AND FITTINGS

In order to discuss mechanical transference it is necessary to analyze the several types of shanks existing in the styli known at present. Figure 1 shows some of the types of styli in use, which differ only in shanks:

Type *A*—Common shank for cutterheads like Grampian, Presto, RCA, Audax, Cook, etc.

Type *B*—Special shank for Westrex cutterheads.

Type *C*—Special shank for Ortofon cutterheads.

Type *D*—Not represented in Fig. 1, but it is similar to type *B* and is intended for use with Neumann cutterheads and the only difference is the addition of a small rim close to gem.

In Fig. 1, a comparison is also made with the popular long shank cold cutting type. Types *B*, *C*, and *D* were created to overcome some defects of type *A* (see below).

Type A—Defects and Corrections

Figure 2(*A*) shows the type *A* stylus fitted in a cutterhead. Point *O* is the point where the shank is secured to the styli holder by means of a screw. When not subjected to the driving force of the revolving blank, the shank touches point *K* but when driving force *F* is applied, the shank leaves point *K* and approaches point *L* without touching it, as shown by the dotted lines.

When this happens, the shank transforms itself into a bar clamped at only one end and introduces a transverse vibration of a highly resonating character that greatly distorts the sound waves fed to the stylus.

Figure 2(*B*) is a simplification of the modes of vibration with the maximum excursions determined by an ellipse cut by line *AB* representing the center of the groove.

Another form of vibration happens with type *A* styli, which is produced by an imperfect union between aluminum shank and jewel tip. The modes of these fundamental vibrations in the joint [point *P* Fig. 2(*A*)] are represented in Fig. 2(*C*), and the maximum excursions are represented by the circle between lines *A* and *B*.

Adding the two different modes of vibrations [see Fig. 2(*D*)] that represent maximum deviations from point *Q*, which would be the point where the stylus tip should normally be, we get an ellipse in which the excursions of tip are greater than the sound modulation applied to the stylus, sometimes producing distortion figures of 100%.

Comparing the typical sine wave represented in Fig. 2(*E*) with the pattern represented in Fig. 2(*D*), we can make a

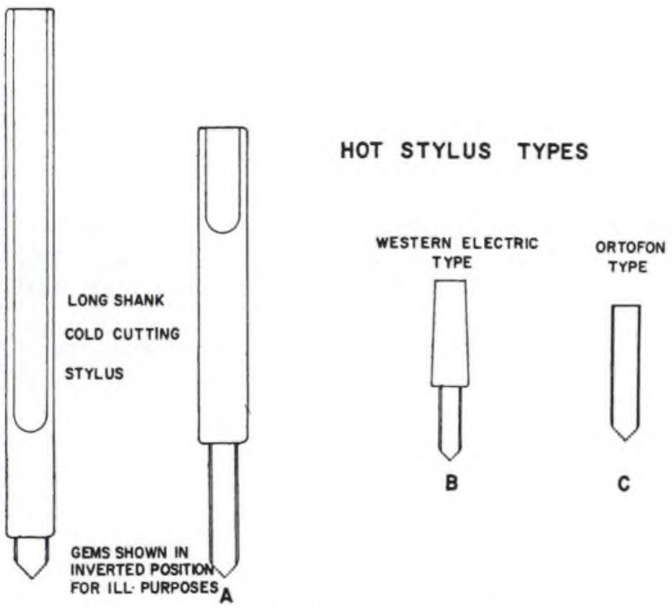


FIG. 1. Recording stylus types.

comparison of amplitudes between distortion and modulation.

The first procedure adopted to reduce the amplitudes of the detrimental vibrations was to reduce the total length of the stylus by 22% or more. This represents reducing a ¾-in. stylus by approximately 5/32 in. and can be done without any serious harm in practically all recorders with a slight readjustment of the angle of cut.

Referring again to Fig. 1, the cold type of cutting stylus can be compared with the shorter type *A*. This procedure will somewhat reduce the efficiency of the transference of the sound to the blank, but the improvement of quality and reduction of cutterhead resonances will be quite noticeable.

Figure 3 shows what can be done from reducing the

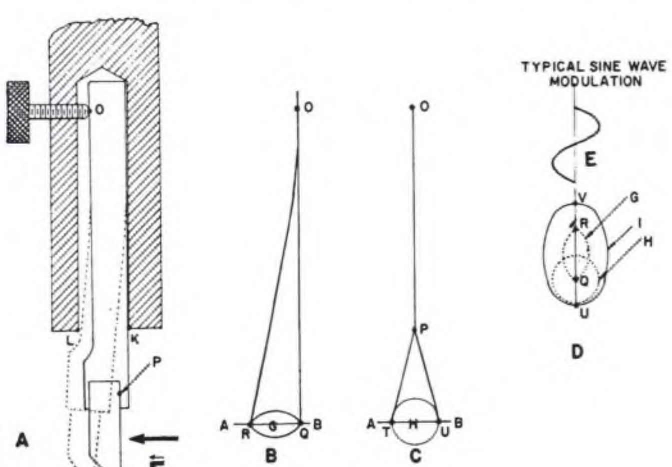


FIG. 2. Distortion produced by shank holder and misfit of gem and shank.

length. First, the length of the chamfer was reduced to the absolute minimum so that only point *O* where clamping is made departs from a cylindrical surface. This cylindrical surface is of very great importance, because when force *F* is applied, the cylindrical surface of the shank touches the inner cylinder of the stylus holder at point *L* and adheres to it because force *F* is constant and tends to increase slightly as the recording reaches the center of the blank [see Fig. 3(*B*)].

Reduction of the second mode of detrimental vibrations was made by increasing the length of the gem to more than two-thirds of the total length of the stylus, which was made possible by the reduction of the chamfer and by changing the usual process of securing the gem to the aluminum or brass shank by pressure in favor of the bonding process.

Several types of special glues are available that can join two pieces of steel and withstand severe stresses for almost infinite periods of time. A way of securing the gem tightly

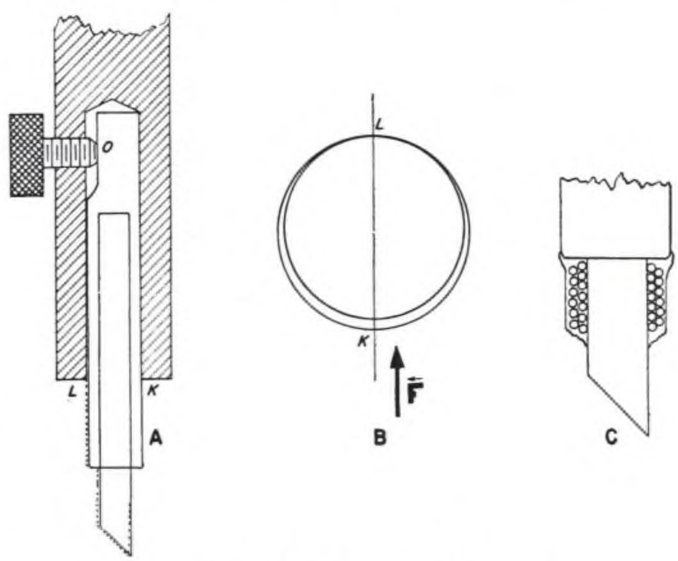


FIG. 3. Correction methods.

to shank consists in carefully cementing the heating resistance with zinc-oxide cement (SS White dental cement) as described in a previous paper.[1] Figure 3(*C*) shows this reinforcement.

With the above precautions, and with a firm tightening of the screw, at least 98% of the detrimental vibrations can be eliminated.

Type B—Defects and Corrections

Referring back to Fig. 1, we can analyze the method adopted by Westrex in manufacturing their recent cutterheads, and Neumann in Germany who followed the same idea of securing the stylus.

First, they reduced the length of the shank to a minimum,

[1] C. E. R. A. Moura, J. Audio Eng. Soc. 5, 90 (1957).

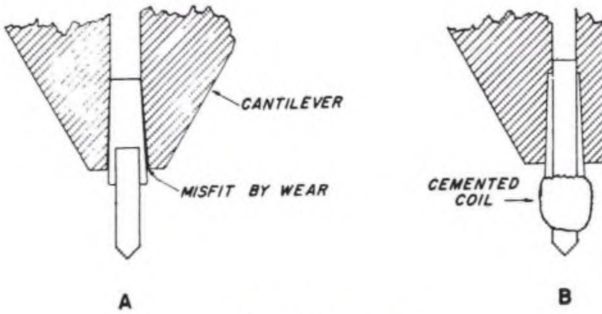


FIG. 4. WE system.

but maintained approximately the same distance from the cutterhead to the blank as with the type *A* styli. To avoid vibrations caused by misfitting of the shank to cantilever they adopted a conical hole and a tapered shank. Theoretically this is a perfect solution, but results indicate that it is not the best solution that could be obtained.

In type *B* the cantilever is made by casting. Great hardnesses of material are not obtained by this forming method, so the tapered hole is subjected to rapid wear, thus departing from the original dimensions in short periods of time. With type *D* this problem is somewhat attenuated because the cantilever is made of tempered steel. From the mechanical point of view, the production of a tapered hole is quite difficult, and with respect to the dimensions of the tapered hole discussed, the use of the grinding process is impracticable, so we are forced to admit that the conical hole will be somewhat imperfect. Production of conical shanks is also a problem, tools are very small, thus showing rapid effects of wear. Figure 4(*A*) shows the normal misfit in type *B* after 500 changes.

Suppose that a 100% fit is obtained in this system, then normally, a new head and a new stylus should provide a perfect fit. Two other forms of spurious vibrations can be found in this system, the first, similar in cause and effect with that which occurs in type *A*, can be attenuated in a similar manner [see Fig. 4(*B*)]. The second is the vibration of the gem itself, which is considerably thinner than the one used with type *A*, 0.5 mm for type *B* and 1 mm for type *A*. This effect can be reduced by the addition of mass and cementing the resistance.

As this system utilizes no mechanical reference in the shank, there are no precise means of adjusting the mirror facet of the gem into an exact 90° angle to the groove even with the aid of special tools. Another drawback in this system is the difference in length from stylus to stylus.

Type C—Description and Suggestions

Type *C* styli were developed by the Lyrec-Fonofilm Industries in Denmark for its Ortofon cutterheads, and in some aspects it represents the finest that could be done in securing a stylus.

The total length of the stylus is 4.5 mm (about 0.18 in.) less than one-third of the small type *A* and requires the use of tweezers in handling. Its major characteristic is the total absence of shank.

The stylus is secured in a pressure mandrel clamped around the body of the gem. Figure 5(*A*) shows a partial view of this mandrel.

However, sometimes this system can produce minor vibrations, as shown in Fig. 5(*A*), due to only one point of contact and can be definitively solved by using a rounded end stylus [see Fig. 5(*B*)], thus providing another good point of contact.

Incidentally, a rounded end gem is easier to grind than a 90° flat end, and it provides better uniformity in the length from one piece to another.

The diameter of the gem is 0.75 mm, and the operating distance is smaller than the preceding types (around 0.10 in. from cutterhead to blank) thus showing a reduced tendency for self-vibration. The only serious drawback in this type of stylus is the same problem of adjusting the mirror surface to groove as with *B*.

THE STYLUS

The stylus as a cutting tool is analogous to the cutting bit used for turning, boring, and other similar mechanical operations.

This small piece of precious stone, sometimes smaller than a grain of rice, is probably the most important part in disk recording.

There are thousands of pages written in mechanical manuals about cutting bits, and every detail of the construction is explained, giving reasons for the difference of the types required. Steel and aluminum cannot be cut at the same efficiency with the same cutting bit.

The terminology of shank, base, heel, face, point, cutting edge, nose, shape, flank, neck, flat, chip breaker, shank angle, back-rake angle, side-rake angle, relief angle, clearance angle, etc., are words known by heart by every good lathe operator. However, almost all recording lathe operators and audio engineers know practically nothing about the

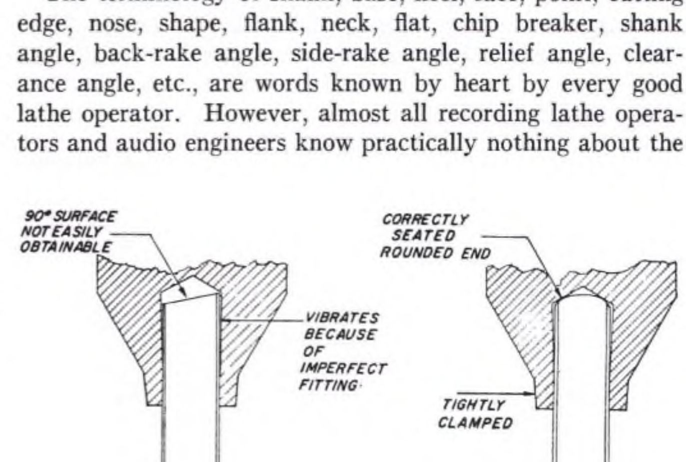


FIG. 5. Ortofon correction.

stylus. The following paragraphs explain the elements of the stylus and how changes in these elements affect its performance.

Material

Different types of precious stones can be used for professional styli: natural or synthetic sapphires, ruby, diamond, aquamarine, and others.

Manufacturing

When natural stones are used, cutting and turning operations are necessary to transform the shape of the stone into one suitable for the faceting operations. With synthetic stones, considerably less work is involved because the stones are cast directly into cylindrical shapes.

Beginning with these cylindrical shapes (Fig. 6 *S1*), one big facet is ground in the cylinder and covers its entire length (Fig. 6, *S2*). This facet is called the "mirror" and is the largest facet of the stylus (*F1* in Fig. 6, *S2*).

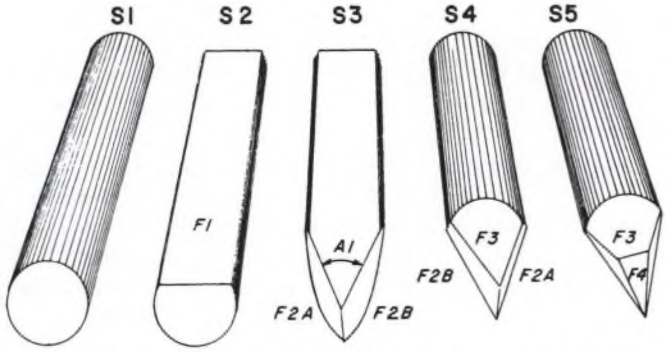


Fig. 6. First steps in the lapidation of a recording stylus (faceting).

Next (Fig. 6, *S3*), two more facets are ground in one extremity of the cylinder, *F2A* and *F2B*, and receive no particular terminology because they are only of constructional interest. At this point, the cylinder is divided in two parts for nomenclature purposes, the extremity with *F2A* and *F2B* is called the "vertex" and the remaining part "fulcrum" or "body."

The angle formed by *F2A* and *F2B* in the mirror facet is called "included angle" by Americans, while we prefer to call it the "reluctance angle."

Turning the cylinder 180°, so that the mirror is away from us, another facet, *F3*, is ground (Fig. 6, *S4*). In *S5*, *F4* is ground. These facets receive no particular names because they will be reduced to simple lines in final polishing, and in some cases, facet *F4* will be nonexistent.

In Fig. 7 (*S6*), four more facets are polished. *F5A* and *F5B* are called "major facets," and *F6A* and *F6B* are called "minor facets," and in some cases, as pointed out before, may be nonexistent. Here, *S7* represents *S6* with a torsion of 90°.

The next step is the polishing of the tip of the vertex

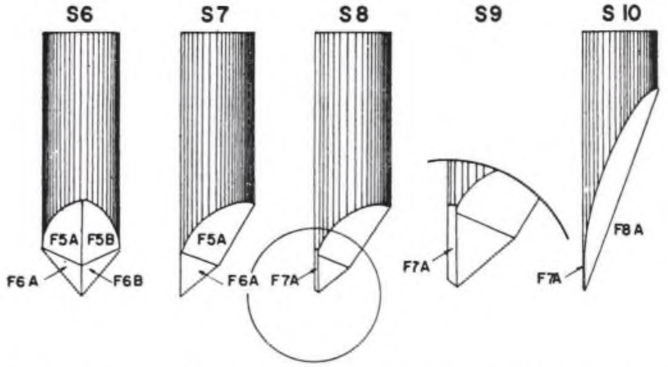


Fig. 7. Steps in the lapidation of a recording stylus (fine polishing).

into an arc of a circle, called "tip radius," and at this point we have a complete stylus suitable for wax recording or even acetate recording. This type of stylus is called "feather edge." Imperfections in the cutting edges of the reluctance angle make this stylus impractical to manufacture and use, so additional steps of very fine polishing are required to correct these imperfections. Frank L. Capps was first to perform this extremely fine polishing operation in the recording stylus, and the process consists in grinding a very thin facet along the cutting edges of the reluctance angle (Fig. 7, *S8*, *S9*) which is called the "burnishing" or "polishing" facet.

Knowledge of the angles and edges formed in the faceting operations described is of great importance, because substantial improvements in sensibility, frequency response, noise, damping, and control of distortion result from simple alterations made in this aspect.

Figure 8 shows the nomenclature and correct locations of the edges of a recording stylus. In order of importance, they are: (1) Cutting edges, actually cut the walls of groove. (2) Polishing edges, give final polishing to the walls of groove. (3) Section, cuts and polishes the bottom of groove.

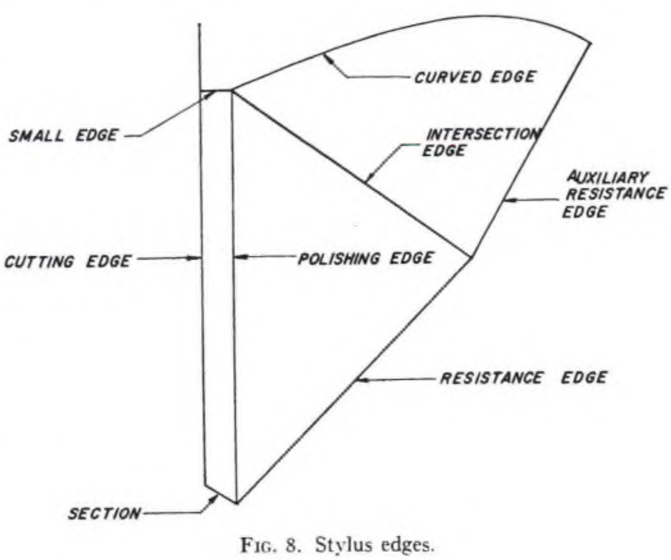


Fig. 8. Stylus edges.

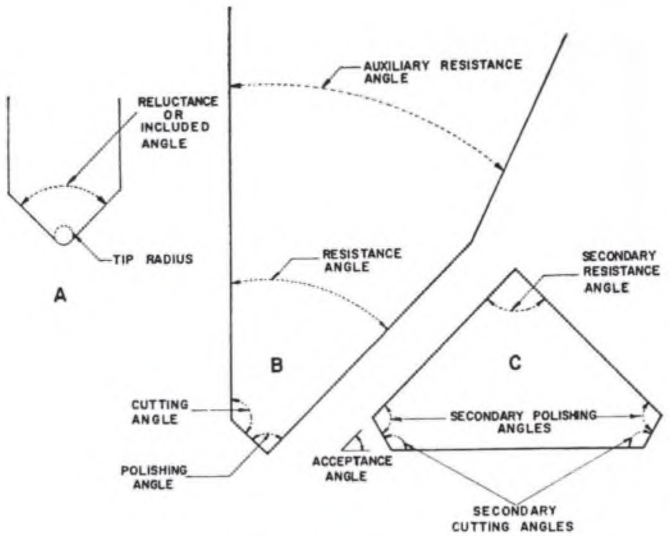


FIG. 9. Stylus angles.

(4) Resistance edge, interferes only occasionally in the cutting action of the stylus. (5) to (8) Auxiliary resistance edge, intersection edge, curved edges, and small edges are of constructional interest and do not interfere in the performance of the stylus.

Figures 9(*A*), 9(*B*), and 9(*C*) show the angles existent in a recording stylus, and these angles determine the dimensions of the facets and the characteristics of its performance.

The reluctance angle [Fig. 9(*A*)], also called "included angle," is the angle that determines the shape of the groove. It can be made from 70 to around 95°. European preference is for 70 to 80° styli, and American preference is from 87 to 95°.

Many factors are involved in the determination of a correct reluctance angle and can be solved only by a compromise setting. Open angles (85 to 95°) make the reproducing stylus track the groove nearer its bottom, thus reducing the possibility of tracking small scratches of the surface. Open angles also make the separation of the several galvanoplastic parts easier.

However, acute angles (85 to 70°) reduce acetate damping and produce a more resistant chip making it easier to use high heating temperatures, thus resulting in less noise. Thus, open angles are better for processing, and acute angles better for recording. For monophonic and stereophonic, 80° is best.

With stereophonic, especially in the 45/45 system, some might think it surprising that the 80° stylus would provide much better results than the 90° one. The reason is that with the 90° stylus, when a sudden transient is applied in only one channel, as the burnishing facet is positioned at 45°, the force will be distributed equally in the area of the facet, thus providing a very high load to the stylus that will damp and distort the transient. With the 80° stylus, the force will not be distributed equally, so the stylus will cut by slicing action and not by shearing action. However, in very short periods of time shearing action will occur. Tracking of the reproducing tip will be the same for both types.

The tip radius [Fig. 9(*A*)] is actually not an angle; it determines the width of the bottom of the groove, larger radii used for normal grooves (now almost abnormal), smaller radii for microgroove recording, and the smallest for sub-microgroove and stereo work.

The first important point of tip radius is that it must be at least half the radius of the reproducing tip. Type *V* groove recording is made by using a stylus with a tip radius many times smaller than normal in order to allow also for the tracking of the groove by reproducing tips of smaller diameters. For instance, a record with type *V* groove of normal width allows for reproduction with a standard tip and also with a microgroove reproducing tip.

A smaller tip radius also reduces damping of the blank, reducing consequently the load on the cutterhead resulting in less distortion in the registration of sound. The findings of C. E. Watts are in strict accordance with the author's.

The compromise that should be made in respect to this point is to find which is the lowest tip radius that will allow a good mechanical resistance of the tip and a trouble-free galvanoplastic processing.

From the recording aspect of the question, the smaller the better. Tip radius normal figures are at maximum 0.002 in. and can go to 0.0001 in. or even less.

The cutting angle is the one formed by the mirror and burnishing facet. Its point of measurement is made at the point where the mirror touches the section. At this point the section splits this angle and forms the secondary cutting angles [see Figs. 9(*B*) and 9(*C*)].

The measurement of this angle can go from 100 to 127°.

The primary function of this angle is cutting, but it also controls the polishing to some extent, because its dimensions directly affect the dimensions of the polishing angle. Several factors such as mechanical resistance, frequency response, and noise affect the correct determination of this angle; in a stylus with the mirror arranged at a plane exactly 90° from the surface of the blank, a good point of compromise is around 120-125°.

The polishing angle is formed by the section and the resistance edge, and its dimensions are subordinated to the dimensions of the cutting angle and the resistance angle. Its measurement is made in the following manner: A stylus with a cutting angle of 125° and a resistance angle of 50° would be as follows:

$$180^\circ - 125^\circ = 55^\circ,\ 55^\circ + 50^\circ = 105^\circ.$$

From these simple calculations we deduce that the polishing angle of such a stylus is 105°. The secondary polishing angles differ somewhat from these dimensions, but the knowledge of the exact figures is unimportant.

The resistance angle is the one formed by the mirror surface and the resistance edge, and as the name implies it controls the mechanical resistance of the tip against shocks but not against abrasion.

Normally, the maximum dimension of this angle is 60° and the minimum as close as possible to 0°. It is the dimension of this angle and the size of the burnishing facet that separates a good stylus from a poor one.

The resistance angle controls the top limit of the frequency engraving capacity of the stylus, the efficiency, and the ability of the stylus to maintain the frequency with the reduction of the diameter of the groove spiral. The optimum point of compromise will be explained later in detail.

The auxiliary resistance angle is only of constructional interest, and in styli with resistance angles smaller than 30° this angle is nonexistent.

The incidence or acceptance angle is the one formed by the mirror and the major facets, and its measurement is a consequence of the size of the resistance angle [see Fig. 9(*C*)]. The acceptance angle controls the acceptance effect.

Acceptance is the effect noticed when the same recording is made in two different disk speeds, the recording at the highest speed will sound better because it will show the greatest acceptancy of the material for the modulation. Of course, a better registration for the highest speed in disk recording will also be a consequence in the reduction of the tracing distortion in reproduction.

In a recording stylus, each angle is dependent on another; so, any alteration in an angle will alter the complete design of the stylus. The most important point is to achieve a perfect harmony in this design.

CHIP REMOVAL

Another important chapter in disk recording is the removal of the material cut by the stylus to form the groove. Several nonprofessional methods involve the use of brushes, and one professional method, now seldom used, involved the cutting of the grooves from the inside of the blank to the outside, thus throwing the chip to the center of the record and avoiding the accumulation of the chip in the path of the stylus. The universally adopted method actually involves the use of suction devices to avoid possibilities of accidents with the chip and other detrimental effects as will be related.

Almost any mechanical device that can produce air suction can be utilized for the purpose. The most elementary and easiest is to use a common household vacuum cleaner coupled to a nozzle somewhere around the stylus. More elaborate systems use reverse blowers (RCA), bellows (Van Epps), reversed air compressors, etc.

All these systems must include some means of filtering in order to avoid the highly flammable chip reaching the mechanical parts that are producing the suction. These filters may be of the net type (a cloth intercepting the chip) which is very inefficient, the water type, also inefficient, and the simple and highly efficient gravity type, where the chip is separated from the air current by its own weight. The type of device that produces the highest velocity and steadiest suction will be the one that gives the best results.

At this point it is convenient to explain the necessity of a constant suction. Four forces are involved in mechanical recording: first is the revolving action of the turntable which must be perfectly constant; second is the spiral feed force which must be uniform or uniformly varied in the case of variable pitch; third is the motion of the stylus produced by the cutterhead; and fourth is the suction applied to remove the chip.

In order to avoid any perturbation to the movement of the stylus, all the remaining forces must be uniform. Inconstancy of the revolving action of the turntable is a worry of the past, and feed pitch very seldom presents problems.

Only in rare instances are suction devices integral parts of the recording lathes; so the adaptation of this accessory must be performed by the user, and this adaptation is of great importance in the performance of the whole system.

Since the suction force is applied directly to the tip of the stylus and its magnitude is greater than the force produced by the cutterhead, it is logical to think of an interaction of the two forces, which will result in distortion when modulation is present, and in noise in the absence of modulation.

In order to reduce the interaction of these forces, the first and most important point to be studied is how to combine these forces. Figure 10(*A*) shows that it is possible to use a suction nozzle anywhere around the stylus in a circle of ¼ in. to 2 in. In some recorders, such as RCA 73B, Presto 6N and 8DG the suction nozzle is located at 90° to the recording surface, as shown in Fig. 10(*B*), with the suction nozzle having a trumpet-flared circular entrance.

The great inconvenience of this type of suction tube is the great friction point at *W* [Fig. 10(*B*)]. To reduce this friction point, different types of nozzles were designed. Figure 10(*C*) shows one that proved more efficient, and it is based in picking up the chip at almost horizontal angles and bending the tube at a gradual rate to allow the chip to be driven out vertically. Figure 10(*D*) shows the modifications made at the entrance of the nozzle.

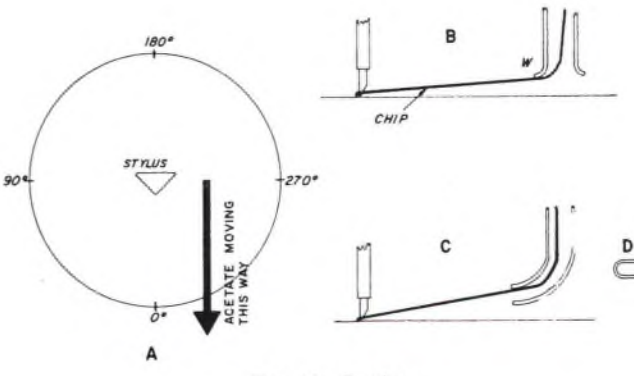


Fig. 10. Suction.

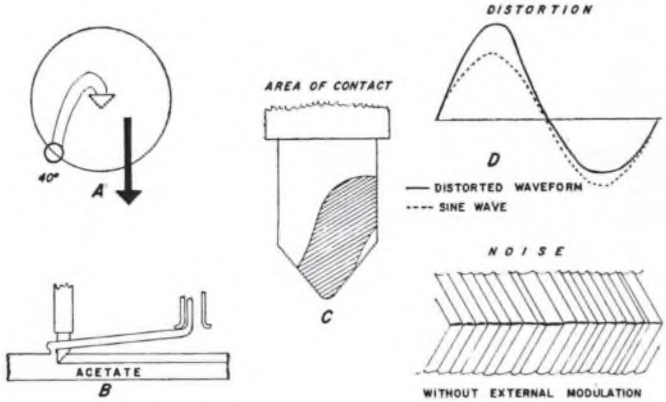


FIG. 11. Front suction.

Front Suction

Figure 11(A) shows the adopted location for front suction in the circle of Fig. 10(A); the nozzle is positioned at approximately 40° from the groove that is being cut. The path made by the chip is shown at Fig. 11(B), and in order to follow this path the chip touches the mirror surface of the stylus in several points, as shown in the shaded area of Fig. 11(C).

This area of contact extends beyond the area bound by the reluctance angle and enters the region of the fulcrum where the heating resistance is wound [see pattern on Fig. 11(C)], thus placing the chip in direct contact with the high heat resistance. Frequent losses of acetates by chip burning are quite common.

A close analysis of Fig. 11(C) also reveals that the contact points are located almost totally in the right side of the mirror, so there is a dragging force actuating the stylus tip, making difficult the movement of the stylus in the right half-cycles, and enhancing the left side ones. Numerous tests made to obtain distortion figures gave a minimum of 8% harmonic distortion with the sine waves at an operating condition where the suction was barely sufficient to remove the chip out of the recording acetate. Distortion figures increased in a somewhat linear correlation with the increase of suction velocity. Figure 11(D) compares the waveform produced by this system with a typical sine wave.

Another detriment of this system is the slight bending of the stylus tip made purposely by some styli manufacturers to help the output of the chip toward the center of the record. This bending is from 2 to 5° and produces a minimum of 3% harmonic distortion and a higher percentage of intermodulation. Another incorrect practice is to advance the cutterhead ⅛ in. or so to facilitate the removal of the chip.

In order to avoid distortion the stylus must not be twisted, and the recording should coincide exactly with the radius of the blank.

Regarding noise, the drag on the right side produces vibrations of a very high frequency order, and a microscopic examination would reveal grooves highly indented, similar to Fig. 11(E). Best noise figures obtainable with front suction are around 40 to 45 db below maximum signal.

From the facts explained above, the rule that can be outlined for best results with front suction is to use the least suction velocity possible to lower distortion and noise, and to reduce the possibility of burned chips. Of course, this operation involves the use of moderate heating, which limits the usefulness of the system to fresh blanks.

Rear Suction

Noise figures of minus 40-45 db are insufficient for modern LP and stereo recording techniques, and with front suction any further attempt to reduce these figures will be unsuccessful. Experiments were made to relocate the suction nozzle in another collecting angle, and more sensible results were obtained when the angle was increased from 40 to 90°, but even better figures were observed when the angle was increased beyond that point.

At this stage of research, the small torsion of the tip began to create problems in catching the chip immediately after the cutterhead was lowered to the blank. This torsion was eliminated, and the catching of the chip became infinitely more easy, requiring only a small puff to reduce the possibilities of failure to less than 1%. Removal of the torsion resulted favorably in an increase of approximately 1 db in over-all sensibility, a gain of 2 db at 15 kc in the center of the record, and clearer high frequencies.

Experiments made with WE cutterheads with suction angles of 150 and 210° provided better results than with 90°. The next step, as everything indicated, was to try the 180° position, but in some machines there was an advance ball system placed exactly in the rear of the stylus, so a complete redesign of this system was necessary to allow a free way for the removal of the chip at exactly 180°.

In some lathes, the advance ball was already located in a more favorable position, while in others advance ball was nonexistent. The nozzle shape was represented in Figs. 10(C) and 10(D).

The very first results showed noise figures of −55 db. Some very striking features were observed, such as the rules formulated for front suction suddenly becoming reversed. The previous rules advocated the minimum of suction to obtain the least noise; the rear suction system accepted any amount of suction velocity and figures showed the higher the better. Apparently no explanation was obtained for this phenomenon, but a close observation revealed the reasons.

Figure 12(A) shows a typical rear suction device, with nozzle positioned at some distance from the blank, and the chip forming with the blank surface an angle of approximately 40°. The pattern of the points of contact of the chip with the mirror surface of the stylus is shown in Fig. 12(B), and comparing this pattern with the one produced by front suction [Fig. 11(C)], we can note a substantial

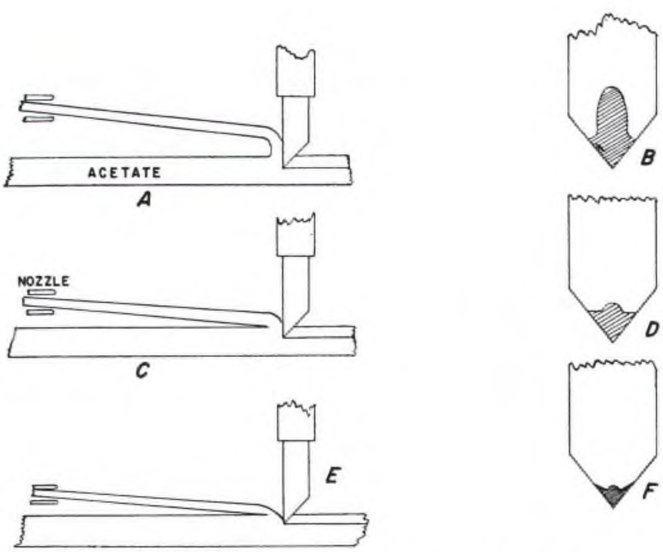


FIG. 12. Rear suction—angles and areas of contact.

reduction of the area of contact, and furthermore, a perfect symmetry of all points in the pattern produced by rear suction. This symmetry indicates that rear suction does not produce dissimilarities in half-cycles of applied modulation. But the area of contact still can be reduced substantially, so the suction nozzle was lowered to the very limit of safety, almost touching the blank surface, in order to reduce the angle formed by the chip and blank to less than 40° [Fig. 12(*C*)].

An angle of 15° was the minimum obtainable and resulted in a substantial reduction of the contact area as shown in Fig. 12(*D*), resulting in lower noise figures. The next step was to increase the suction velocity up to the point where the mechanical strength of the chip could hold, thus the angle was further reduced [Fig. 12(*E*)] and the area of contact reduced to its minimum figure as shown in Fig. 12(*F*). It is interesting to note that this area of contact is smaller than the section of the groove, thus the suction now had an active part in the groove cutting. The suction force was combined in such a way to help the cut, acting in the opposite direction to the rotating force of the turntable. Noise figures of 60 db or even better were obtained with this process.

Charcoal Formation

The excellent noise figures referred to above were, however, good only for the first acetate cut with the same stylus, because subsequent cuts showed an increase in noise figures to –45 db.

Each time the stylus was changed, the first and sometimes the second resulted in excellent low noise cuts, but the remainder was poor. As it is a nonprofitable business to change a stylus for each blank, a solution was sought for this problem. As soon as the stylus increased the noise level, it was removed and subjected to a microscopic examination. Styli examined showed a deposit of charcoal in the mirror surface with an effect similar to that occurring in cutting bits in mechanical cuttings (a deposit of a small layer of cut material on the face of the tool).

The action of cutting with a stylus is not only performed by the cutting edges of the stylus but also by the mirror surface, which forces the material forward while the action of the cutting edges makes the actual separation.

Figure 13(*A*) shows an enlarged pattern of the points of contact with the mirror surface, and it is this area that forces the material in cutting. Due to the conformation of the material to be removed there is a spot in this pattern that is subjected to more efforts than the remaining area, and it is in this same spot that the charcoal formation was observed [see Fig. 13(*B*)].

The texture of the gem material and the finite limit of polishing leave a certain roughness on the surface of the mirror so that a small abrasive action exists. This abrasive action removes minute particles of the acetate. These particles accumulate on the spot, and due to the constant contact with the heated gem the solvents volatize and the nitrate particles are transformed into charcoal particles.

With the transformation of the acetate particles into charcoal, the abrasive action will be greater, and there will be also an accumulation of more acetate particles, thus causing high friction as shown in Fig. 14. The presence of this high friction is the cause of the increased noise previously observed.

It is interesting to note that the deposit of charcoal always occurs in the same spot. Observation on the incidence of this formation revealed that it was dependent on the condition of the blank. The heat utilized and the polishing of the mirror facet were the contributing factors in this effect. Several attempts were made to eliminate this effect, such as lubrication of the tip with silicone, reduction of the heating temperature, and reduction of the suction velocity, but they all reduced the signal-to-noise ratio. Further examination revealed that after the first blank the formation was small and could be easily removed from the mirror surface. The difference in noise level was not greater than 2 or 3 db from the beginning of the face to the end. This difference will

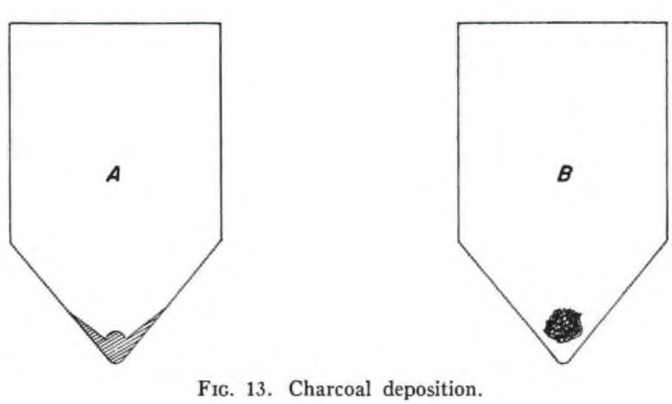


FIG. 13. Charcoal deposition.

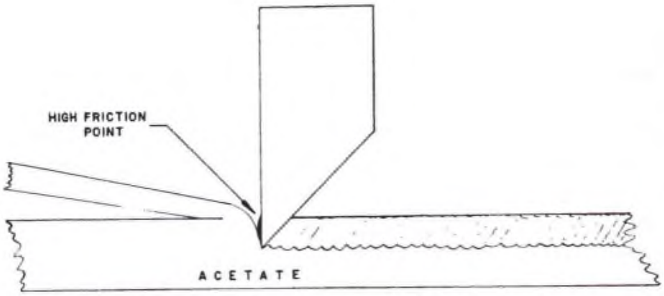


FIG. 14. Noise produced by charcoal deposition.

pass unnoticed and the cleaning of the mirror facet of the stylus can be done with a piece of slightly dampened cotton rolled onto a stick. This cleaning should be performed immediately after the cutting of each face and will be effective in 90% of the cases.

As cleaning does not completely remove the formation, after several cuts the formation will adhere so tightly that its removal will be useless. The only way to remove this adherent charcoal formation is to immerse the stylus in concentrated nitric acid for 15 min and then thoroughly wash the stylus with running water; after that, the tip should be carefully rubbed with a dry cloth.

Aluminum and Dural shanks are not attacked by the concentrated acid, but the *heating resistance* should be removed before the stylus is immersed in the acid. Exposures of more than 15 min should be avoided because of the acid attack on the bonding substance, but when the gem is secured to shank by pressure, longer exposures are allowed.

Extreme care in the handling of acid is mandatory to avoid the possibility of personal injury. Rubber gloves are recommended, and if acid is splashed on the skin it should immediately be washed off with water.

Styli cleaned by the above method can have a very long life. One stylus of the author's lasted 116 cutting hours in perfect condition.

Charcoal formation was also observed with cold cutting, and in this case the previous method of cleaning is 100% effective.

STYLUS AZIMUTH

The effect of azimuth adjustment is well known. Due to the extremely small dimensions of gap or slit, the alignment of the recording and reproducing gaps or slits must be extremely accurate. With tape at 15 ips, for instance, the deviation of a single degree from the correct adjustment causes a drop of 6 to 8 db in the 15-kc response plus the distortion of several per cent in that frequency. With speeds less than 15 ips this problem is crucial.

While studying the cause of the great disparity between Buchmann-Meyer light pattern frequency measurements and playback measurements in disk recording, the author's attention was drawn to the problem of azimuth adjustment (rake angle) in disk recording and reproduction.

Common Methods of Adjustment

The only standard available was an old one, but it still is the foundation stone of the recording industry; that is, 90° plus or minus 3°. This standard is not bad for 78 rpm, but the actual methods used for the adjustment are very erratic so that even this standard is difficult to achieve without an error of several degrees.

The normal method widely recommended consists in the observation of the reflected image of the stylus in the blank, so that when the image formed a straight line with the object the stylus was positioned at 90°. But, if we consider the refraction effect, the astigmatism of the eye, warped blanks, the small size of the exposed part of the stylus, errors of 8, 10, or even 15° will pass unnoticed.

Other methods for this adjustment were sought. Some used the parallelism method; that is, building parallel surfaces to the stylus cantilever, and then applying a drawing triangle or protractor to measure the deviation and perform the required correction. This parallelism method, however, is subject to several errors such as the additive ones and misalignment of the cantilever that may be adjusted to the magnetic center instead of the vertical axis, and thus drive the user to more erratic conclusions than the previous method.

Accurate Adjustment

There is a method that allows the correct adjustment of the stylus azimuth to a tolerance of less than one minute of a degree or, in other words, to the sixtieth part of a degree that, compared with the old standard, represents approximately 500 times better accuracy. Patent is pending for this method. The general idea of this method is to adjust the cantilever to an exact position independent of the other parts of the cutterhead. For this purpose, the stylus is substituted by a steel azimuthal gauge of the same height of the styli used [Fig. 15(*A*)].

The gauge is made up of a steel pin and a steel strip soldered to the pin. The constructional details for the steel pin will be dependent on the type of shank used, it may be cylindrical or conical. The exposed part of this pin is made as a cylinder of larger diameter, about 5/32 or 3/16 in., where a slit is sawed to provide means of securing a 4-in. steel strip.

The milling of surface *S* must be very carefully performed because this surface must be a perfect plane [see Fig. 15(*B*)]. Then the two parts are soldered together in the manner described below.

Gauge Calibration

The best way to perform gauge calibration is to use the turntable and the cutterhead. For turntables that hold the blank by suction a flattened aluminum disk is required. Common turntables should have rubber, felt, or plastic pads removed for this calibration.

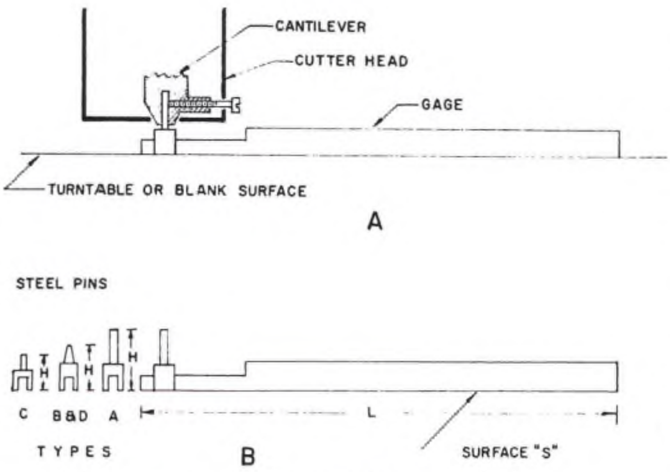


FIG. 15. Azimuthal gauge.

The gauge is then inserted in the cutterhead, with the strip parallel to the mirror plane. The height of the cutter should compensate for the absence of blank or pad. Then the head is lowered, and the strip soldered to the pin in a manner that in this position the surface *S* of the strip leaves no passage for light rays coming from the rear, thus being in perfect contact with the turntable surface.

The soldering can be done with a soldering pencil or gun. After this, the gauge is turned 180°, and the difference of contact of the strip with the surface observed. It is important that in the first soldering the gauge be slightly less than 90°. The gauge should touch the turntable only at the pin or at surface *S* but never in the extreme away from the pin.

Measure carefully the distance *X* (that is, the distance from the end of the strip to the turntable surface) with a block and a micrometer. After this, get a block of half this distance, loosen the screws securing the cutterhead, and tilt the head till it reaches the distance of one-half *X* (Fig. 16); retighten the screws.

Turn back the gauge 180° again, and you will note that the strip will be raised from the surface by approximately ½ *X*. Unsolder the joint, make the steel strip touch the turntable plane, and resolder.

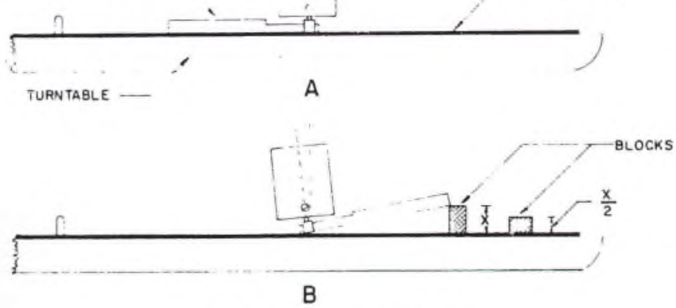
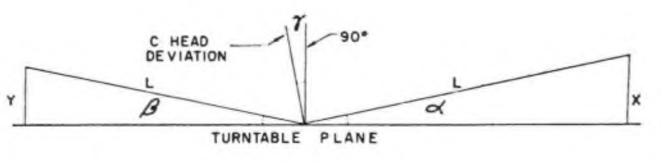


FIG. 16. Gauge calibration.

Repeat these procedures again and again to obtain closer accuracies. Normally, a certain point will be reached where the distance *X* will be so small that difficulty will be experienced in finding blocks sufficiently thin to cover this distance. Thin paper or metallic foil will be a great help in this case.

This method is similar to the one adopted with tape playing it forward and then reversing the tape to reach the optimum azimuth adjustment.

Finally, when a point is reached that can be considered satisfactory, simple calculations, as outlined in Fig. 17, can reveal the exact error. Generally the tolerance of the cantilever adjustment will be smaller than the error of the gauge. Gauges with an angle of 89°, 59 min, 32 sec were constructed with this method.



$\beta = \sin\beta$

$\sin\beta = \frac{Y}{L}$

$\alpha = \sin\alpha$

$\sin\alpha = \frac{X}{L}$

$\gamma = \frac{\alpha - \beta}{2}$

ERROR $= \beta + \gamma$

OR

ERROR $= \alpha - \gamma$

γ = DEVIATION ANGLE OF CUTTER HEAD

β = DEVIATION OF GAGE MINUS HEAD DEVIATION

α = SUM OF HEAD AND GAGE DEVIATIONS

L = LENGHT OF STRIP

Y = LEFT HAND MEASUREMENT

X = RIGHT HAND MEASUREMENT

FIG. 17. Error formulas.

Use of the Gauge

The gauge is not only useful in centering the stylus correctly but also in adjusting the angle of cut. The centering operation is as used for calibration, and adjustment of angle of cut with gauge is used at 90° from the centering positions as indicated in Fig. 18.

Advantages of perfect centering will be a uniform load for the two half-cycles of the modulation, resulting in reducing distortion to a great extent, less tendency for noise production, and permitting the tracking of the highest frequencies by the reproducing tip. In monophonic reproduction it will reduce tracing distortion, and in stereo it will also maintain accurate balance between the two channels and provide maximum separation of channels in the highest frequencies. The only chance of unequal groove walls will remain in defective styli.

For the precise adjustment of the angle of cut (90°), the gauge is inserted in the manner indicated before and the head is lowered to the turntable and the height of the carriage adjusted till the optimum point is reached. This will

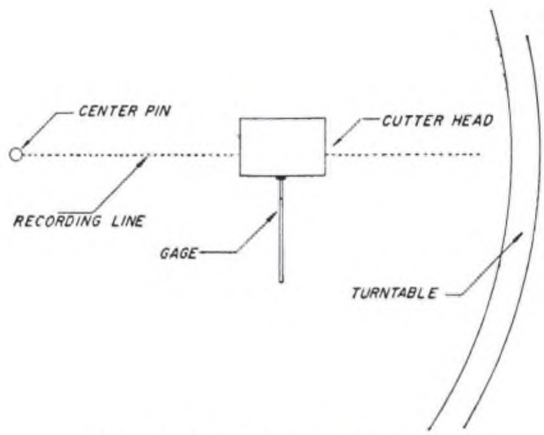


FIG. 18. Angle of cut adjustment.

be the reference calibration point. The reference calibration point must be marked, in some cases by a spacing block marking the position of the carriage adjusting screw or by a radial mark on the screw.

Next, the operating positions are marked, and the thickness of the pad, acetate, and groove depth are computed and subtracted from the calibration mark to allow exact angles of cut in the normal operating conditions. Spacing blocks are efficient in marking the several operating points.

As the gauge has the same size of the stylus, the groove depth should be added to the operational blocks, and in the case of fixed pitch the formula below will allow the calculation of the groove depth based on a 60 to 40 groove-to-land basis.

$$GD = [(0.3/N)/\tan A] - [(R/\sin A) - R]$$

where GD is the groove depth in inches, N is the pitch in lines per inch, R is the tip radius, and A is half the included or reluctance angle.

In the case of variable pitch where the 60 to 40 groove-to-land basis ceases to exist, the same formula can be applied on a basis of 224 lines per inch. The gauge should not be used often: it should be kept as a primary standard, well protected to avoid deviations in the calibration. The operational blocks should be used for ordinary reference, checking from time to time against the gauge.

Effects

Records cut according to these principles will show less marked differences of optical and playback measurements. The radial reduction of the high frequencies will be less, the top frequency limit will be set at a higher frequency, and tape noise that normally was disregarded will be reproduced at the full amplitude thus becoming a nuisance.

Uniformity of cut between blanks will be automatic, the aspect of the cut blank will be black, pure black, not dull or rainbow or excessively brilliant in appearance.

By definition, angle of cut is relative to the position of the stylus in respect to the blank surface, while cutting angle is an integral part of the stylus. It is an error to recommend a change in the angle of cut to avoid vibration effects of the stylus thus forcing the stylus to work at angles that will greatly affect reproduction, while the requirement for such a change is caused by a defective cutting angle of the stylus.

Based on these observations, the region of 120-127° proved to be a good compromise for the cutting angle as explained earlier.

THE AUTHOR

Carlos E. R. A. Moura has worked in disk recording with several companies engaged in manufacturing disk-recording equipment.

He has a patent on tape recording, and several patents pending on disk recording.

The RCA Victor Dynagroove System*

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The RCA Victor DYNAGROOVE system is a comprehensive system of improvements in sound recording by means of disc records. All aspects of the process are taken into consideration, starting with the artist's conception of the music and ending with the reproduction of the sound in the listener's home.

INTRODUCTION

THE RCA Victor DYNAGROOVE system is a planned evolution of improvements in all aspects and elements of sound recording by means of disk records. The main objective has been to provide a distinct improvement in performance from a new cohesive and integrated system. The purpose of this paper is to describe the complete system, from the artist's conception of the music to the reproduction of the sound as perceived by the listener in the home.

A schematic diagram of the complete DYNAGROOVE disk recording and reproducing system is shown in Fig. 1. Every element of the system shown in Fig. 1 was examined by the artist, the scientist-engineer and the artist-scientist team. The main objectives were as follows: To develop the musical plan and the arrangement for the production of the original music. To reduce all types of distortion by the development of new and improved equipment. To develop new equipment and methods for monitoring and recording to facilitate the production of the master and submaster magnetic tapes and the original lacquer disk record. To study the subjective aspects of sound reproduction and apply them to the pickup of sound in the studio and the dispersion of sound in the living room in the home.

In the reproduction of sound in the form of instrumental music and voice the artist and scientist are involved in a program of mass dissemination of information in the form of music. In this connection, the importance of the artist-scientist team concept was already realized several years ago.[1] In the DYNAGROOVE project the artist, scientist and engineer worked as a team[2] to produce the maximum artistic impact upon the listener in the home. From this point of view, the obligation of the recording process is, first, to recreate in the listener's mind a vivid recollection of his experiences in the concert hall while he is listening in his living room, and second, to give him a realistic musical experience with every type of music whether he has had the live experience or not. This is the frame of reference—and it is this set of conditions that is involved if a new recording system is to justify its existence. To bring these conditions about, studies of the music and its rendition by the artist have been carried out by the artist, musician, musical director, recording engineer and scientist.

The collaboration of the artist and the scientist-engineer is depicted in Fig. 1. The artist and scientist-engineer team developed the overall plan, arranged the musical instruments, worked out the placement of the microphones, monitored the reproduced sound in the listening room and recorded the master tape as depicted in Fig. 1*a*. In the next step, as a result of extensive subjective studies and tests, the team developed the Dynamic Spectrum Equilizer. They worked together to monitor and produce the submaster tape as depicted in Fig. 1*b*. During the recording of the original lacquer from the submaster tape as shown in Fig. 1*c*,

* Presented October 17, 1963 at the Fifteenth Annual Fall Convention of the Audio Engineering Society, New York.

† The author's biography appears in the January 1964 issue of the *Journal* on p. 44.

[1] See D. Sarnoff, "New Developments in Electronics," Address at the Annual Winter General Meeting, American Institute of Electrical Engineers, January 31, 1955, New York, New York.

[2] John Pfeiffer developed and implemented the artist-scientist team plan of action in the DYNAGROOVE project.

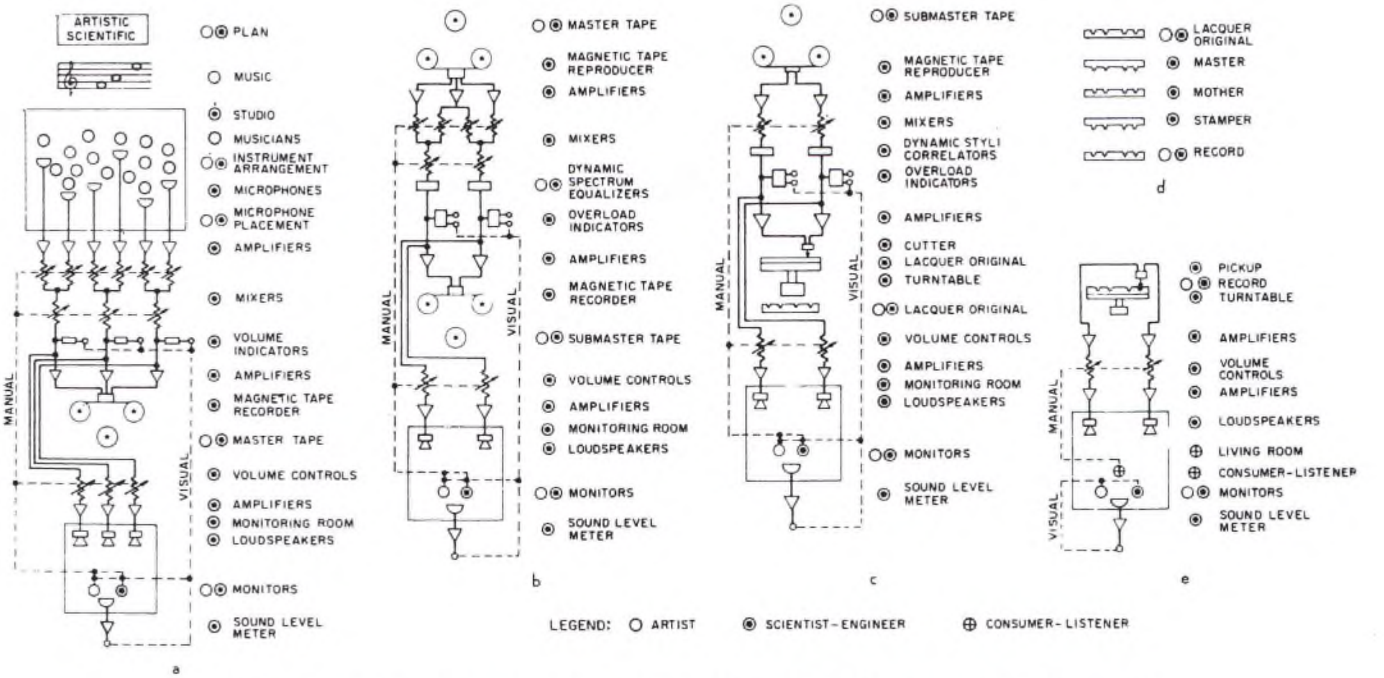

FIG. 1. DYNAGROOVE disk sound system. *a*. System for producing the master tape. *b*. System for producing the submaster tape. *c*. System for producing the lacquer original disk. *d*. Processes in the production of the disk record from the lacquer original. *e*. System for reproducing the disk record in the home.

they monitored the signal used in the cutting. They checked the disk record produced by the process shown in Fig. 1*d* in a typical living room, as shown in Fig. 1*e*. They also collaborated in determining the environment and listening habits of the average listener.

SOUND SOURCES

The characteristics of musical instruments have been under study for many years.[3] One of the objectives has been to determine the mode of operation of the instruments so as to produce the finest possible reproduced sound. The studies have included the frequency range, the frequency spectrum, the directional pattern and the growth, duration, and decay of the tones produced by the instruments.

The frequency range of musical instruments covers the entire audible frequency range. Therefore, to achieve faithful sound reproduction the reproducing system must cover the audible frequency range.

The directional patterns of musical instruments with respect to frequency and orientation are exceedingly complex. Therefore, the timbre, which plays such an important part in the identification of the instrument, varies with the orientation. As a consequence great care must be taken in providing the orientation with respect to the microphone which will give the most realistic reproduced sound identified with the particular instrument.

The transient response of the recording system must be adequate to insure faithful reproduction of instruments with rapid growth and decay characteristics.

All the characteristics of a musical instrument vary with the manner in which the instrument is played. The type of execution and rendition of a musical number differs in the recording for reproduction in the home as contrasted to that employed in the concert hall for a live performance.

To summarize: the technical data on musical instruments are used in the recording of the instruments to provide the finest sound reproduction in the home.

STUDIOS

With the sound produced by an artist or musical aggregation performing in a studio or hall, there are at an observation point the direct sound from the source and the reflected sounds from the boundaries of the enclosure. Thus, it will be seen that the studio plays a very important part in the recorded sound. Accordingly, the characteristics of studios for the recording of sound were examined.[4,5] The main objective was to provide the ideal growth and decay characteristics and a smooth response frequency transfer characteristic.

In the past the general characteristic which has been considered to provide the significant information on the acoustics of an enclosure has been the decay characteristic.

[3] See H. F. Olson, *Musical Engineering* (McGraw-Hill Book Co., New York, 1952).

[4] The development of the acoustics of recording studios for the DYNAGROOVE Project was carried out by John Volkmann.

[5] B. Bolle, H. Voldner, A. A. Pulley, A. Stevens and J. E. Volkmann, "The New RCA Italiano Recording Studios in Rome, Italy," *J. Audio Eng. Soc.* 11, 80 (1963).

However, studies have indicated that the growth characteristics are more important than the decay characteristics in providing the desired artistic effects of an enclosure. The relative importance of the reflected sounds decreases with each reflection because the intensity decreases with each reflection, due to absorption at each encounter with the boundaries (see Fig. 2). The action shown in Fig. 2 illustrates the importance of the initial reflected sounds as compared to subsequent reflections. The direct sound, D, and the four first reflections, R_1, R_2, R_3 and R_4 are shown in Fig. 2*a*. The direct sound, D, the four first reflections, R_1, R_2, R_3 and R_4 and the four second reflections R_5, R_6, R_7 and R_8 are shown in Fig. 2*b*. In Fig. 2*c* the steady state conditions have been established. The intensity of the reflections decreases with each encounter with the boundaries. The delay between the direct and reflected sound increases with the number of reflections. As a result, from a communication theory standpoint, the first reflections carry more information than the higher order reflections and therefore play a more important part.

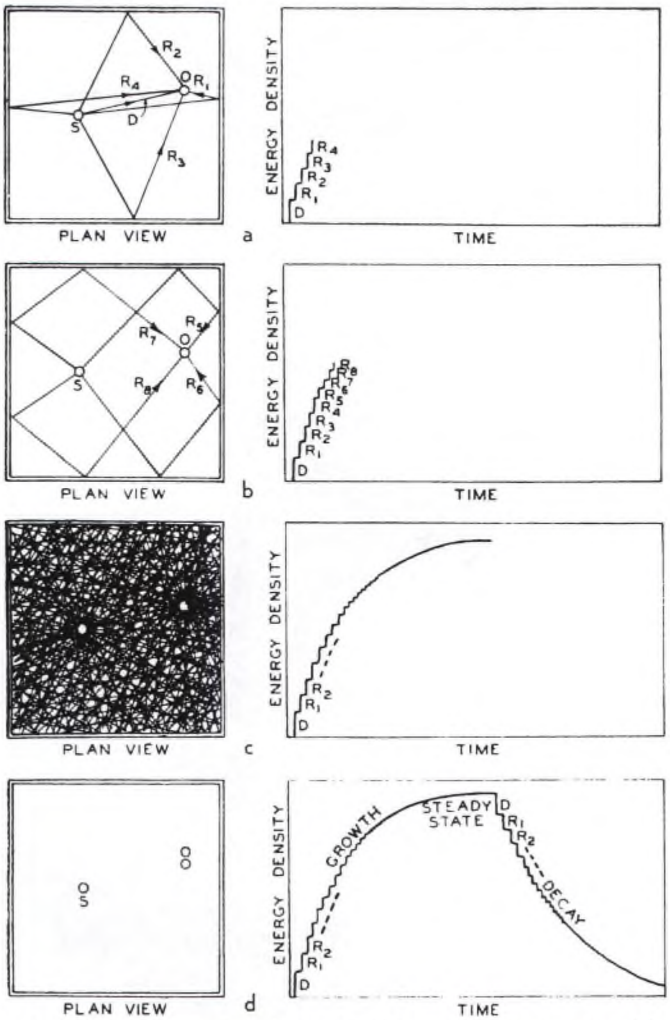

FIG. 2. A two-dimensional version of the growth and decay of the sound in a room. S is the source of sound and O is the observation point. *a*. The direct sound energy D and the reflected sound energy R_1, R_2, R_3 and R_4 from the four walls. *b*. The addition of the second reflections R_5, R_6, R_7 and R_8. *c*. A large number of reflections which approximate steady-state conditions. *d*. The decay of sound energy after the source has stopped.

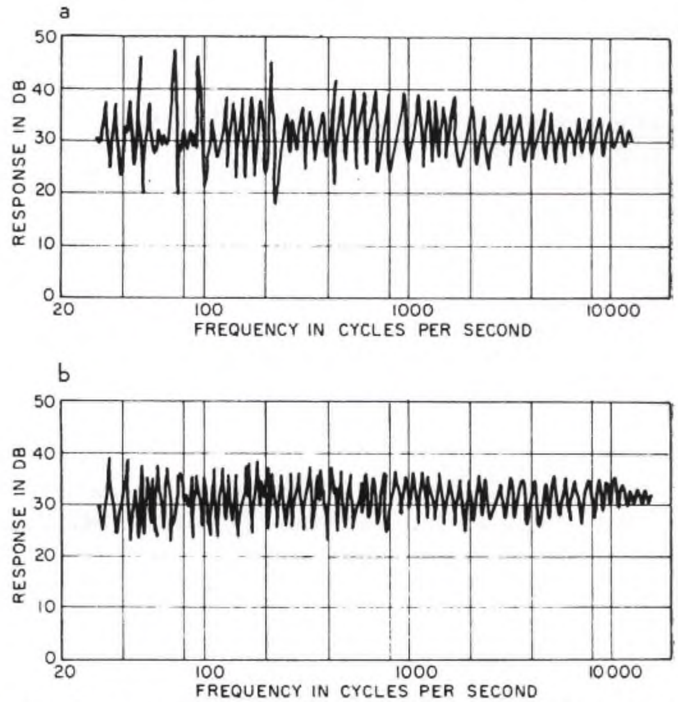

FIG. 3. Response frequency transfer characteristics of a studio. *a*. Undesirable transfer characteristic. *b*. Satisfactory transfer characteristic.

The decay characteristic depicted in Fig. 2*d* with relation to the desirable characteristic for different types of music has been explored at great length for a half century, and will not be discussed here.

The transfer characteristic from a sound source to an observation point is another important criterion which shows the anomalies in the sound pickup system. The envelope of the transfer characteristics should be smooth and free of sharp spikes or dips of the type shown in Fig. 3*a*. Such marked deviations will produce frequency augmentations, diminutions, or discriminations in the sound which

FIG. 4. An interior view of the new RCA Italiano Recording Studios in Rome, Italy.

is picked up and recorded, and are therefore undesirable. The relatively smooth transfer characteristic shown in Fig. 3*b* provides the desired performance in sound pickup.

Growth, decay, reverberation and transfer characteristics were obtained on all the studios used in the recording of DYNAGROOVE records. The studios were designed to provide the proper growth, decay, reverberation and transfer characteristics. In the case of existing studios, modifications were made to obtain the desired characteristics.

The interior of one of the studios incorporating the advanced acoustical design principles outlined above is shown in Fig. 4.

MICROPHONES

Three characteristics play a most important part in the performance of a microphone, namely, the response frequency characteristic, the directivity pattern and the nonlinear distortion characteristic.

The response frequency characteristic of the microphone should fall within the limits on amplitude and frequency shown in Fig. 5 in order to provide the pickup of sound with negligible frequency discrimination.

The directivity pattern should be uniform with respect to frequency in order to prevent frequency discrimination

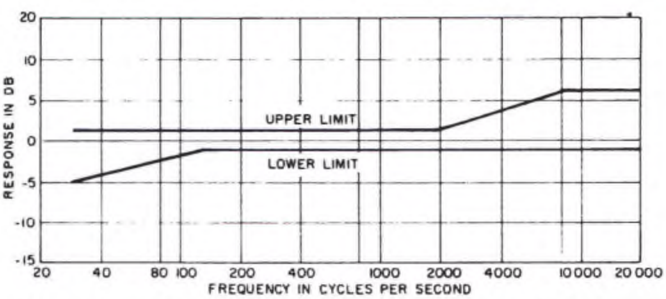

FIG. 5. Upper and lower limit characteristics for microphone response.

in the direct and reflected sounds. If the pattern for two frequencies varies, there will be frequency discrimination of a severe order in both the direct and reflected sound arriving from points removed from the axis. Therefore, the directivity pattern of a microphone should fall within the limits on amplitude and angle shown in Fig. 6.

The directivity pattern which falls within the limits of Fig. 6 provides somewhat higher directivity than the cardioid pattern. Tests have shown that microphones which exhibit higher directivity than a cardioid in general yield improved auditory perspective in sound pickup for stereophonic sound reproduction.

When a microphone with a cardioid pattern is used, stipulations similar to those shown in Fig. 6 hold for the variations of the directivity pattern with respect to frequency.

The nonlinear distortion in a microphone should be less than 0.1% for a level of 120 db over the frequency range of 30 to 15,000 cps.

The microphones selected for the recording of DYNAGROOVE records satisfied the above specifications on performance. One such microphone, the RCA MI 11010-A, is shown in Fig. 7.

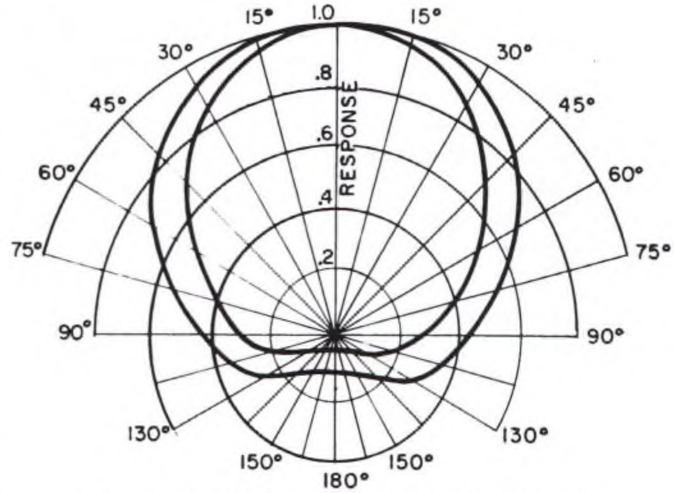

FIG. 6. Limits for microphone directivity response.

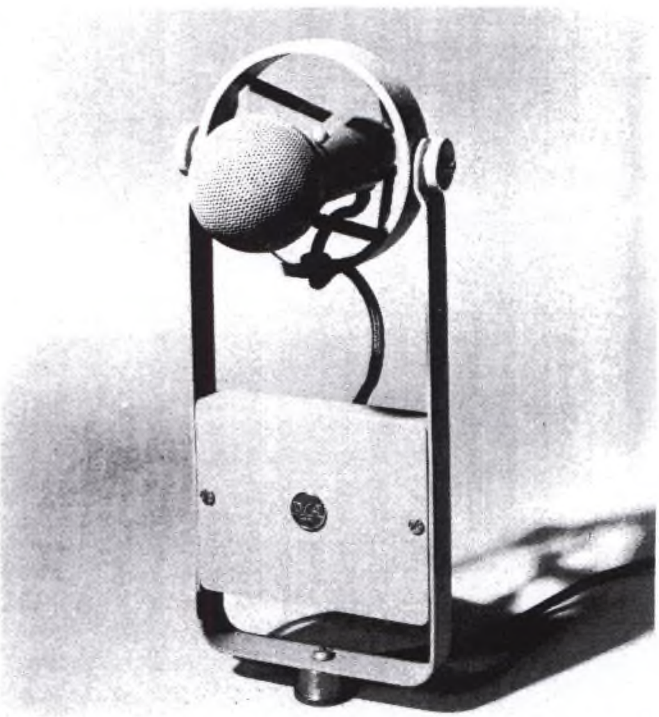

FIG. 7. One of the microphones used in DYNAGROOVE recording, the RCA MI 11010-A.

MICROPHONE PLACEMENT

The placement of the microphones with respect to the sound sources plays an important part in the subjective aspects of the reproduced sound in the living room. A studio and living room system is shown in Fig. 8. The sound sources and the microphone placement in the studio are studied by means of listening tests in a typical living room.

The fundamentals of stereophonic sound reproduction as

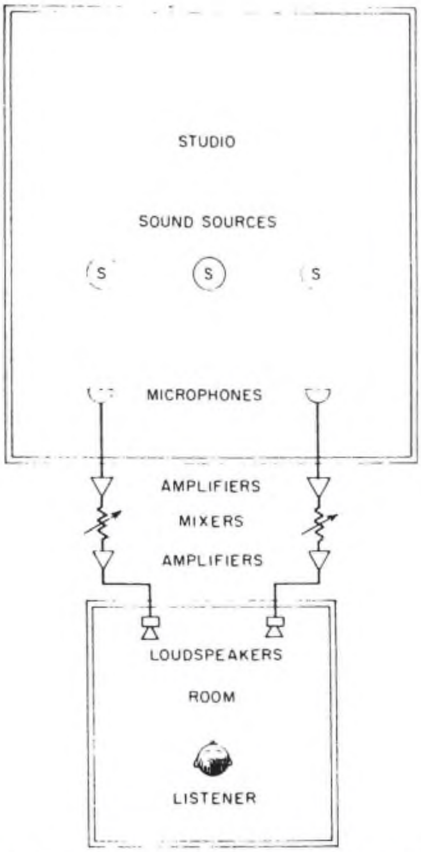

FIG. 8. Stereophonic sound reproducing system used in subjective tests of microphone placement with respect to the sound source.

applied to the experiments of Fig. 8, listed in the order in which they become apparent but not in the order of importance, are: sound location, separation and identification of tone qualities, and room acoustics. These three items constitute the elements of auditory perspective.

Reproduction of sound in auditory perspective provides a subjective illusion of the distribution of the reproduced sound sources in lateral directions as well as in depth, in a geometrical configuration and correspondence which approximate the disposition of the original sound sources.

In stereophonic sound reproduction, if the separation is sufficient, each instrument appears to come from a different location. Through his binaural sense, the listener to stereophonic sound reproduction can segregate and associate together overtones and fundamental tones which belong to each instrument.

Stereophonic reproduction provides a solution to one of the most vexing problems, namely, the microphone placement for satisfactory pickup of the reverberant sound in the recording studio. In stereophonic sound reproduction it is possible to restore reverberation to the optimal amount for direct listening. For instance, instruments whose direct sound is reproduced from one loudspeaker can have the reverberant sound reproduced from the other, and vice versa. This suggests that for maximum spacious room-effect, the instruments should be made to favor either one or the other microphone or groups of microphones for direct sound pickup and that the other microphones should be angled so as to not pick up direct sound but mostly reverberant sound for those instruments. If the difference in path to the two microphones or groups of microphones is large enough to be counted as a first reflection, the direct sound can also be picked up by the second group of microphones. Stereophonic sound reproduction provides great flexibility in obtaining realistic reverberation by the appropriate location of the microphones.

Subjective tests were carried out by means of the system shown in Fig. 8 to establish sound location, separation and identification of tone qualities, room acoustics and reverberation in relation to the placement of the microphones and the location of the sound source. It is beyond the scope of this paper to provide a detailed description of the subjective aspects which have been reported at considerable length elsewhere.[6,7]

MICROPHONE AMPLIFIERS

The design of amplifiers for use between the microphone and the mixers may appear to be so straightforward that very little attention is required for this element of the system. However, many amplifiers in this part of the chain have been found to be overloaded, particularly on sound levels of the order of 120 db which may occur on close pickup of some musical instruments. Accordingly, low distortion microphone amplifiers have been designed and employed in DYNAGROOVE recording.

MASTER RECORDING CONSOLE

A new type of recording console,[8] shown in Fig. 9, was

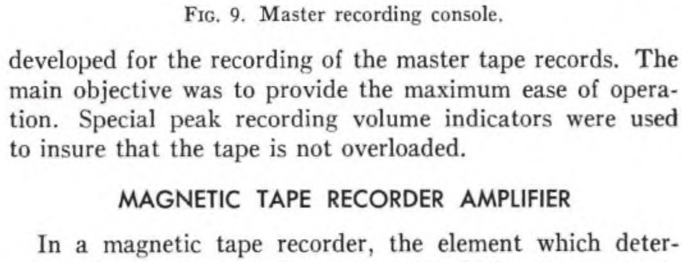

FIG. 9. Master recording console.

developed for the recording of the master tape records. The main objective was to provide the maximum ease of operation. Special peak recording volume indicators were used to insure that the tape is not overloaded.

MAGNETIC TAPE RECORDER AMPLIFIER

In a magnetic tape recorder, the element which determines the overload of the system should be the magnetic tape. An examination of the transfer characteristic of the amplifier, heads and tape indicated that in some recorders

[6] H. F. Olson, "Stereophonic Sound Reproduction in the Home," *J. Audio Eng. Soc.* 6, 80 (1958).

[7] H. F. Olson and H. Belar, "Acoustics of Sound Reproduction in the Home," *J. Audio Eng. Soc.* 8, 7 (1960).

[8] Development and design of the master recording console was carried out by D. L. Richter.

the amplifiers driving the heads were on the borderline of being inadequate in power output to provide overload of the tape without considerable distortion in the amplifier. In these instances, corrective measures were taken to insure that the magnetic tape was the limiting element in the system from the standpoint of overload.

MAGNETIC TAPE RECORDER AND REPRODUCER

The three main factors to be considered in the performance of a magnetic tape recorder and reproducer are frequency range, signal-to-noise ratio and tape speed constancy.

Three different types of magnetic tape recorders were investigated for use in the DYNAGROOVE master recordings,

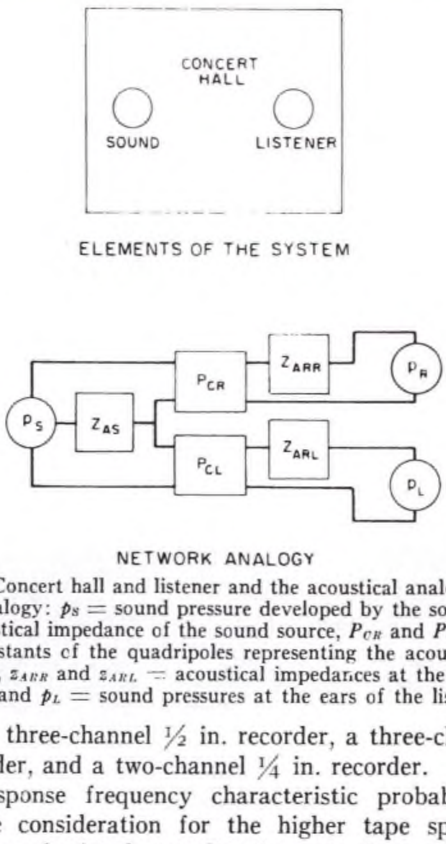

FIG. 10. Concert hall and listener and the acoustical analogy. In the network analogy: p_S = sound pressure developed by the sound source, z_{AS} = acoustical impedance of the sound source, P_{CR} and P_{CL} = propagation constants of the quadripoles representing the acoustics of the concert hall, z_{ARR} and z_{ARL} = acoustical impedances at the ears of the listener, p_R and p_L = sound pressures at the ears of the listener.

namely, a three-channel ½ in. recorder, a three-channel 35 mm recorder, and a two-channel ¼ in. recorder.

The response frequency characteristic probably needs very little consideration for the higher tape speeds employed in professional recording because there is no problem in achieving uniform response from 30 to 15,000 cps within a fraction of a db.

An adequate signal-to-noise ratio presents one of the most difficult problems to achieve in the recording of master magnetic-tape records. There are three forms of noise in magnetic tape recorders: the uncorrelated or random noise due to the particle nature of the tape, the uncorrelated or random noise due to modulation produced by small nonrecurrent but rapid variations in motion of the tape and to head contact with the tape, and the correlated noise due to print-through. An improvement of 3 db was achieved in the random signal-to-noise ratio by using a tape speed of 30 ips as contrasted to 15 ips. A significant increase in the signal-to-noise ratio as well as in the maximum output level was obtained by the development of a tape with a higher retentivity and a lower noise level. A lower print-through was in part achieved by the use of a uniform particle coercivity and a heavier base material. In general, wow and flutter are reduced by the use of a higher speed. Employing a tape speed of 30 ips as contrasted to 15 ips reduced the wow and flutter by about 50%.

DYNAMIC SPECTRUM EQUALIZER

An original master tape produced with the equipment described in preceding sections will exhibit clarity, presence, low nonlinear distortion, uniform transfer charactertistic and the other desirable characteristics described. This tape can be transferred without any alterations to produce an excellent disk record for reproduction in the home. The question arises as to whether it will sound like the original music when played back on the consumer's home instrument in his own living room. The answer is "no," for reasons which will be developed in this section.

A simple approximation for a concert hall, a single sound source, and a listener are represented by the dynamical analogy of Fig. 10. There are first the sound pressure, p_S and the acoustical impedance z_{AS} of the sound source. The sound source is coupled to two quadripoles, with propagation constants P_{CR} and P_{CL}, which represent the acoustics of the concert hall between the sound source and the ears of the listener. At the listener, there are the acoustical impedances z_{ARR} and z_{ARL}, and the generated sound pressures at the ears of the listener, p_R and p_L.

A simple approximation of the chain consisting of a sound source in a studio, a two-channel stereophonic recording and reproducing system, and a listener in a living room

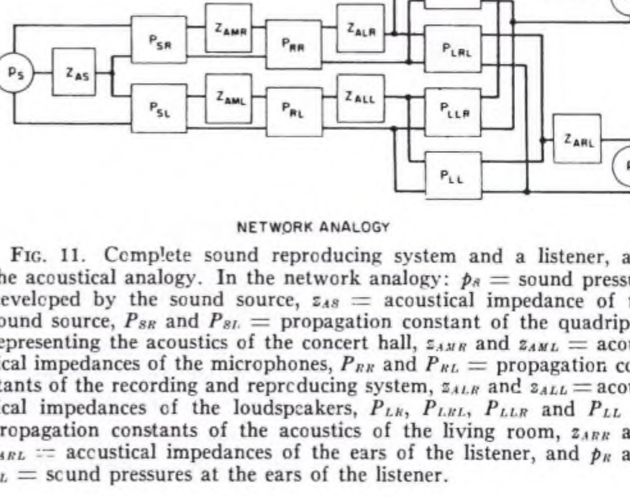

FIG. 11. Complete sound reproducing system and a listener, and the acoustical analogy. In the network analogy: p_S = sound pressure developed by the sound source, z_{AS} = acoustical impedance of the sound source, P_{SR} and P_{SL} = propagation constant of the quadripole representing the acoustics of the concert hall, z_{AMR} and z_{AML} = acoustical impedances of the microphones, P_{RR} and P_{RL} = propagation constants of the recording and reproducing system, z_{ALR} and z_{ALL} = acoustical impedances of the loudspeakers, P_{LR}, P_{LRL}, P_{LLR} and P_{LL} = propagation constants of the acoustics of the living room, z_{ARR} and z_{ARL} = acoustical impedances of the ears of the listener, and p_R and p_L = sound pressures at the ears of the listener.

is represented by the dynamical analogy of Fig. 11. Again, there are first the sound pressure, p_S and the acoustical impedance z_{AS} of the sound source. The sound source is coupled to two quadripoles with propagation constants, P_{SR} and P_{SL}, which represents the acoustics of the studio between the sound source and the microphone. The acoustical impedances of the microphones are z_{AMR} and z_{AML}. The microphones are coupled to quadripoles with propagation constants P_{RR} and P_{RL}, which represent the sound recording and reproducing equipment. The outputs of the quadripoles representing the sound reproducing equipment are coupled to the loudspeakers, with coupling acoustical impedances z_{ALR} and z_{ALL}. The loudspeakers are coupled to four quadripoles with propagation constants P_{LR}, P_{LRL}, P_{LLR} and P_{LL}, representing the acoustics of the living room between the loudspeakers and the ears of the listener. At the listener, in the living room, there are the acoustical impedances z_{ARR} and z_{ARL} and the generated sound pressures at the ears of the listener, p_R and p_L.

Certainly no one could conclude that if the sound recording and reproducing equipment of Fig. 11 displayed a perfect transfer characteristic, then the results obtained at the listener in Fig. 11 would be the same as those in Fig. 10. Therefore, a perfect transfer characteristic is not the ideal transfer characteristic.

From the preceding discussion, the conclusion follows that a perfect transfer characteristic in the sound recording and reproducing system is not the answer to a simulation of concert hall performance. The next logical step is a consideration of the factors involved in providing the listener with sound reproduction which exhibits the highest order of artistic and subjective resemblance to that in the concert hall.

Extensive subjective tests have been conducted on the stereophonic reproduction of sound at various loudness levels under the acoustical conditions and environments of the average living room in the home. Studies have also been made of the reproduction level employed by consumers in their homes. These tests have shown that peak level of sound reproduction in the homes of consumers runs from 70 to 90 db for 90% of the listeners. The average listener in the home operates a record reproducing system at a peak level[9] of 80 db.

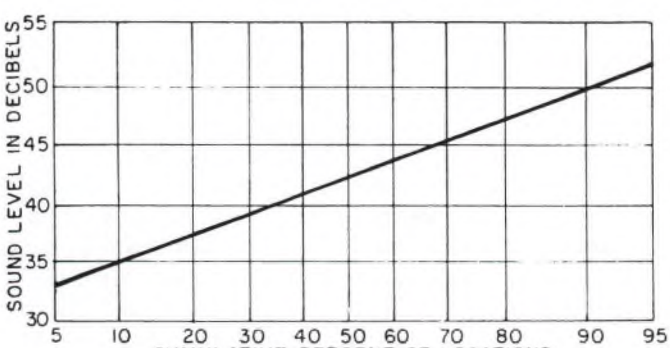

Fig. 12. Room noise in residences. (After Seacord)

The peak level of sound delivered by a symphony orchestra in the concert hall is about 100 db. Thus, it will be seen that the peak level of sound reproduction in the home is much lower than the level in the concert hall.

The main reason why the average listener prefers a lower level of sound reproduction in the home as contrasted to the sound level in the concert hall is that the tolerable peak sound level[10] in a small room is lower than the tolerable peak sound level in a large hall. The shorter mean free path and resultant faster growth and decay of sound in a small room appears to lead to a lower tolerable peak level in the small room. Subjective tests have indicated that the same results are obtained regardless of whether the sound program is live or reproduced.

The ambient noise level[11] in the average residence is a factor that must be considered in the reproduction of sound in the home. The ambient noise level in the average residence, as shown in Fig. 12, is 43 db. The ambient noise in 90% of the residences falls between 33 and 52 db.

The spectrum of room noise[12] shown in Fig. 13 is a factor which must be considered in the reproduction of sound in the home. The most significant aspect of the graph of Fig. 13 is that noise increases with decrease of frequency.

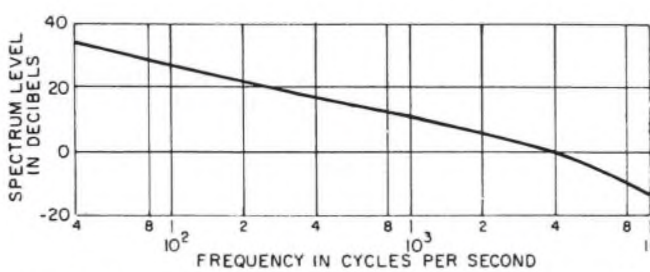

Fig. 13. Average noise spectrum for residences, with 0 db = .000204 dyne per square centimeter. (After Hoth)

The next subject is the masking[13] of the reproduced sound program by the ambient noise. In the case of wide-band noise of the type encountered in rooms, it is possible to obtain the masking from the spectrum level of Fig. 13 and the masking contours of wide-band noise shown in Fig. 14. The masking M is the difference between the threshold of hearing in the presence of noise and in a quiet place. The quantity B is the spectrum level defined by

$$B = 10 \log (I/WI_0) \quad (1)$$

where I = sound intensity in the frequency bandwidth W, W = bandwidth, and I_0 = zero reference level.

[9] Peak level in these considerations is used to designate a level such that 95% of the program lies below it.

[10] The term "tolerable peak sound level" is used to designate the peak level of sound reproduction which the listener feels is acceptable and agreeable.

[11] D. F. Seacord, "Room Noise at Subscribers' Telephone Locations," *J. Acoust. Soc. Am.* **12**, 183 (1940).

[12] D. F. Hoth, "Room Noise Spectra at the Subscribers' Telephone Locations," *J. Acoust. Soc. Am.* **12**, 499 (1941).

[13] H. Fletcher and W. A. Munson, "Relation Between Loudness and Masking," *J. Acoust. Soc. Am.* **9**, 1 (1937).

The masking curve can be deduced from the intensity level per cycle curve, termed the spectrum curve of Fig. 13, the masking contours of Fig. 14 and the threshold curve of Fig. 15. For example, for a spectrum level of 9 db at 1000 cycles the masking level is 25 db. The masking curve for

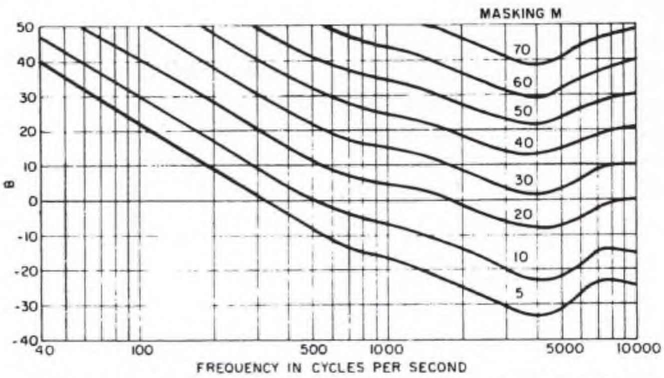

FIG. 14. Masking contours for wide band noise M is the difference between the threshold of hearing in the presence of noise and a quiet place. B is the spectrum level. (After Fletcher and Munson)

average room noise, obtained by the above procedure, is shown in Fig. 16. The graph of Fig. 16 depicts the hearing limits for pure tones, i.e., a tone of a level below this curve cannot be heard. A direct listening test was carried out to determine the threshold of pure tones for a small room exhibiting an ambient noise spectrum as shown in Fig. 13. These tests substantiated the characteristic of Fig. 16 within the usual limits of subjective tests. The threshold characteristic of Fig. 16 then established the lower level of hearing in a room in the average residence.

Not everyone has the perfect hearing represented by the threshold curve of Fig. 15. The threshold of hearing[14] for

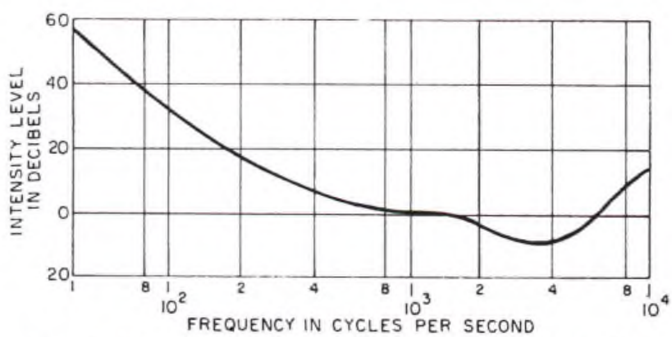

FIG. 15. The threshold of hearing in quiet place, with 0 db = .000204 dyne per square centimeter.

50% of the population is shown in Fig. 17. Thus, the thresholds of 50% of the population lie above this characteristic and 50% below this characteristic. In a comparison of Figs. 16 and 17 it will be seen that the ambient noise curve lies above the threshold curve for the average listener. Therefore, for the average listener, the ambient noise determines the threshold.

[14] J. C. Steinberg, H. C. Montgomery and M. B. Gardner, "Results of World's Fair Hearing Tests," *J. Acoust. Soc. Am.* **12**, 291 (1940).

The peak level of sound reproduction in the average listener's home is 80 db. The data of Fig. 16 coupled with the peak level of 80 db of sound reproduction establish the amplitude limits of the reproduced sound in the home for the average listener. The relatively low peak level of sound reproduction and the ambient noise level in the home are among the main factors which must be considered in providing realistic sound reproduction for the consumer in the home. Within this framework, there are three other important characteristics that must be considered in the reproduction of sound in the home, namely, the loudness *vs*

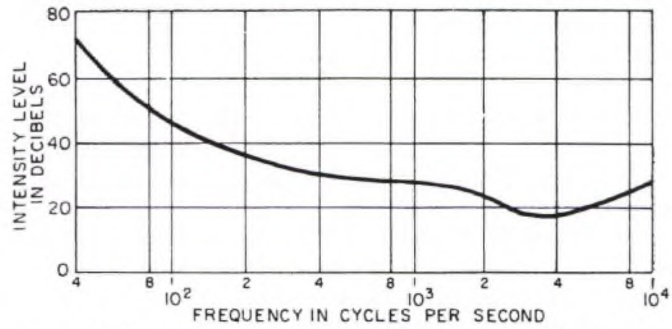

FIG. 16. Hearing limits for pure tones for a typical listener in a typical residence, with 0 db = .000204 dyne per square centimeter.

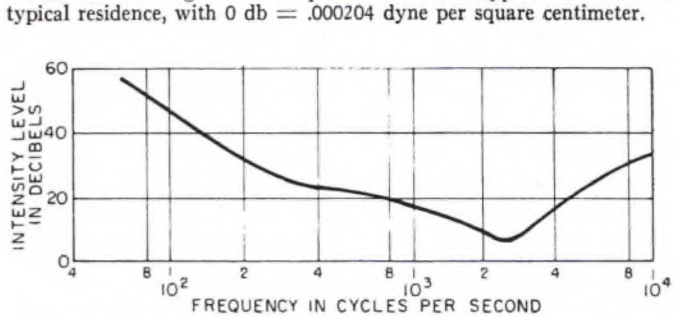

FIG. 17. Contour of the threshold of hearing for 50% of the population, with 0 db = .000204 dyne per square centimeter. (After Steinberg, Montgomery and Gardner)

intensity relation in hearing, the response frequency characteristic of equal loudness in hearing, and the reverberation characteristic of the average room in the home.

The equal-loudness frequency relation of hearing, or the response frequency characteristic of the human hearing mechanism, has been determined by several investigators. The characteristics shown in Fig. 18 are an average of the investigations carried out by Fletcher and Munson,[15] Churcher and King,[16] and Robinson and Dadson.[17] With respect to the compensation in going from one level to another for a change of 20 db or less, which is the point of interest in the design of the dynamic spectrum equalizer, the data from the three different investigations yield very

[15] H. Fletcher and W. A. Munson, "Loudness—Its Definition, Measurement and Calculation," *J. Acoust. Soc. Am.* **5**, 82 (1933).

[16] B. G. Churcher and A. J. King, "The Performance of Noise Meters in Terms of the Primary Standard," *J. Inst. Elect. Engrs.* (London) **81**, 57 (1937).

[17] D. W. Robinson and R. S. Dadson, "A Redetermination of the Equal Loudness Relations for Pure Tones," *British J. Appl. Phys.* **7**, 166 (1956).

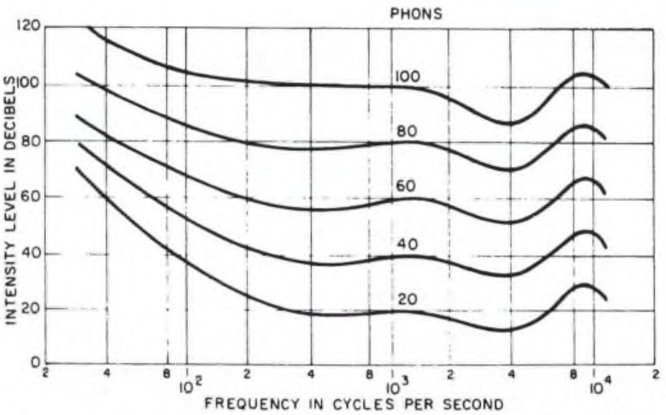

FIG. 18. Contour curves of equal loudness, with 0 db = .000204 dyne per square centimeter.

nearly the same results. Essentially the same results are obtained from the response frequency characteristics of the human hearing mechanism from the draft recommendation of the I. S. O.[18] Therefore, it seems logical to use the average data of the three investigations listed above. In this connection, the response frequency characteristics of the human hearing mechanism serve as a guide for the subjective tests which will be described later on and which were used to establish the performance characteristics of the dynamic spectrum equalizer.

The response frequency characteristics of the human hearing mechanism as depicted in Fig. 18 indicate that certain of the frequency ranges must be increased or decreased in amplitude in order to maintain the quality bal-

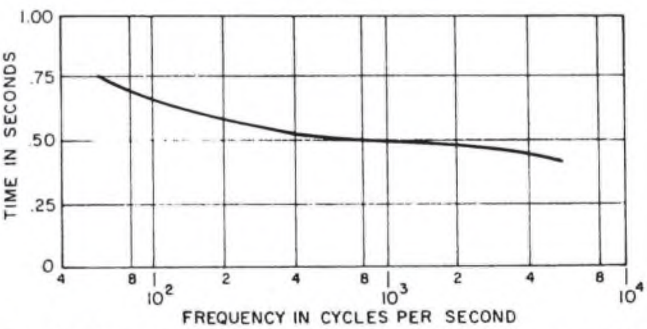

FIG. 19. Reverberation time as a function of the frequency for typical rooms in residences.

ance of music when it is reproduced at a lower level than the original. One of the attempts to solve this problem has been approached by sound reproducing equipment manufacturers by the installation in the preamplifier stages of a loudness control combined with the volume control to emphasize the lower and higher frequency ranges when the volume is low. These measures consider the static characteristics of hearing and the relation to the psychological judgement of the sound at various levels. These systems influence low-level music as well as high-level music in the same ways and have no dynamic amplitude discrimination.

[18] International Organization for Standardization, *Draft Recommendation R 226*, 1961.

Therefore, means must be provided which will introduce frequency accentuation appropriate to the instantaneous level.

In the reproduction of sound in a room there are two sources[19] of sound with respect to the listener, namely, the direct sound from the loudspeaker and the generally reflected sound. The acoustical characteristics of the average room in a residence accentuates the low-frequency response, as can be deduced from the reverberation characteristic of Fig. 19. Most direct radiator loudspeakers exhibit increased directivity with increased frequency. The combination of

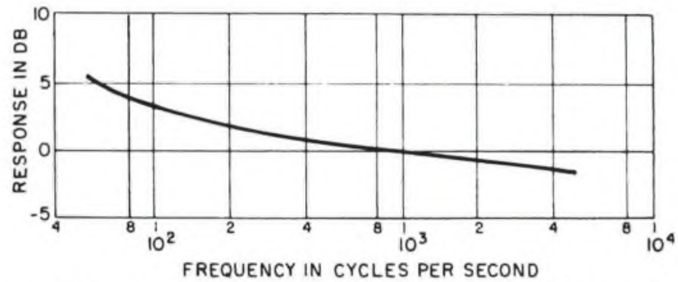

FIG. 20. The relative response derived from the dynamic average of the direct and reflected sound reproduced in a room in a residence.

the acoustical characteristics of the room and loudspeakers conspires to produce an accentuation in the low-frequency response as perceived by the listener. The relative response at normal listening distances derived from the dynamic average of the direct and generally reflected sound for the case of music reproduced in a room in a residence is shown in Fig. 20.

The loudness *vs* loudness level[20,21] depicted in Fig. 21

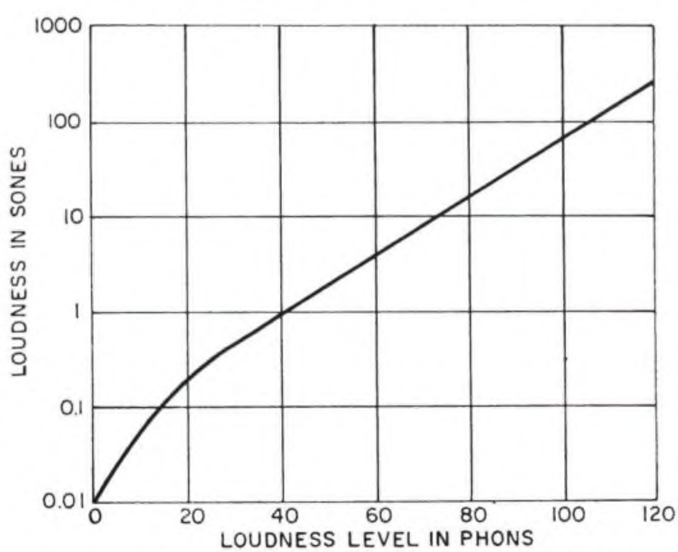

FIG. 21. The relationship between loudness in sones and the loudness in phons.

[19] H. F. Olson, *Acoustical Engineering* (D. Van Nostrand Co., Princeton, N. J., 1957).
[20] S. S. Stephens, "The Measurement of Loudness," *J. Acoust. Soc. Am.* **27**, 815 (1955).
[21] J. P. A. Lochner and J. F. Burger, "Pure-Tone Loudness Relations," *J. Acoust. Soc. Am.* **34**, 576 (1962).

must also be considered in the transition from the concert hall or studio to the small room in the home. The loudness level of the reference tone expressed in phons is the intensity level of the reference tone (1000 cps) in decibels. The loudness of any other sound is determined by adjusting the reference tone until it sounds equally loud. The loudness level of a sound, in phons, is numerically equal to the intensity level in decibels of the 1000 cps pure tone which is judged by the listener to be of equivalent loudness. Thus, the phon is the unit of loudness level as specified in the preceding sentence. A scale showing the relation between loudness in sones and loudness level in phons is shown in Fig. 21. The loudness *vs* loudness level is another factor which must be considered in the transition from a relatively high sound level in a large room to a low sound level in a small room.

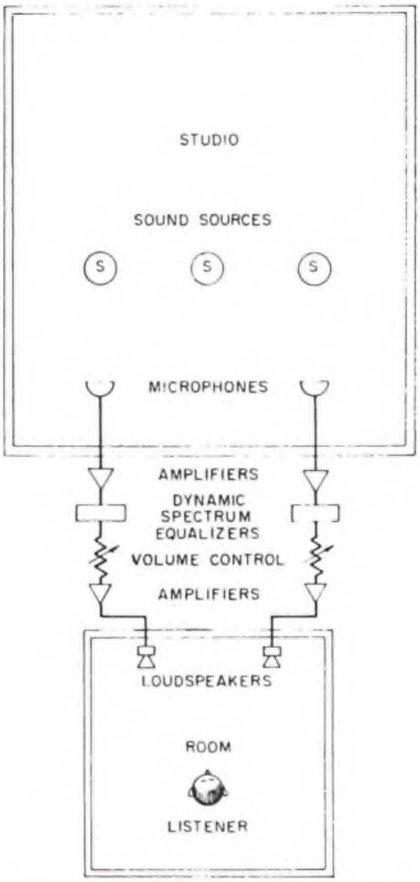

FIG. 22. Stereophonic sound reproducing system used to develop the characteristics and operation of the Dynamic Spectrum Equalizer.

The main equalizations are due to a drop in the level of about 20 db, which can be deduced from Fig. 18, and the increased low-frequency response from the acoustics of the room of Fig. 20. This process leads to a first approximation of the required equalization.

To summarize, there are six characteristics that must be considered in the reproduction of sound in a room in the home: the peak level of sound reproduction in the home, the ambient noise level in a room in a home, the spectrum of the ambient noise, the loudness *vs* loudness level, the response frequency characteristic of the ear, and the reverberation characteristic of the average room in a home.

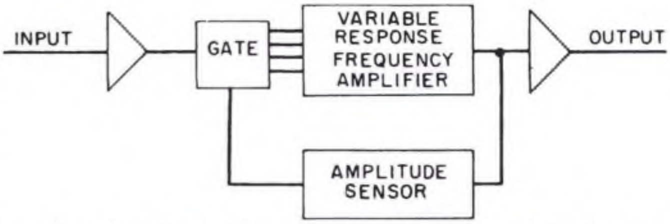

FIG. 23. Schematic block diagram of the Dynamic Spectrum Equalizer.

Employing the first-approximation equalization derived as above, the next consideration is any indicated modification of this characteristic by use of subjective tests so as to provide the listener in his home environment with sound reproduction which exhibits the highest order of artistic and subjective resemblance to that of the corresponding condition in the concert hall.

The arrangement shown in Fig. 22 was used to carry out this project. The small room was similar in all respects to the typical living room in the home. As a result of the subjective experiments, a dynamic spectrum equalizer was

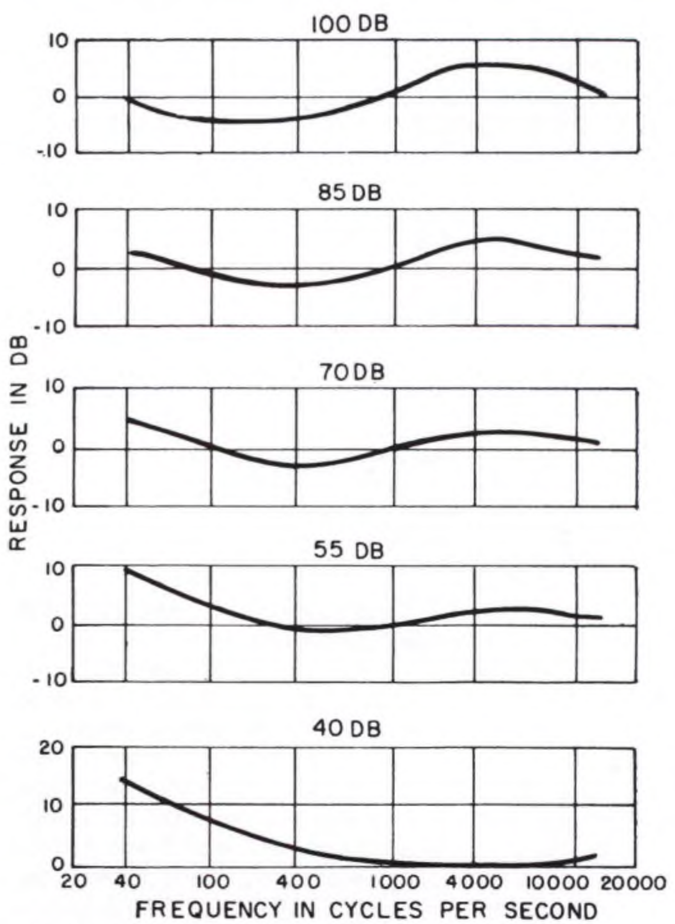

FIG. 24. Response frequency characteristics of the Dynamic Spectrum Equalizer for various sound levels of the program.

developed which, in the broadest terms, translates the sonic equalities of the original performance into stimuli which will project that performance into the perception of the home listener with the greatest possible proficiency. A block diagram of the dynamic spectrum equalizer is shown in Fig. 23. The system operates in a continuous manner to change the response frequency characteristic as a function of the amplitude. Typical response frequency characteristics for various levels are shown in Fig. 24. There is a continuous variation in response from one level to another. The response frequency characteristics differ for different types of musical selections; in effect, when the levels are low the low-frequency components are accentuated. For medium levels there are slight accentuations in the low-frequency region and the presence region of 2000 to 6000 cps, and a reduction in response in the region from 400 to 1000 cps. For high sound levels there is accentuation in response in the presence region, and a reduction in the frequency range below 1000 cps. When the sound level of the program is low, the objective is to raise the sound level of the appropriate frequency regions so that the music can be appreciated under the ambient noise and surrounding conditions of the average residence. When the sound level of the program is high, the level of the presence region is raised and the level of the low-frequency range is lowered. This procedure does not upset the dynamic balance but rather enhances this aspect of sound reproduction in a small room.

The dynamic spectrum equalizer was designed to provide a dynamic alteration of the projection qualities of sound

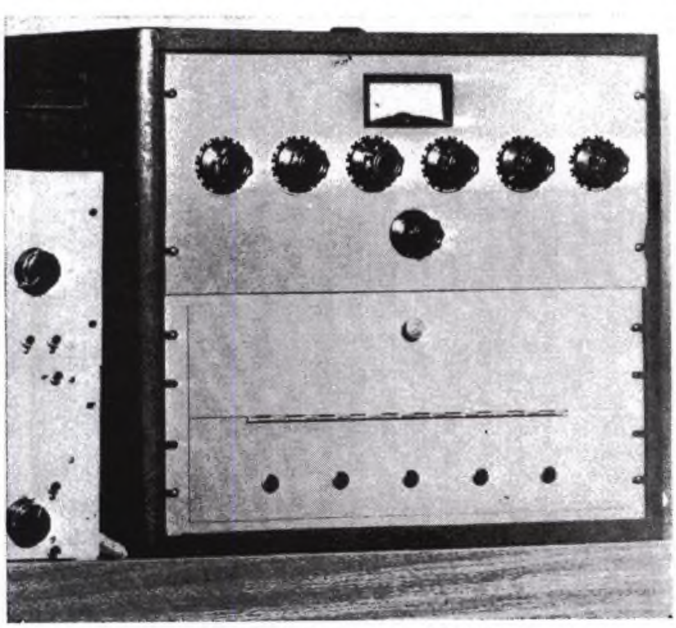

FIG. 25. Dynamic Spectrum Equalizer.

so that under conditions of playback, which differ from those in which the music was performed, the best perception of the qualities of the original performance would be obtained. The unit used in the DYNAGROOVE project is shown in Fig. 25.

RECORDING OVERLOAD INDICATOR

The recording overload indicator[22] was designed to provide indications of the maximum allowable signal which can be applied in the cutting of the master stereophonic disk record. A schematic block diagram of the indicator for one channel is shown in Fig. 26. The recording overload indicator is provided with two separate indicating meters in each of the two stereophonic channels. One meter is calibrated to show the occurrence of program peaks which will cause curvature overloading[23] in the high frequency range.

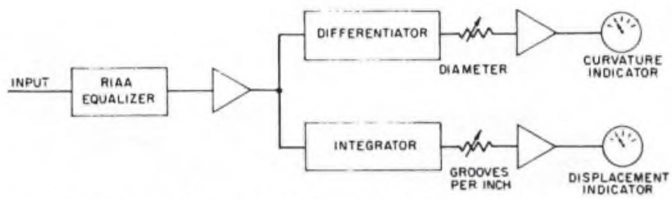

FIG. 26. Schematic block diagram of the Recording Overload Indicator.

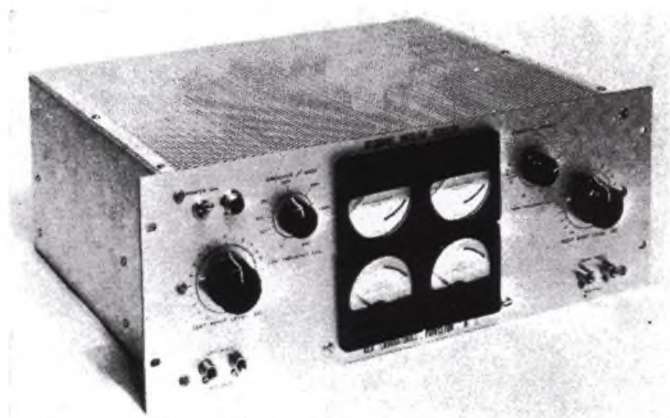

FIG. 27. Stereophonic disk recording overload indicator.

This is accomplished by the use of a differentiator and meter which indicates curvature overload. The other meter is calibrated to show displacement overloading in the low frequency range by means of an integrator which indicates amplitude overload. A photograph of the stereophonic recording overload indicator is shown in Fig. 27.

SUBMASTER RECORDING CONSOLE

A recording console[24] has been designed for recording the submaster magnetic tape from the master tape. The submaster recording console (see Fig. 28) contains the dynamic spectrum equalizer and employs the recording overload indicator as well as the auxiliary control equipment for producing submaster magnetic tape.

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[22] The recording overload indicator was developed by R. W. George, J. G. Woodward and E. C. Fox.

[23] L. W. Septmeyer, "A Curvature Meter for Use in Disk Recording," *J. Acoust. Soc. Am.* **19**, 161 (1947).

[24] The development and design of the transfer recording console was carried out by D. L. Richter.

as shown in Fig. 29*Ya*. However, the groove which the chisel-shaped recording stylus cuts in the master record is not a sinewave, but rather is distorted in such a manner that the resultant motion of the ball-tipped reproducing stylus will be a sinewave (see Fig. 29*Yb*). Under these conditions, the motion of the reproducing stylus will be a sinewave, as shown in Fig. 29*Yc*. The electrical wave output of the pickup will also be a sinewave, as shown in Fig. 29*Yd*. The mechanism of Fig. 29 indicates that the introduction of complementary distortion in the recording process will reduce tracing distortion in the reproduction of the record. An electronic system which provides this type of distortion has been developed and termed a Dynamic Styli Correlator.[30]

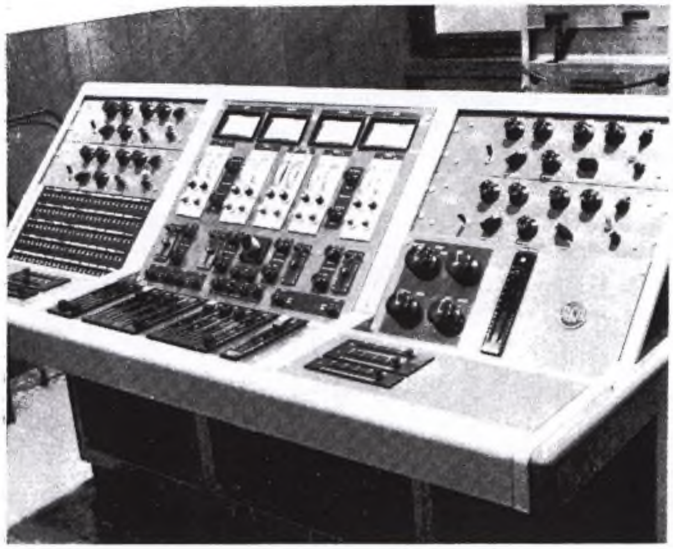

FIG. 28. Submaster recording console.

MONITORING ROOM

The audio monitoring of the recording of the submaster magnetic tape is carried out in a room with dimensions and acoustics similar to those of a typical living room. A sound level meter is used to check the level of the reproduced sound.

DYNAMIC STYLI CORRELATOR

The master stereophonic disk record is cut with a chisel-type stylus and the replica of the master disk record is reproduced by a ball-tipped stylus. Therefore, there is a discrepancy[25-29] between the motion of the cutting stylus and reproducing stylus, which becomes more pronounced at the shorter wavelengths. The recording and reproducing process in one channel of a stereophonic disk system is depicted in Fig. 29*X*. The input signal is depicted as a sinewave in Fig. 29*Xa*. The groove which the chisel-shaped stylus cuts in the master record is also a sinewave, as shown in Fig. 29*Xb*. When the ball-tipped reproducing stylus moves along the groove, the motion is not a sinewave but is distorted in a manner which is characteristic of a vertical recording system, as shown in Fig. 29*Xc*. The wave of the electrical output of the pickup is unsymmetrical, which means that the major distortion component is the second harmonic (see Fig. 29*Xd*). The distortion can be eliminated by the introduction of complementary distortion, as depicted in Fig. 29*Y*. The input signal is again a sinewave,

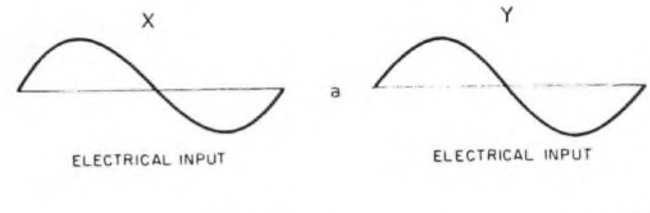

FIG. 29. *X*. Cutting with no predistortion. *Y*. Cutting with predistortion.

Tracing distortion does not arise from a simple nonlinear transfer characteristic, but rather is due more basically to a phase modulation process. This can be illustrated by the wave shapes shown in Fig. 30. The signal wave shape is a sinewave, as shown in Fig. 30*a*. The desired wave shape of the groove is shown in Fig. 30*b*. The signal wave shape and the desired groove wave shapes are shown in Fig. 30*c*. A consideration of the latter figure shows that the desired

[25] M. J. Di Toro, "Distortion in the Reproduction of Hill and Dale Recording," *J. Soc. Motion Picture Engrs.* **29**, 493 (1937).

[26] J. A. Pierce and F. V. Hunt, "Distortion in Sound Reproduction from Phonograph Records," *J. Soc. Motion Picture Engrs.* **31**, 157 (1938).

[27] W. D. Lewis and F. V. Hunt, "Theory of Tracing Distortion in Sound Reproduction from Phonograph Records," *J. Acoust. Soc. Am.* **12**, 348 (1941).

[28] M. S. Corrington, "Tracing Distortion in Phonograph Records," *RCA Review* **10**, 241 (1949).

[29] M. S. Corrington and T. Murakami, "Tracing Distortion in Stereophonic Disk Recording," *RCA Review* **19**, 216 (1958).

[30] E. C. Fox and J. G. Woodward, "Tracing Distortion—Its Causes and Correction in Stereodisk Recording Systems," *J. Audio Eng. Soc.* **11**, 294 (1963).

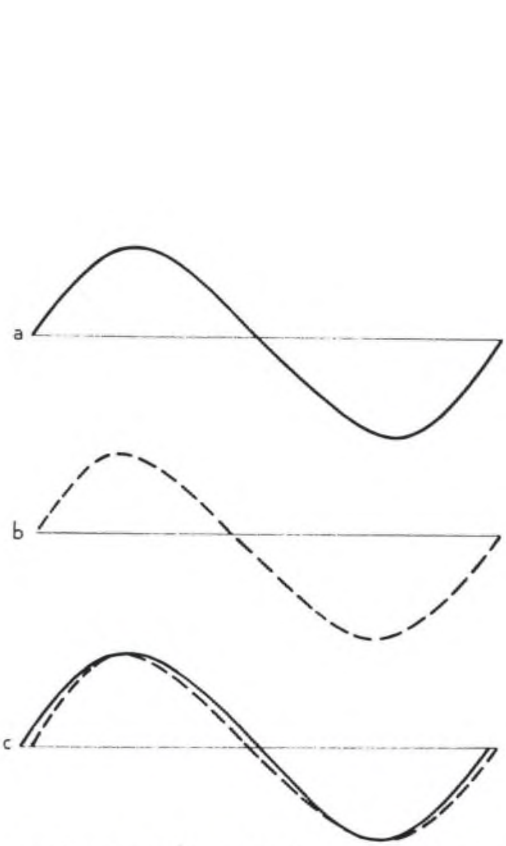

FIG. 30. *a.* Signal wave shape. *b.* Desired wave shape. *c.* Signal wave and groove shapes depicting the instantaneous phase shift between the two waves.

wave shape can be obtained from the signal wave shape by the application to the signal of an appropriate instantaneous and continuous phase shift. In the Dynamic Styli Correlator, an output signal voltage of the desired phase shift at any instant is provided, so that the resultant output of the pickup will be the same as the signal voltage applied to the recording system. Accordingly, the Dynamic Styli Correlator consists of a delay line with taps along the line (see Fig. 31). The delay line in conjunction with the sampling gates provides the proper electrical phase correction as a function of frequency and amplitude, so that the electrical wave-shape output of the pickup in reproducing corresponds to the electrical wave-shape input to the cutter in

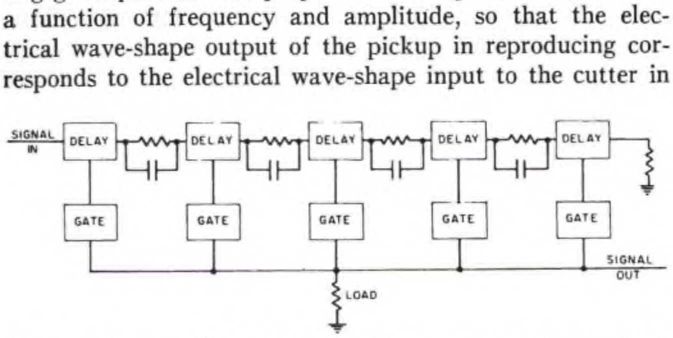

FIG. 31. Schematic block diagram of the Dynamic Styli Correlator.

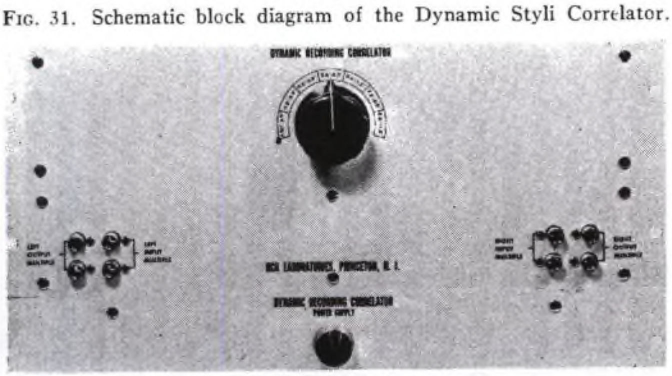

FIG. 32. Dynamic Styli Correlator.

recording. That is to say, the groove cut in the record for a sinewave input to the cutter corresponds to that shown in Figs. 29*Yb* or 30*b*.

A photograph of the Dynamic Styli Correlator is shown in Fig. 32.

The ultimate test of the principle of electronic simulation of tracing distortion as embodied in the Dynamic Styli Correlator is its effectiveness in reducing distortion in actual record-playback operation. A number of test signals were employed. While it is beyond the scope of this paper to show all of these, a more complete discussion has been carried out elsewhere.[31]

A typical case of the reduction in tracing distortion is shown in Fig. 33 in which a combination of a 400 cps tone

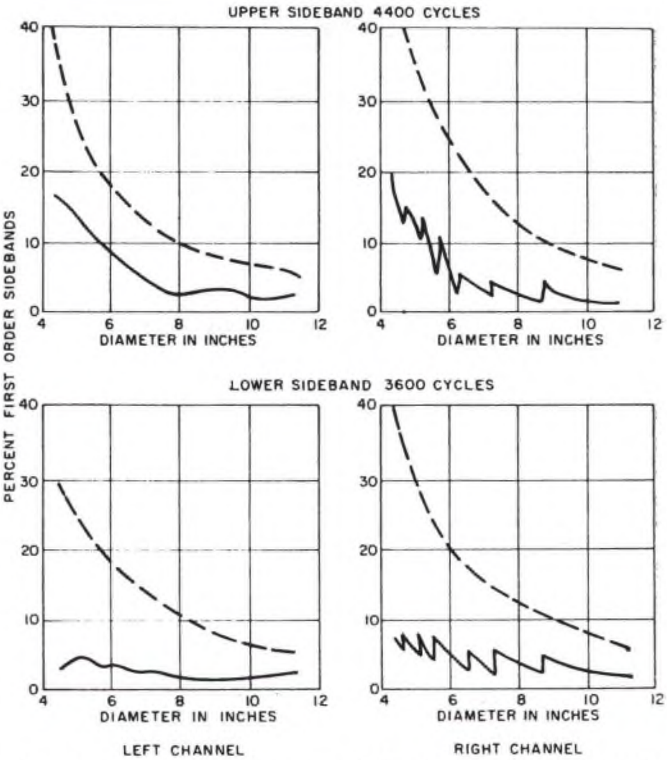

FIG. 33. The percent first order sidebands with and without the Dynamic Styli Correlator for a signal input of 400 and 4000 cps as a function of the groove diameter. Solid and dashed lines depict operation with and without the Dynamic Styli Correlator respectively.

and a 4000 cps tone were applied to the cutter with and without the correlator. The reduction in distortion is indeed very significant, being of the order of 6 to 1 at the inner grooves of the record.

VERTICAL TRACKING

A disparity between the effective vertical angle in the recorder cutting a modulated groove and the vertical tracking angle of the pickup will introduce harmonic and inter-

[31] E. C. Fox and J. G. Woodward, "Tracing Distortion—Its Causes and Correction in Stereodisk Recording Systems," *J. Audio Eng. Soc.* 11, 294 (1963).

modulation distortion in the output from the pickup.[32,33] A diagram depicting the vertical tracking angle in a magnetic pickup is shown in Fig. 34, and a diagram showing the geometrical tracking angle in a stereodisk recorder in Fig. 35. A discrepancy between the effective vertical angles of the cutter and pickup will introduce nonlinear distortion, as depicted in Fig. 36. If the pickup stylus follows the sinusoidal modulation the motion is given by

$$y = y_o \sin (2\pi x/\lambda) \qquad (2)$$

where y = instantaneous vertical displacement at a point x, y_o = peak vertical displacement, x = instantaneous longitudinal displacement, and λ = wavelength.

But the stylus is constrained to move along a line making

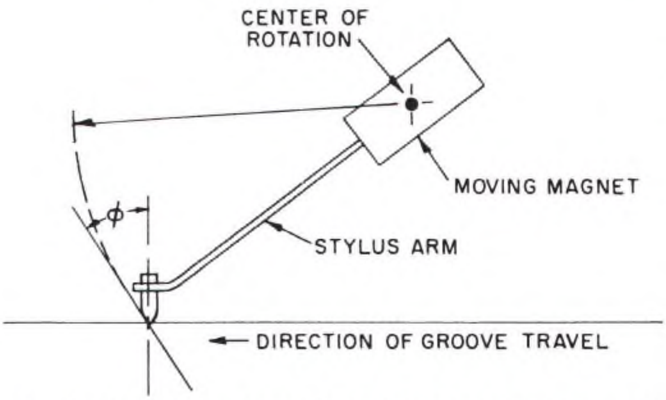

FIG. 34. Diagram showing the vertical tracking angle in a moving magnet pickup.

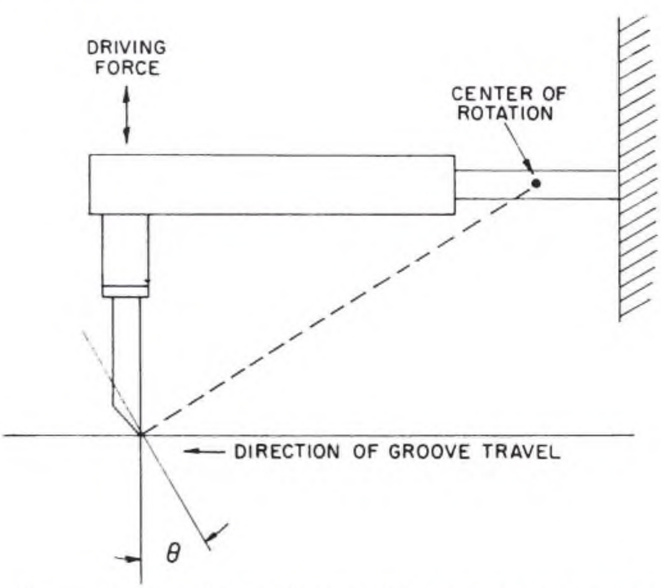

FIG. 35. Diagram showing the vertical tilt angle in a stereophonic disk recorder.

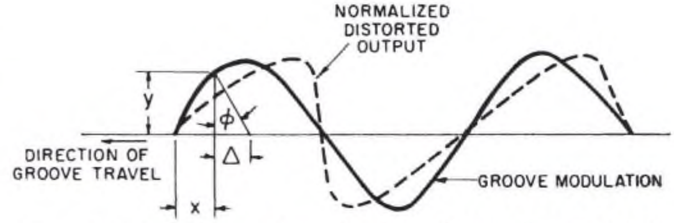

FIG. 36. Distorted waveform of a sinusoidal vertical modulation played back with a pickup having a vertical tracking angle ϕ.

an angle ϕ with the vertical, as shown in Fig. 34. The result is an instantaneous phase shift of

$$\Delta = y \tan \phi \qquad (3)$$

where Δ = phase shift shown in Fig. 36.

During positive displacements, Δ has positive values representing phase advances. As a result the output of the displacement-sensitive pickup will have the waveform depicted in Fig. 36. The output shown in Fig. 36 is normalized to illustrate phase distortion, and therefore does include amplitude modulation which is also present.

The effective vertical recorded angle of the groove modulation may differ considerably from the design angle of the recorder as determined by the internal geometrical configuration of the cutter of Fig. 35. A major portion of this difference in angles may be attributed to bending of the recording stylus resulting from the drag force produced by the record material being removed from the groove, as shown in Fig. 37, while a minor portion may be attributed to longitudinal lacquer springback. The two effects introduce essentially the same type of tracking angle deviation.

A large number of nonlinear distortion tests have been made with various geometrical tilt angles of the stereodisk records. A series of playback distortion measurements using a signal input to the cutter of the combination of a 400 cps tone and a 4000 cps tone, with a geometrical vertical angle of the cutter of 33° and various pickup vertical tracking

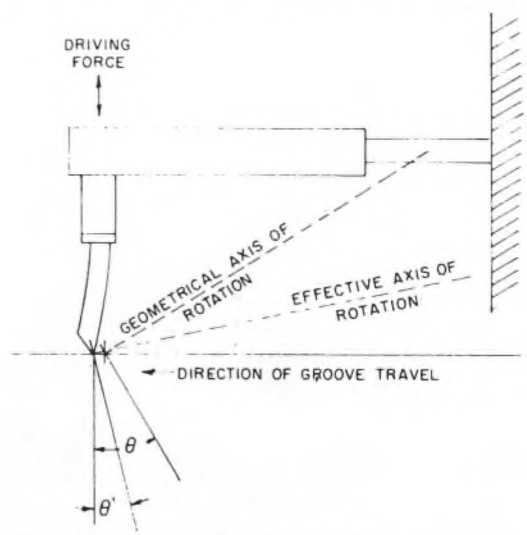

FIG. 37. Diagram showing the geometrical axis and effective axis of rotation of a stereophonic disk recorder.

[32] J. G. Woodward and E. C. Fox, "A Study of Tracking-Angle Errors in Stereodisk Recording," *IEEE Trans. on Audio* **AU-11,** 56 (1963).

[33] B. B. Bauer, "The Vertical Tracking Angle Problem in Stereophonic Record Reproduction," *IEEE Trans. on Audio* **AU-11,** 47 (1963).

angles, are shown in Fig. 38. The results of the particular test of Fig. 38 show that a pickup with a vertical tracking angle of about 15° will yield the lowest nonlinear distortion.

The data on distortion depicted in Fig. 38 were obtained without the use of the Dynamic Styli Correlator. When the correlator is used, the distortion at the nulls will be reduced further by the amounts shown in Fig. 33.

The tentative standard angle for stereophonic disk records is 15°. Accordingly, a suitable nominal geometrical tilt angle was introduced in the Westrex Recorder and other modifications of the cutter were incorporated so that an effective angle of 15° was cut in the master record.

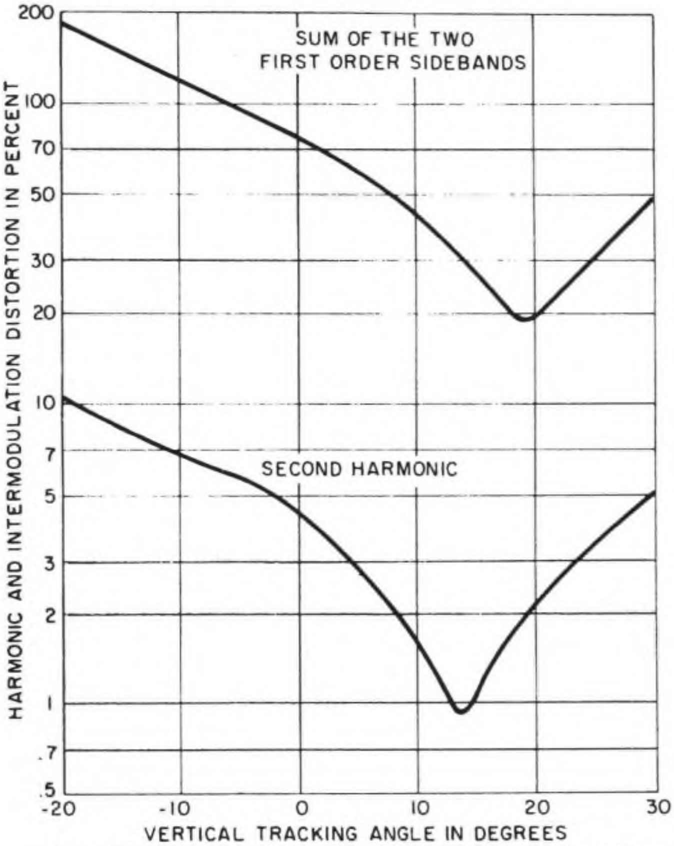

FIG. 38. Playback distortion measurements of a 400 + 4000 cps test signal recorded by a Westrex 3-C Recorder with a 10° wedge. Peak vertical velocity of 7.0 cm/sec at 400 cps and 1.76 cm/sec at 4000 cps. Groove diameter is 7.8 in.

STEREODISK CUTTER AMPLIFIERS

A consideration of the amplifiers used for driving the stereodisk cutter indicated that the maximum power available was very close to the required peak power. As a result there was some evidence that distortion was introduced in this link. Accordingly, the power output of the stereodisk cutter driving amplifier was increased to the point where there would be no appreciably nonlinear distortion from this source.

LACQUER ORIGINAL RECORDING CONSOLE

A recording console has been designed for recording the lacquer original from the submaster tape. Since the submaster tape is recorded in a manner designed for transfer to the original lacquer record without any appreciable modifications except for those introduced by the Dynamic Styli Correlator, the manual controls are comparatively simple. The lacquer original recording console includes the Dynamic Styli Correlator and the Recording Overload Indicator. The latter is included to check on the signal applied to the lacquer record.

LACQUER ORIGINAL

The preceding sections have outlined the development of elements which will produce the most nearly correct motion of the stylus that cuts the groove in the lacquer original and thereby determines the final disk record. In the course of a study of the cutting process with the main objective of reducing the noise in the lacquer original, the shape of the cutting stylus was examined from this standpoint. A review of the operation of the heated stylus was also made, with the main purpose of maintaining constant temperature. The indications are that the smoothness of the groove depends to a major degree upon the temperature of the cutting stylus tip. The problem of reducing the "horns" in the lacquer original was also given considerable attention. As a result of these investigations, there has been a significant improvement in the quality of the lacquer original.

DISK RECORD

The minimum number of significant steps between the original lacquer and the final disk record were shown in Fig. 1*d*. Intensive research and development of the plating processes which lead to the final stamper carried on during the past five years, combined with quality control in the factory, has led to a marked improvement of the stamper. Furthermore, studies of the stamping process resulted in the placing of very rigid controls on the cycling and temperatures. Studies of the plastic for the production of the disk record were carried out in a joint research program with the manufacturer of the plastic. One significant development was the new electrical conducting plastic[34] which helps to keep the record dust-free and thereby reduces surface noise: everyone is familiar with the attraction and accumulation of dust on conventional records due to the tremendous electrostatic charges and resultant voltages developed on the surface of the records. As a result of the research and development in the production of records, the signal-to-noise ratio of the surface of the commercial DYNAGROOVE records is now more than 65 db.

REPRODUCTION OF SOUND

Reproduction in the Concert Hall

In the reproduction of the sound of a musical aggregation, the first problem is faithful reproduction of the original sound under the same acoustical conditions as for the orig-

[34] G. P. Humfeld, "Control of Static Electricity on a Phonograph Record," *J. Audio Eng. Soc.* **10**, 290 (1962).

inal rendition. The second problem is to provide the listener in his home with a realistic reproduction of the original rendition. If the first cannot be solved, the second, obviously, cannot be solved either.

Successful solution of the first problem above was demonstrated on July 29, 1947 at the Berkshire Festival in Tanglewood, Massachusetts, when the full Boston Symphony Orchestra was compared with the reproduction of a phonograph record of the orchestra. The demonstration took place before an overflow audience in the Music Shed at Tanglewood, Massachusetts.

The last four minutes of Beethoven's "Egmont" were recorded by the Boston Symphony Orchestra with Serge Koussevitsky conducting. In the demonstration, the Boston Symphony Orchestra, with Serge Koussevitsky conducting, played the preceding portion of the selection; then a switch was made from the original sound to the reproduced sound. Many rehearsals were carried out to insure continuity of the music as well as of the amplitude level during the switch from live to reproduced music.

Comments from both music critics and laymen were that the tone color, dynamic range and general fidelity of the reproduced sound matched the live sound so closely as to be practically indistinguishable from the original sound. It is beyond the scope of this paper to give the details of the test and equipment and the comments of the music critics, which have been published elsewhere.[35]

The point of the above test is that equipment which represented the advanced state of the art in the laboratory at that time, operating under the same acoustical environment as the orchestra, gave a very realistic reproduction of the original sound. Incidentally, the laboratory equipment used in the Tanglewood tests is now commercially available, with some additional refinements resulting from developments during the intervening 15 years.

Reproduction in the Home

From the preceding description it will be seen that the fidelity of performance of high-grade sound reproducing equipment is completely adequate for the reproduction of music with realism under the acoustical conditions of the original rendition. However, the crux of the entire matter is that the listener does not listen to reproduction of the disk records in the concert hall—he listens to the reproduction of the records he buys in his own home. The main objective of the DYNAGROOVE development has been to provide the listener in his home environment with sound reproduction which exhibits the highest order of artistic and subjective resemblance to that of the original rendition in the concert hall.

Subjective tests of the system were performed in many listening rooms which simulate the acoustics of living, recreation and music rooms in homes. All of these listening rooms are equipped with disk record reproducing equipment of high quality termed "reference sound reproducing systems." A block diagram of the reference sound reproducing system is shown in Fig. 39.

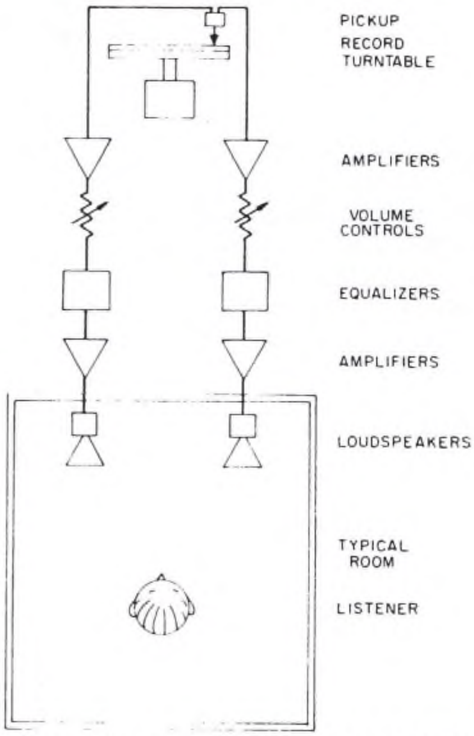

FIG. 39. Diagram depicting the elements of the reference sound reproducing system.

The loudspeaker mechanism used in the reference sound reproducing system is the RCA LC1A mounted in the RCA LS11A cabinet. The design of the cabinet had the objective of reducing the deleterious effects of diffraction. The response frequency characteristic of the loudspeaker is shown in Fig. 40. Another important loudspeaker characteristic in stereophonic sound reproduction is the directivity pattern, which should be independent of the frequency over the listening area in order to provide faithful auditory perspective. The directivity characteristics of the LC1A for four frequencies are shown in Fig. 41. Over a total angle of 90°, which covers more than the listening area, there is no significant frequency discrimination due to the directivity characteristic. The nonlinear distortion is another important loudspeaker characteristic. As mentioned in the section on the Dynamic Spectrum Equalizer, the peak sound level of reproduction in the home is 80 db. The

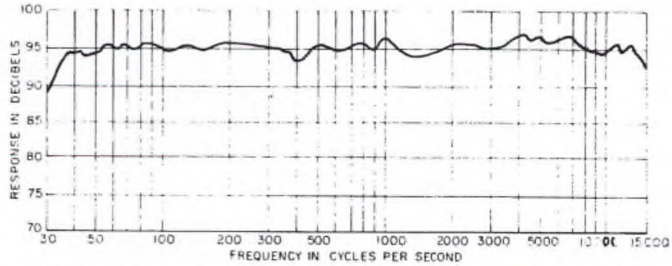

FIG. 40. Response frequency characteristic of the RCA LC1A loudspeaker mechanism mounted in the RCA LS11A cabinet.

[35] H. F. Olson, *Acoustical Engineering* (D. Van Nostrand Company, Princeton, N. J., 1957).

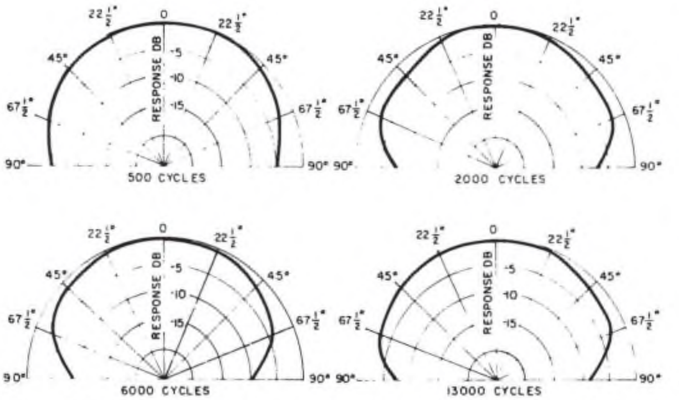

Fig. 41. Directivity characteristics of the RCA LC1A loudspeaker for four different frequencies.

RCA LC1A loudspeaker will deliver sound levels of 80, 90 and 100 db in a typical living room for electrical inputs of .05, .5 and 5 w respectively. The nonlinear distortion frequency characteristics of the RCA LC1A loudspeaker for inputs of .05, .5 and 5 w are shown in Fig. 42. This nonlinear distortion is very low. If the distribution of the components in music with respect to frequency is considered, the resultant distortions will be imperceptible even for a level of 100 db.

The amplifier used in each channel of the system exhibits less than 0.1% nonlinear distortion up to 20 w output. The response is uniform to within a fraction of a decibel over the frequency range of 30 to 15,000 cps.

The response frequency characteristic of the RCA MI 11866-7 phonograph pickup at the output of the amplifier, including the conventional magnetic pickup compensation, is shown in Fig. 43. The cross-talk separation between channels is also an important consideration. Subjective tests have shown that there is no deterioration in auditory perspective if the separation between channels is more than

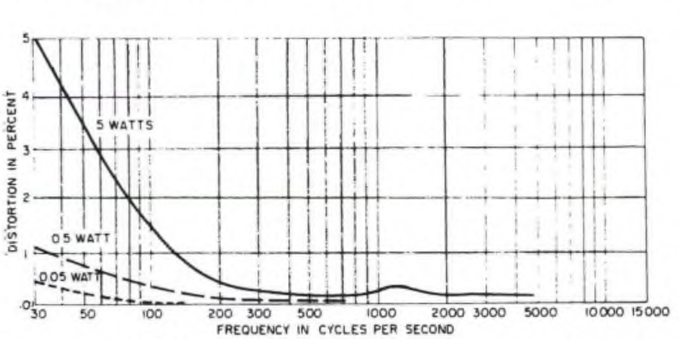

Fig. 42. Nonlinear distortion frequency characteristic of the RCA LC1A loudspeaker for three different electrical power inputs.

20 db between 100 and 8000 cps and more than 10 db at the extreme ends of the frequency range, 30 and 15,000 cps. A consideration of the characteristics of Fig. 43 shows that the cross-talk characteristic is more than adequate to insure stereophonic reproduction with good auditory perspective.

The turntable used in the reference system is the RCA BQ-2C. The wow and flutter are less than .25% peak to peak or .09 rms.

The above description of the characteristics of the reference sound reproducing equipment indicates a high order of performance of sound reproduction.

In addition to the subjective tests on the reference equipment, both objective and subjective measurements were

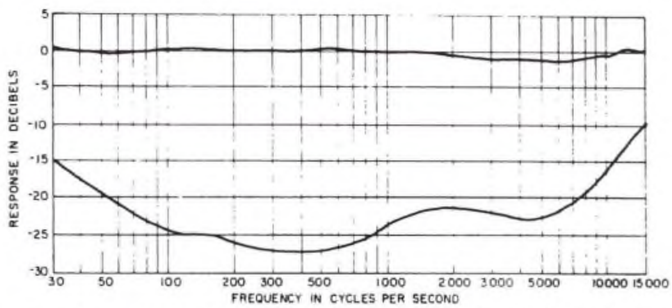

Fig. 43. Response frequency characteristic of the RCA MI 11866-7 pickup. The lower curve is the cross-talk in the other channel.

carried out on all manner of commercial phonographs. The general conclusion was that the original objective has been achieved, namely, to provide the listener in his home environment with sound reproduction which exhibits the highest order of artistic and subjective resemblance to that of the rendition in the concert hall.

ACKNOWLEDGMENTS

The DYNAGROOVE concept was initiated, developed, directed and implemented by George Marek, Vice President, RCA Record Division. In a project of this magnitude it is impossible to list all who contributed to make it a success; however, it should be mentioned that many of the listening tests were conducted in other divisions of RCA as well as by RCA distributors and dealers. Those intimately concerned with the artistic elements and tasks of the DYNAGROOVE project include John Pfeiffer and J. A. Somer of the RCA Record Division. Those intimately concerned with the scientific elements and tasks of the DYNAGROOVE project include R. L. McClay, H. E. Roys, and D. L. Richter of the RCA Record Division and J. G. Woodward, John Volkmann, E. C. Fox and R. W. George of the RCA Laboratories.

A Brief History of the Recording Industry in the Soviet Union*

A. I. ARCHINOV

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FOREWORD

During the Summer of 1969, John Woram of RCA Records Division went to the Soviet Union under the auspices of the Citizen Exchange Corps. The Citizen Exchange Corps is a non-profit organization specializing in people-to-people contacts at the nondiplomatic level. The group assists Americans in meeting Soviet citizens with similar occupations and interests. While there, he visited recording studios and talked shop with Soviet engineers including Boris P. Lebedev, General Secretary of the Committee for USSR Participation in International Power Conferences, and P. Vasilevsky, Senior Engineer of the Soviet Agency for Radio and TV Broadcasting. In the Spring, Mr. Lebedev had received the Audio Engineering Society's invitation to participate in the 37th AES Convention. As a result he and John talked at length about the advantages of active participation in the society. In conclusion, Mr. Lebedev said he would seriously consider sending representatives of the Soviet recording industry to New York in the Fall. Negotiations for the visitors continued through the summer, and shortly before Convention time, A. I. Archinov, Chief Engineer of the Gramophone Recording Studio of the USSR (Melodia) and Dmitry B. Katzman, of the Melodia Research Institute, arrived in New York City. They were joined by A. A. Khruschev, Director of the Sound Technique Division of the Cinema and Photo Research Institute, and F. F. Provorov, Director of Photography, Mosfilm Studios, who were already in the United States to attend the SMPTE Convention on the West Coast.

At the Audio Abroad III session, Messrs. Archinov and Khruschev both presented papers. Mr. Khruschev's paper was also given at the SMPTE convention earlier in the month in Los Angeles.

Although the manuscripts are the work of their authors, John Woram assumes responsibility for any lapses in clarity, since considerable editing was done to transform the papers from literal translations into what is, he hopes, a more easily read format for English language readers. The editors are most grateful to Mr. Woram for his help.

* Presented October 16, 1969 at the 37th Convention of the Audio Engineering Society, New York.

DEVELOPMENT OF THE SOVIET RECORDING INDUSTRY

The technique of mechanical sound recording has a relatively short history. Less than 100 years have passed since the invention of the first phonograph by the American inventor, Thomas A. Edison, in 1877. And only 50 years have passed since the birth, in 1919, of the Soviet gramophone industry. Having been employed in this field since 1932, I am familiar with its progress from the early days up to the present.

Soviet record production now amounts to about 200 million records per year, with volume of some popular records reaching more than one million copies. In the early years the entire production was only about 12,000 records a year.

In 1930, we changed from the acoustical to the electroacoustical system. At that time, we acquired a Neumann recording lathe with Telefunken electromagnetic Type ELP 5111 cutterhead. The nondamped resonance of this cutterhead was about 4000 Hz. Some damping was realized with the help of the wax recording discs. Of course, the frequency response deteriorated at the smaller diameters, but in those early days musicians and engineers were pleased with the recordings.

Our Telefunken mixing console had only two microphone inputs and our microphone complement consisted solely of the Neumann condenser microphone Type ELA M14. With this equipment, we travelled throughout our large country, making recordings of the music and songs of the many nationalities of the USSR. The heavy recorded discs were sent back to Moscow for processing. The popularity of the phonograph record increased rapidly, and we were unable to keep up with the demand.

Recording productivity was very high. At a 3 to 5 hour session, we were able to record up to 40 minutes of music. Now, with the help of very clever and advanced techniques, we are able to produce less than half that much music, which later requires even more time to edit.

In 1936, I visited the United States with a group of our specialists to study American recording techniques at the RCA studios and factory. At that time, RCA used small studios, ribbon microphones, and simple mixing consoles. They used wax blanks similar to ours, bronze powder for surface metalization of the recorded waxes, and shellac for record pressings. RCA was then experimenting with the so-called flow waxes, which had a fine grain structure, and produced a quieter surface. These flow waxes were metalized with the help of gold sputtering in vacuum; Vinylite was used for pressings made by this process. Those discs were of superior quality, and were used only for broadcasting.

With the permission of the directors of Columbia, Decca, and Audio Devices Corp., we became acquainted with their facilities. At that time, Audio Devices was producing lacquer blanks, and Columbia had developed chemical silvering of lacquers and high-speed cathode rotating nickel baths. It was then an advanced technical technology.

We left America with many new ideas. Later, our industry purchased RCA ribbon microphones, Scully cutting machines, lathes for grinding sapphires, and much other recording equipment. Unfortunately, our contacts with American companies ended after our visit.

In 1939, the Soviet gramophone company, Gramplastrest, built a nine-story sound recording building with four concert and dramatic studios and with up-to-date equipment. A sound research laboratory was also organized, which later became the Sound Recording Institute (VNIIRT). The Institute publishes the journal *Transactions of the VNIIRT.*

After World War II, the Soviet recording industry developed at a faster pace. In 1950, we began to produce long-playing records, and in 1960 stereo records were introduced. At the present time, the entire gramophone record industry in the Soviet Union is united under the label Melodia. Melodia has four departments: head office, recording studios, record factories, and central storehouses. Studios are in Leningrad, Riga, Vilnius, Tallinn, Erevan, Alma-Ata, Tashkent, and Kiev, as well as the central studio in Moscow. Every year we make about 600 hours of new recordings on magnetic tape, which are then transferred to about 2000 new disc recordings. At the present time, all tape-to-disc transfer work is done in Moscow, where we have four complete transfer facilities. We use Ortofon and Neumann lathes, with automatic pitch and depth control. For mono recordings, we use Ortofon cutterheads, Type DSS 522, and for stereo, Type DSS 601. We plan to buy a new Neumann SX 68 cutterhead, which has excellent parameters.

We have five pressing plants, with a sixth now under construction in the Ukrainian Republic. This new plant will have a capacity of 40 million records per year.

For the conductive coating of lacquers, we have tried many processes—graphitizing in 1932, chemical silvering in 1934, then vacuum evaporation of gold and silver, and finally, a return to chemical silvering using a spray process. We intend to replace all manual work in silvering with a fully automated process. For this purpose, we have designed and built a special silvering chamber which is showing favorable results. For the metal fathers, mothers and stampers, we began with a pure copper process. Then we passed to nickel-copper, and now we have a pure nickel process using Russian, Danish, and Swedish baths. We produce about 500 fathers, mothers, and stampers every day. In most cases, the mothers need small engraving repairs and about 5 to 10% require rerecording. Most defects are a result of manual operations, and we would like to see a fully automated process that would help minimize these defects.

Most of our records are produced on Russian steam presses. We also have some Alpha presses from Sweden. For experimental work, we plan to buy some automatic injector machines.

Like all record companies, we began at 78 rpm, passing later to 33 and 45. From our point of view, there is no need for the 45 rpm format, and we use it only for the purpose of international exchange. Perhaps it is now time to exclude this speed from the standards.

On the subject of standards, we regret the lack of an international standard for measurement records. We are now developing our own Russian standards, and would welcome international recommendations.

SOVIET RECORDING TODAY

Today, Soviet recording practices are essentially similar to European techniques. Our specialists maintain close contacts with major European firms. We regret, however, that until the Fall 1969 AES Convention, we have had little contact with our American counterparts.

Our recording practice differs somewhat from that in the United States. Our recordists are usually musicians with some knowledge of engineering techniques, as opposed to your practice of a technically trained person with perhaps some knowledge of music. We believe our method is superior, since we find it easier for a musician to establish balances than to require it of an engineer.

For recording classical music, we prefer to use the best of our concert halls rather than a regular recording studio. At several of the better halls, we have installed permanent recording equipment. The best recording locations are the big hall of the Moscow Conservatory, the Leningrad Philharmonic Hall, and the opera house in Kiev. Some years ago, Mercury Records visited Moscow for a recording at the Moscow Conservatory with Bryon Janis. At that time, Mercury engineers were very pleased with the acoustics.

For modern light music, we are now using dead studios and close-up microphone placement. We are beginning to use more microphones per session than we did in the past. After much experimentation, we now prefer to use only condenser microphones, such as the Neumann M-269, U-87, and SM69. We also like the AKG C-12A and the new FET Type C-451.

We have no strict rules regarding microphone and instrument placement. Every producer follows his particular tastes and practices. Nevertheless, our producers regularly demonstrate their techniques to their colleagues, and we encourage an interchange of ideas, both among ourselves and with other countries. Many Moscow producers have won international prizes.

Consoles

The modern mixing desks now in use are rather complicated devices. During the past three to four years, we have replaced all our equipment with modern W.S.W.

and Studer consoles, as well as with Russian equipment. A typical mixing desk may contain 20 inputs and four outputs. We utilized the latest frequency correctors, presense filters, compressors, limiters, and noise-ex. We are able to produce recordings in A-B, X-Y, or M-S formats, as well as mixed systems. The mixing desk is a critical location in the technological chain, where one can produce fine or, what is much easier, poor recordings. The sound recording process is a creative process, and the debates, arguments, and discussions about all aspects of recording must continue. The art is further complicated by the need for stereo recordings to be fully compatibile on mono systems. We now produce a relatively high percentage of monoural products, and of course the compatibility problem will only cease to exist when all recording and broadcasting is in stereo.

In our practice, classical sessions are recorded on two tracks at 38 cm/sec (15 ips). Light music is recorded on four-track 1 in. tape. We use either Agfa 555 tape or Russian Type 6, which also gives good results.

Even the most professional artists are rarely satisfied with their recorded performances, and often will make many takes of a given work, which are later edited and spliced together. At first, we saved only these spliced master tapes, which eventually became spoiled as the splices deteriorated with age. Now, all producers make a splice-free copy which is filed with the original.

In 1964, Melodia began to manufacture flexible records for the sound magazine, Crugozor (Horizon). These discs are very popular in our country, and we issue 60 million of them each year. Many companies don't produce these records because of their relatively poor quality; however, recent achievements in the production of PVC plastics and improved pressing methods are now giving good results.

LOOKING AHEAD

We are very interested in new developments in the art of recording, and look forward to reading about American developments. For example, there was an interesting article by John M. Woram in the May and June issues of *Audio Magazine*, describing his practice of recording light music. Mr. Woram did not mention that the method described is not a really stereophonic one. It is, rather, a blending of monaural tracks to produce a pseudo-stereo image. Practically, such a method gives good stereophonic results, but it is unsuitable for classical music. I should like to see someone write an article about classical stereophonic recording.

Here are some other questions to which we hope the AES Journal will someday publish the answers:

Recording work in the U.S. is now almost exclusively stereophonic. The problems of stereo–mono compatibility are still not all solved. In the June issue of *J. AES*, John Eargle presented a survey of the compatibility problems. Why do American engineers prefer the A-B system, which is theoretically incompatible, to the compatible X-Y, or M-S technique? It might be time for some standardization of the parameters of compatible recording.

I notice that there are a variety of loudspeakers in use within the recording industry. It would be advisable if a standard speaker were prescribed, so that there would be some uniformity between studios.

There is a vast archive of artistically valuable old recordings that suffer from poor sound quality. I would like to see some papers describing the various methods used to improve the quality of old recordings.

We feel it is now time to exclude the 45 rpm format from standards, as we do not think it is necessary to have both 33 and 45 rpm speeds.

In our archives, we have more than 150.000 metal discs. We find that many of these discs deteriorate with age. To minimize this, we apply a thin nickel layer to the face of the metal disc. We would like to know what techniques are used by other firms.

CONCLUSION

From my short account, I hope I have given some idea of the Soviet recording industry and its history from its earliest days. We have many specialists who know and enjoy their profession. Records give our people ready access to great works and afford much esthetic enjoyment. We wish the gramophone record a long and healthy life, and hope that our first meeting with the Audio Engineering Society will be the beginning of a long and mutually profitable professional relationship.

THE AUTHORS

A. I. Archinov, center, talks to Society members at 37th AES Convention. A. A. Kruschev, whose article on sound systems appears on the opposite page, can just be seen in conversation with John Woram, right.

Methods of High-Quality Recording and Reproducing of Music and Speech Based on Telephone Research*

J. P. MAXFIELD AND H. C. HARRISON

Bell Telephone Laboratories

The paper deals with an analysis of the general requirements of recording and reproducing sound, with the nature of the inherent limitations where mechanical records are used, and a detailed description of a solution involving, first, the use of electrical equipment for the purposes of recording and, second, the use of mechanical equipment based on electric transmission methods for reproducing.

Probably the most useful feature of the paper is the complete description of the application of electrical transmission theory to mechanical transmission systems. A detailed analysis is made of the analogies between the electrical and the mechanical systems.

Guest Editor's Note

This paper described one of the first examples of the advantages of replacing mechanical impedances in the analysis of vibrating systems used as transducers for the generation and reception of sound radiation. It made possible the orthophonic phonograph and had a great influence on the development of the entire audio industry.

The authors were early pioneers in the concept of an optimum reverberation time for various recorded materials as well as in developing the shapes of the room and character of the damping surfaces.

Joe Maxfield died in 1977 and all of us who were acquainted with him admired his great gifts and vivid personality. He was especially helpful to me in guiding my early efforts in sound motion picture recording around 1930.

John K. Hilliard

INTRODUCTION: The problem with which this paper is concerned, in its broadest sense, may be stated as that of taking sound from the air, storing it in some permanent way, and reproducing it again without appreciable distortion. It is immaterial from the general standpoint whether the means used are mechanical or electrical or a combination of the two. The choice of which method to use will depend largely upon the commercial requirements accompanying the specific purpose for which the reproduction is being made. For instance, it is quite probable that the means chosen for reproduction in residences would differ materially from those used in large ballrooms or in the presentation of synchronized motion pictures.

Before considering the methods and results referred to in the title of this paper, it may be well to make a rough division of the problem. The storing or recording of sound requires, first, a mechanical system which will respond faithfully to the sound waves that are to be recorded. Then there is required some material in or on which this sound may be recorded and an intervening system which permits the sound waves to make the record in this material. In the usual case, and in that with which we are particularly concerned here, there is a mechanical system that will vibrate in response to the sound which is to be recorded and directly through some mechanical linkage or less directly through an electrical linkage, drive a cutting

* Presented at the Midwinter Convention of the A.I.E.E., New York, NY, February 8–11, 1926. Reprinted with permission from the *Transactions of the American Institute of Electrical Engineers*, vol. 45, pp. 334–348 (Feb. 1926).

mechanism which will impress a wax record.

The first consideration, therefore, is the character of the sound which is to be recorded, including all of the effects of reverberation and the general questions of studio design. Next to be considered is the manner in which the cutting instrument shall impress this speech or musical record upon the constantly rotating wax disk, which disk is commonly called the wax master. In this connection, there will be discussed also the relative value of the electrical and mechanical linking of the cutting knife with the mechanism that receives the sound waves. Following the discussion of these problems and a brief reference to the state of the prior art, there remains to be considered the reproduction of the sound which is stored in the cuts or grooves of the wax record.

In the case of reproduction, also, there is required a mechanical system which will respond to these cuts in the wax and a system which will set up in the air sound waves essentially identical to those picked up by the first mechanism of the recording system. Between these two systems, a mechanical linkage intervenes in the case under discussion, but reference is made to the relative advantages of this system compared with the use of an electrical linkage.

First to be described is the character of the sound which is to be recorded and reproduced and the effects of reverberation and transients upon the listener's sensation of this sound.

STUDIO CHARACTERISTICS AND TRANSIENTS

Phonographic reproduction may be termed perfect when the components of the reproduced sound reaching the ears of the actual listener have the same relative intensity and phase relation as the sound reaching the ears of an imaginary listener to the original performance would have had. Obviously, it is very difficult, if not impossible, to fulfill all of these requirements with a single-channel system, that is, with a system which does not have a separate path from each ear of the listener to the sound source.

The use of two ears, that is, two-channel listening, gives the listener a sense of direction for each of the various sources of sound to which at a given moment he may be listening, and, therefore, he apprehends them in their relative distribution in space. It has been found possible with a single-channel system, however, by controlling the acoustic properties of the room in which the sound is being recorded, to simulate to a considerable degree in the reproduced music the effective space relationships of the original. In this case, with a one-channel system, the directional effect is, of course, entirely absent, and the spatial relationship which is apprehended is probably due to the increased apparent reverberation of the instruments situated at the far end of the room as compared with those in the near foreground.

In recording work, therefore, one of the important acoustic characteristics of a room is its time of reverberation. Although it is probable that this is the most comprehensive single factor, experiment has shown that the shape of the room and the distribution and character of the damping surfaces play a part in the excellence of music in such a room.

It has been shown by Sabine [1] that for piano music, studios should have a time of reverberation measured by his method of 1.08 seconds. Experience has indicated that this figure is also very closely correct for other types of music. This figure of Sabine's assumes binaural listening. With single-channel systems, such as most of the present reproduction systems, whether for radio or the phonograph, the ability of the listener to separate the reverberation from the direct music by means of the sense of direction is completely removed and there is thrust upon his attention an apparently excessive amount of room echo. Experiment has shown that a time of reverberation for the recording room ranging from slightly more than ½ to slightly less than ¾ of Sabine's figure affords in the reproduced music the effect of a room with proper acoustics. When this effect is accomplished, the person listening to the reproduced music has the consciousness of the music being played in a continuation of the same room in which he is listening and also has a sense of spatial depth.

Experiment has indicated further that any transients set up by the recording or reproducing system constitute a second cause of apparent increased reverberation. The data obtained thus far are insufficient to permit assignment of quantitative values to the importance of these two factors.

At the present state of the art, the most important requirement of a recording or reproducing system is its frequency characteristic. This involves two factors—intensity versus frequency, and phase distortion versus frequency. The effect of the second of these factors is not thoroughly understood, but as it is closely related to the production of transients it has to be considered, as mentioned above. The system to be described is relatively free, however, from violent phase shifts within most of the range covered, but does have some undesirable phase-shift characteristics with small accompanying transients near its limiting cut-off frequencies.

FREQUENCY REQUIREMENTS

The frequency range which it would be desirable to cover, if it were possible, with relatively uniform intensity for the transmission of speech and all types of music including pipe organ is from about 16 Hz to approximately 10 000.

It may be interesting to examine the record requirements for a band of frequencies this great. For the purpose of this illustration, a lateral-cut record will be assumed, although in all the factors, except the time which the record will run, the arguments apply in a similar manner to the hill-and-dale cut. Since, for mechanical reproduction, the sound at a given pitch is radiated by means of a fixed radiation resistance, it is necessary that the record be cut with a device the square of whose velocity is proportional to the sound power. Under these conditions, it is seen that for a given intensity of sound the amplitude is inversely proportional to the frequency of the tone, and that a point will be reached somewhere at the low end of the sound spectrum where this amplitude will be great enough to cut from one groove into the adjacent groove, or in case of vertical cut, to cut so deeply that with present materials the wax will tear instead of cut away with a clean surface. This means that there is an inherent maximum amplitude beyond which it is not commercially feasible to go. Similarly the minimum radius of curvature of sine waves of various frequencies cut at constant velocity is inversely proportional to the frequency, so that as higher and higher frequencies are reached, the radius of curvature becomes smaller and smaller until finally it becomes too small for the reproducing needle to follow. There is, therefore, an inherent limit at the upper end.

In order to extend these limits, it is necessary in the case of the low end to make the spiral coarser and in the case of the high end to run the record at a higher speed. Both of these changes tend to decrease the time which a record of a given size can be made to play. The only alternative of these methods is to cut the record less loud than is the present standard practice and make the reproducing equipment more sensitive. This could be done easily if it were not for the "record noise" or "surface noise," as it is commonly called. Since this surface noise is already loud enough in comparison with the reproduced music to be somewhat objectionable, no appreciable gain in this direction can be made until the technique of record manufacture has been distinctly improved.

In this connection there is one other interesting point. It has been suggested that if electric reproduction were used, it would be possible to cut the record with a characteristic other than uniform velocity sensitiveness and correct for the error by an electrical system whose characteristic is the inverse of the characteristic of record. If the change which is made in the recording characteristic tends toward cutting at uniform acceleration sensitiveness, the amplitude varies inversely as the square of the frequency, and hence the difficulties at the low end of the scale are greatly enhanced. Similarly, if the records are cut more nearly at constant amplitude, the radius of curvature of the sine waves decreases as the square of the frequency, hence the difficulties are placed at the upper end. In the process which is being described in this paper, these limitations have been met commercially by having a frequency characteristic of the uniform velocity type between the frequencies of 200 and approximately 4000 Hz. Below 200 Hz it has been necessary to operate at approximately constant amplitude with a resulting loss in intensity which loss increases as the frequency decreases. Above 4000 Hz it has been necessary to operate at approximately constant acceleration with its consequent slight loss in intensity at the very high overtones. With a characteristic of this type, a range of frequencies from 60 Hz to 6000 Hz can be recorded with reasonable success, although the very low and very high range are slightly deficient (see Fig. 14). With a record having such a frequency characteristic, the inherent limitations are divided between the two ends of the frequency band, and where electrical reproduction methods are used, it is possible to employ a reproduction system whose frequency characteristic compensates for that of the record.

It should be pointed out that an attempt to record notes lower than the low cutoff of the above-mentioned apparatus would result in recording only those harmonics of the notes which lie above the cutoff. This in no way prevents the listener from hearing the notes, reproduced by means of the harmonics only, as notes with the pitches of the missing fundamentals, although it does somewhat change the quality of the tone. If it were not for this ability of the ear to add the fundamental pitch of a note, of which only the harmonics are being reproduced, most of the older phonographs and loudspeakers would have been totally useless for the reproduction of speech and music.

MECHANICAL VERSUS ELECTRICAL RECORDING

In attacking the recording part of the problem, two ways at once present themselves: first, the direct use of the power, of the sound being recorded, to operate the recording instrument; and second, the use of high-quality electric apparatus with vacuum-tube amplifiers in order to give more freedom to the artists and better control to the process. The amount of power available to operate the recorder directly from the sound in the recording room is so small as to make it extremely difficult to make records under natural conditions of speaking, singing, or instrumental playing. As the use of high-quality electric apparatus with associated amplifiers has a very distinct advantage over the acoustic method, they have been adopted for the recording part of the process. Fig. 1**a** shows a group of artists recording by means of the sound power directly, while Fig. 1**b** shows a record being made by the same artists with the electric process.

It will be noticed in Fig. 1**a** that the artists are grouped very closely about the horn. In the case of the weaker instruments such as violins, it has been possible to use only two of standard construction. The rest of the violins are of the type known as the "Stroh" violin which is a device strung in the manner of a violin but so arranged that the bridge vibrates a diaphragm attached to a horn. This horn is directed toward the recording horn, as shown by the player in the foreground.

With such an arrangement of musicians it is very difficult to arouse the spontaneous enthusiasm which is necessary for the production of really artistic music.

In Fig. 1**b** the musicians are sitting at ease more nearly in their usual arrangement and all are using the instruments which they would use were they playing at a concert. Furthermore, the microphone is now sufficiently far away from the orchestra to receive the sound in much the manner that the ears of a listener in the audience would receive it. In other words, it picks up the sound after it has been properly blended with the reflections from the walls of the room. It is in this way that the so-called "atmosphere" or "room tone" has been obtained.

In the old process it sometimes happened that after the instruments had been arranged in such a manner that the relative loudness of the various parts had been balanced correctly, it was found that the whole selection was either too loud or too weak. This usually meant a complete rearrangement of the players. With the flexibility introduced by the use of electrical apparatus including amplifiers, the control of loudness is obtained by simple manipulation of the amplifier system and is in no way related to the difficulties of the relative loudness of one instrument to another. The only problem for the studio director in this case is to obtain the proper balance among the various musical instruments and artists. The advantages derived from this added ease of control are also made manifest in that it is much easier and less tiresome for the artists, and it is usually possible to make more records in a given time.

MECHANICAL VERSUS ELECTRICAL REPRODUCING

Where the question of reproduction is concerned, the same two alternatives mentioned for recording present themselves, namely, direct use of power derived from the record itself versus the use of electromechanical equipment with an amplifier. In this case, however, the situation is a little different as the power which can be drawn directly from the record is more than sufficient for home use. Since any method of reproducing from mechanical records by electrical means involves the use of a mechanical device for transforming from mechanical to electrical power and a second such device for transforming from electrical back to mechanical power, that is, sound, it is necessary to use two mechanical systems, one at each end of an electrical system. Where the power which can be supplied by the record is sufficient to produce the necessary sound intensity, as in the case of home use, it is in general simpler to design one single mechanical transmission system than it is to add the unnecessary complications of amplifiers, power supply, and associated circuits. In cases where music is to be reproduced in large auditoriums, the power which can be drawn from the record may be insufficient and some form of electric reproduction using amplifiers becomes necessary.

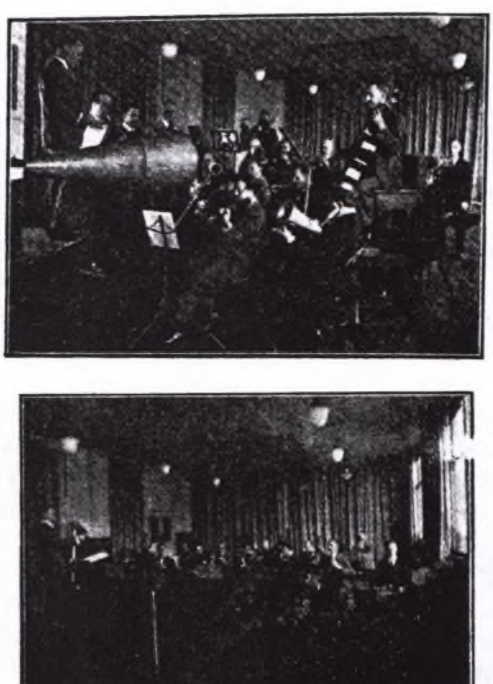


Fig. 1**a**. Orchestra recording. Acoustic process. **b.** Electric process. *(Photo courtesy of the Victor Talking Machine Company, Camden, NJ.)*

BRIEF DESCRIPTION OF RECORDING SYSTEM

The system used for recording consists of a condenser transmitter, a high-quality vacuum tube amplifier, and an electromagnetic recorder. Fig. 2 shows the calibration of the condenser transmitter and the associated amplifiers. The condenser transmitter and amplifiers are so designed that the current delivered to the recorder circuit is essentially proportional to the sound pressure at the transmitter diaphragm. The electromagnetic recorder, which will be described later, is designed to work with this type of system. With the exception of this electromagnetic recorder, apparatus of this type has already been described in the literature [3], [4]. In addition to this equipment which might be called the recording amplifier system, there is a volume indicator for measuring the power which is being delivered to the recorder and also an audible monitoring system. The audible monitoring system consists of an amplifier whose input impedance is high compared with the recorder impedance and a suitable loud-speaking receiver. The monitoring amplifier is bridged directly across the recorder and operates the loud-speaking receiver so that the operator may listen to the record as it is being made.

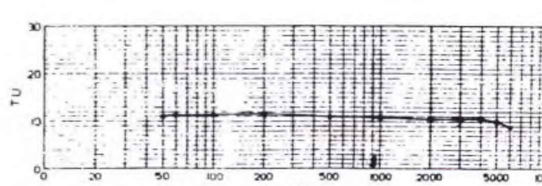


Fig. 2. Calibration of the condenser transmitter and associated amplifiers. This curve shows the relative frequency sensitiveness of the system, the zero line having been chosen arbitrarily.

In the design of the recording and reproducing systems each part of the system has been made as nearly perfect as possible. Errors of one part have not been designed to compensate for inverse errors in another part. Although this method is the more difficult, its flexibility, particularly as regards the commercial possibilities of future improvements, justifies the extra effort [4]. There is, therefore, no distortion in the record whose purpose is to compensate for errors in the reproducing equipment, the only intended distortion in the record being that required by the inherent limitations mentioned above (see Figs. 2, 14, and 20).

GENERAL BASIS OF DESIGN

An interesting feature of the development of the mechanical and electromechanical portions of the recording and reproducing system is their quantitative design as mechanical analogs of electric circuits. Both the recording and reproducing systems are good examples of the use of this type of analogy.

The economic need for the solution of many of the problems connected with electric wave transmission over long distances coupled with the consequent development of accurate electric measuring apparatus has led to a rather complete theoretical and practical knowledge of electrical wave transmission. The advance has been so great that the knowledge of electric systems has surpassed our previous engineering knowledge of mechanical wave transmission systems. The result is, therefore, that mechanical transmission systems can be designed more successfully if they are viewed as analogs of electric circuits.

While there are mechanical analogs for nearly every form of electrical circuit imaginable, there is one particular class of electrical circuits whose study has led to ideas of the utmost value in guiding the course of the present development. This class of circuits consists of infinitely repeated similar sections of one or more lumped capacity and inductance elements in series and shunt and are commonly known as filters. The study of filters began with the work of Campbell [5], and a recognition of their importance as frequency-selective systems in telephone repeaters, carrier systems, radio, signaling systems, etc., led to their intensive study. In the available literature is to be found a fairly complete statement of their properties and details of their design [5].

It will be recalled in the case of the telephone circuit that the introduction of inductance coils at regular intervals in the circuit produced a remarkable change in the transmission characteristic. Over a broad band of frequencies the attenuation was reduced and made fairly uniform over that range while beyond a critical frequency called the cut-off frequency the attenuation became very high. In the ideal filters with zero dissipation the transmission characteristics are of the same nature but more clear cut. Structure of this type with infinitely repeated sections will have one or more transmission bands of zero attenuation and one or more bands having infinite attenuation. The impedance characteristics of such a structure measured from certain characteristic points will be pure resistance more or less uniform in the transmission bands, and pure reactance in the attenuation bands. These terminations are mid-series, that is, the entering element being a series one of half the normal series element; or mid-shunt, that is, the entering element being twice the impedance of the normal shunt element. The corresponding impedances are called the mid-series and mid-shunt characteristic or iterative impedances.

If we retain the first few sections of such a structure and terminate them with a resistance which is equal to the resistance impedance of the infinite line from which they were taken, the characteristics are substantially unchanged. It is understood, of course, that this resistance equals approximately the resistance impedance of the remainder of the infinite line at most of the frequencies in the transmission band in which we are interested.

The presence of small amounts of damping in the various elements also has but slight effect on the general characteristics. These results could in general be readily applied to the various telephone transmission problems because the source and load between which the filter system was inserted generally had or could be made to have a nearly resistance impedance equaling the mid-series or mid-shunt impedance of the filter within the transmission band. The filter and terminating impedances may then be said to be matched. Where adjacent sections in the filter have impedances similar in character but different in absolute magnitude, they may be joined by a suitable transformer.

Many early attempts were made to design mechanical transmission systems having a wide frequency range in which highly damped single or multiresonant systems were employed. In these attempts both of the obvious methods of increasing the damping were used, namely, that of adding a resistance to the system and that of increasing the value of the compliance and decreasing mass in such proportion as to maintain the same natural frequency. The former of these methods reduces the sensitivity of the system at the point where it is most efficient (see Fig. 9), while the second method increases the response at the points where the system is less sensitive, namely, away from its resonance point. Fig. 9 shows four curves—first, a singly resonant system, curve A; second, the same system with friction added, curve B; third, the same system without the added friction but with an increase in compliance and a decrease in mass such that the natural period remains the same, curve C; and fourth, a band-pass type of circuit whose resistance impedance is the same as that of the system shown in curve A (see curve D).

The results of filter theory have shown how these resonances should be coordinated so that when a proper resistance termination is used, high efficiency and equal sensitivity are obtained over a definite band of frequencies by elimination of response to all frequencies outside the band. With the electrical case of a repeated filter, each section considered by itself resonates at the same frequency, but when combined into a short-circuited filter of n sections, there will be n natural frequencies. However, when such a system is terminated with a resistance that equals the nominal characteristic impedance in the transmission band, uniform response in the terminating resistance is obtained over the entire band.

DETAILED ANALYSIS OF MECHANICAL AND ELECTRICAL ANALOGS[1]

Before going on with a detailed treatment of the electrical analog of the mechanical structures used in the problem of phonographic reproduction, a list of the corresponding quantities used in the two systems will be given, together with the symbols employed.

Mechanical		Electrical	
Force	F (dyne)	Voltage	E (volt)
Velocity	v (cm/s)	Current	i (ampere)
Displacement	s (cm)	Charge	g (coulomb)
Impedance	z dyn · s/cm or mechanical ohm)	Impedance	Z (ohm)
Resistance	r (dyn · s/cm)	Resistance	R (ohm)
Reactance	x (dyn · s/cm)	Reactance	X (ohm)
Mass	m (gram)	Inductance	L (henry)
Compliance	c (cm/dyn) [6]	Capacity	C (farad)

In addition to the above certain other quantities, such as angular displacement, pressure, and impedance per unit area, and a few others which have no direct electrical analog, will be used. These quantities, however, are either standard in the literature or may always be reduced to those given above.

As illustrations of the general methods employed, certain important portions of the reproducer will be considered in detail. Considering first the electrical analog of the air chamber[2] between the diaphragm and horn, we make use of the following list of symbols (see Figs. 3, 4, 15, and 16):

- m_3 effective mass of diaphragm, grams
- A_1 equivalent area of diaphragm, cm^2
- c_6 compliance of edge of diaphragm
- A_2 area of throat of horn
- z_h impedance of horn: vector ratio of applied force at throat of horn to resultant linear velocity of air
- s_1 displacement of diaphragm
- v_1 velocity of diaphragm
- s_2 displacement of air in throat of horn
- v_2 velocity of air in throat of horn
- P_0, V_0 initial pressure and volume of air chamber
- F force applied to diaphragm
- p small change of pressure in air chamber.

For a small change p in the pressure within the air-chamber we have

$$p = \frac{n(A_1 s_1 - A_2 s_2)P_0}{V_0} \tag{1}$$

where $n = 1$ for an isothermal change and 1.4 for an adiabatic change. For the case under consideration $n = 1.4$ very nearly.

If the horn opening is closed, $S_2 = 0$, and we get for the compliance of the air chamber as measured from the diaphragm

$$c_7 = \frac{S_1}{pA_1} = \frac{V_0}{np_0 A_1{}^2}.$$

We have the two force equations

$$m_3 \frac{\mathrm{d}v_1}{\mathrm{d}t} + \frac{s_1}{c_6} + pA_1 = F \tag{2}$$

$$z_h v_2 - pA_2 = 0 \tag{3}$$

or substituting the values of p and c_7,

$$m_3 \frac{\mathrm{d}v_1}{\mathrm{d}t} + \frac{s_1}{c_6} + \frac{1}{c_7}\left[s_1 - \left(\frac{A_2}{A_1}\right)s_2\right] = F \tag{4}$$

$$z_h v_2 + \frac{1}{c_7}\left[\left(\frac{A_2}{A_1}\right)^2 s_2 - \left(\frac{A_2}{A_1}\right)s_2\right] = 0. \tag{5}$$

Considering now the analogous electrical circuit, and assuming the velocity, current, force, and voltage to vary sinusoidally, we have the parallel relationship for the steady-state conditions as follows.

1) Mechanical case (Fig. 3):

$$m_3 \frac{\mathrm{d}v_1}{\mathrm{d}t} + \frac{s_1}{c_6} + \frac{1}{c_7}\left[s_1 - \left(\frac{A_2}{A_1}\right)s_2\right] = F$$

$$z_h v_2 + \frac{1}{c_7}\left[\left(\frac{A_2}{A_1}\right)^2 s_2 - \left(\frac{A_2}{A_1}\right)s_1\right] = 0.$$

If $v_1 = j\omega s_1$, etc.,

$$z_1 v_1 - z_m v_2 = F$$
$$z_2 v_2 - z_m v_1 = 0$$

[1] The authors wish to express their appreciation to Mr. E. L. Norton for his courtesy in working out the mathematics of the mechanical and electrical analogs which are shown in this paper.

[2] The use of the air chamber to increase the loading effect of the horn on the diaphragm has been appreciated for a number of years. It has been used in telephone receivers, phonographs, and loud-speaking receivers since their earliest developments. A treatment of the force equations of the air-chamber was given in [7]. The equivalent structure, however, was analyzed as a compliance and resistance in series instead of in shunt.

where

$$z_1 = \mathrm{j}\left(\omega m_3 - \frac{1}{\omega c_6} - \frac{1}{\omega c_7}\right)$$

$$z_2 = \left[z_\mathrm{h} - \mathrm{j}\left(\frac{A_2}{A_1}\right)^2 \frac{1}{\omega c_7}\right]$$

$$z_m = -\mathrm{j}\left(\frac{A_2}{A_1}\right)\frac{1}{\omega c_7}.$$

2) Electrical case with ideal transformer of turns ratio N_2/N_1 (Fig. 4):

$$L_3 \frac{\mathrm{d}i_1}{\mathrm{d}t} + \frac{q_1}{C_6} + \frac{1}{C_7}\left[q_1 - \left(\frac{N_2}{N_1}\right) q_2\right] = E$$

$$Z_\mathrm{h} i_2 + \frac{1}{C_7}\left[\left(\frac{N_2}{N_1}\right)^2 q_2 - \left(\frac{N_2}{N_1}\right) q_1\right] = 0.$$

If $i_1 = \mathrm{j}\omega q_1$, etc.,

$$Z_1 i_1 - Z_m i_2 = E$$
$$Z_2 i_2 - Z_m i_1 = 0$$

where

$$Z_1 = \mathrm{j}\left(\omega L_3 - \frac{1}{\omega C_6} - \frac{1}{\omega C_7}\right)$$

$$Z_2 = \left[z_\mathrm{h} - \mathrm{j}\left(\frac{N_2}{N_1}\right)^2 \frac{1}{\omega C_7}\right]$$

$$Z_m = -\mathrm{j}\left(\frac{N_2}{N_1}\right)\frac{1}{\omega C_7}.$$

The last five equations in each case give the complete solution of the network. By analogy between the two sets of equations, therefore, the air chamber corresponding to the shunt capacity C_7 is spoken of as a shunt compliance, $c_7 = V_0/nP_0A_1^2$, together with a transformer inserted before the horn, which has an equivalent turns ratio equal to the ratio of the areas of the diaphragm and horn openings.

Taking up now the somewhat different illustration of the needle arm, the following symbols are needed (see Figs. 5, 6, 15, and 16):

- l_1 distance from pivot point to end of needle
- l_2 distance from pivot point to center of "spider" (Fig. 15)
- I moment of inertia of needle arm
- m_1 apparent or equivalent mass of arm as measured from center of spider, $= I/l_2^2$
- c_1 compliance of needle point
- c_2 compliance of bearing to turning of the needle arm, as measured from end of arm at spider
- c_3 compliance of end of needle arm attached to spider
- s_1 displacement of tip of needle
- s_2 displacement of end of arm at spider
- s_3 displacement of spider
- z_s mechanical impedance of spider and remainder of structure: vector ratio of applied force to resultant velocity
- θ angular displacement of needle arm
- F applied force at needle point.

We have the three force equations:

$$\frac{s_1 - l_1\theta}{c_1} = F \tag{6}$$

$$I\frac{\mathrm{d}^2\theta}{\mathrm{d}t^2} + \frac{(l_1\theta - s_1)l_1}{c_1} + \frac{\theta l_2^2}{c_2} + \frac{(l_2\theta - s_3)l_2}{c_3} = 0 \tag{7}$$

$$\frac{s_3 - l_2\theta}{c_3} + z_\mathrm{s}\frac{\mathrm{d}s_3}{\mathrm{d}t} = 0. \tag{8}$$

Replacing θ by s_2/l_2 and I by $m_1 l_2^2$ gives

$$\frac{s_1 - (l_1/l_2)s_2}{c_1} = F \tag{9}$$

$$m_1\frac{\mathrm{d}^2 s_2}{\mathrm{d}t^2} + s_2\left[\left(\frac{l_1}{l_2}\right)^2\frac{1}{c_1} + \frac{1}{c_2} + \frac{1}{c_3}\right] - \frac{l_1}{l_2}\frac{s_1}{c_1} - \frac{s_3}{c_3} = 0 \tag{10}$$

$$\frac{s_3 - s_2}{c_3} + z_\mathrm{s}\frac{\mathrm{d}s_3}{\mathrm{d}t} = 0. \tag{11}$$

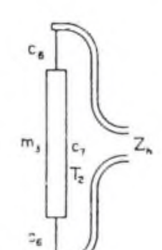


Fig. 3. Schematic mechanical arrangement of diaphragm and air chamber.

Fig. 5. Schematic mechanical arrangement of needle-arm transformer.

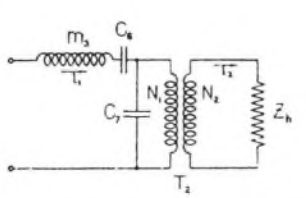


Fig. 4. Electrical equivalent of mechanical system shown in Fig. 3.

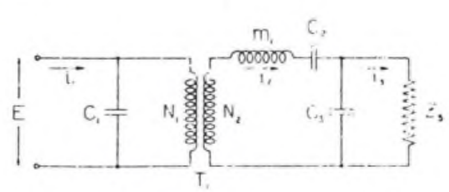


Fig. 6. Electrical equivalent of system shown in Fig. 5 with its termination.

Considering now the analogous electrical circuit, assuming as before sine functions for v, i, F, and E, we have the following equation.

1) Mechanical case substituting $v_1 = j\omega s_1$, etc., in the last equations:

$$-j\frac{v_1}{\omega c_1} + j\frac{l_1}{l_2}\frac{v_2}{\omega c_1} = F$$

$$jv_2\left[\omega m_1 - \left(\frac{l_1}{l_2}\right)^2\frac{1}{\omega c_1} - \frac{1}{\omega c_2} - \frac{1}{\omega c_3}\right]$$

$$+ j\frac{l_1}{l_2}\frac{v_1}{\omega c_1} + j\frac{v_3}{\omega c_3} = 0$$

$$j\frac{v_2}{\omega c_3} + v_3\left(z_s - j\frac{1}{\omega c_3}\right) = 0.$$

2) Electrical case with ideal transformer of turns ratio N_2/N_1:

$$-j\frac{i_1}{\omega C_1} + j\left(\frac{N_2}{N_1}\right)\frac{i_2}{\omega C_1} = E$$

$$ji_2\left[\omega L_1 - \left(\frac{N_2}{N_1}\right)^2\frac{1}{\omega C_1} - \frac{1}{\omega C_2} - \frac{1}{\omega C_3}\right]$$

$$+ j\left(\frac{N_2}{N_1}\right)\frac{i_1}{\omega C_1} + j\frac{i_3}{\omega C_3} = 0$$

$$j\frac{i_2}{\omega C_3} + i_3\left(Z_s - j\frac{1}{\omega C_3}\right) = 0.$$

The analogy between the two sets of equations is quite obvious. It will be noticed that the effect of the lever arm is to introduce an equivalent transformer of a turn ratio which is the reciprocal of the corresponding lengths of the arms.

The general method of deducing the equivalent electric circuits should be clear from the above illustrations of the air chamber and of the needle arm. For example, in the spider section (Fig. 15) the mass is driven directly by the force from the needle-arm compliance, there being a small series compliance in the connection owing to bending of connecting rod. The diaphragm is connected through the compliance of the prongs of the spider. The equivalent circuits are shown in Figs. 7 and 16.

The equations of this network may be obtained from the equations for the needle arm by placing c_1 equal to zero, taking a unity ratio transformer, and substituting m_2 for m_1, c_4 for c_2, c_5 for c_3, and z_d for z_s.

Another type of network which occurs frequently in the building of mechanical vibrating systems is represented diagrammatically in Fig. 8. This is clearly a particular case of Fig. 7 with c_4 made infinite.

By combining Fig. 6 representing the needle arm, Fig. 7 representing the spider section, and Fig. 4 representing the diaphragm, air chamber and horn, the complete reproducer may be built up. The resultant network is shown in Fig. 16. Since methods are available in the theory of electric wave filters to determine the proper values of the elements of the complete network for a free transfer of energy throughout an assigned frequency band, the analogous mechanical elements may be determined in the same manner.

GENERAL DESIGN OF MECHANICAL SYSTEMS

In designing mechanical systems of the band pass type, the problem is threefold: first, that of arranging the masses and compliances such that they form repeated filter sections; second, determining the magnitude of these quantities so that with or without transformers the separate sections all have the same cut-off frequencies[3] and characteristic impedances; third, to provide the proper resistance termination. Where the transmitted mechanical power has not been radiated as sound, this third part has been one of the most difficult to fulfill.

In designing these systems, practical difficulties arose: first, the difficulty of insuring that the parts vibrated in the desired degrees of freedom only, and second, the difficulty of determining the magnitudes of the various effective masses, compliances, and resistances. Before the work to be described could be carried out practically it became necessary to develop a method of measuring mechanical impedances [3]. Such a method has been developed which at the present time covers a range of frequencies from somewhere below 50 to about 4500 pulses per second. Work is still being continued to extend the method to higher frequencies. This method of measurement has been very useful not only in determining the magnitudes of the impedances in the degrees of freedom in which it is desired that they shall operate, but in determining the impedances to motion of the various parts in directions in which they should not be permitted to vibrate. In connection with the measurement of the magnitudes of the parts in the desired degrees of freedom this method enables us to determine the constants of the mechanical networks under their conditions of

[3] It is of course permissible to have a section having a higher cutoff than the others provided its characteristic impedance is the same as that of the others over the transmission band of those having the lower cutoff.

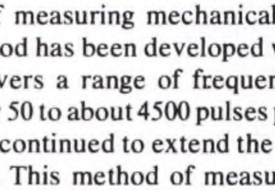


Fig. 7. Electrical equivalent of spider section.

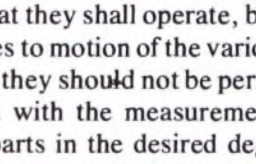


Fig. 8. Electrical equivalent of simple low-pass type of network which occurs frequently in this work.

operation. Experience so far has indicated that when all the degrees of freedom have been taken into account and when the dynamic axes of vibration have been properly chosen, the static and dynamic constants of the parts are the same, and it is then possible to check the parts by simple static measurements. In the early attempts to build these systems very large discrepancies between the static and dynamic characteristics were found.

THE RECORDER

One of the early practical phonographic applications of electric filter design to mechanical problems was the development of an electromagnetic recorder. The instrument as finally constructed is essentially a properly terminated three-section mechanical filter in which the recording stylus and its holder constitute the series mass in the second section. Since a filter of this type appears at its input end as approximately a pure resistance within the transmission band, the current in the series inductances, that is, in the mechanical case, the velocity of the series masses is proportional to the driving force.

Figs. 10–12 show, respectively, a complete recorder, a drawing of the mechanical filter of such a recorder, and a diagram of the equivalent electric circuit. The armature acts as the series mass m_1 in the first section, the magnetic field as the series negative compliance $-c_0$, the shaft between the armature and the stylus holder as the shunt compliance c_1, the balancing springs as the series compliance c_2, the stylus holder and the stylus as the series mass m_2, the shaft between the stylus holder and the disk, coupling the system to the terminating resistance, as the compliance c_3, the coupling disk as the series mass m_3, and the terminating line as approximately a mechanical resistance.

All of these equivalents are seen from the simple analogs previously outlined, with the exception of the terminating resistance and the negative compliance $-c_0$. The terminating resistance was originally made up of a series of filter sections of lumped series masses and shunt compliances with a small amount of damping added to the motion of each of the series masses. Fig. 13 shows one of the early recorders equipped with this type of resistance termination. The reason for using such a complicated termination lies in the fact that most of the known mechanical resistances have values which are functions of frequency or of amplitude or both. Also in most cases, the mechanical resistance is accompanied by either a mass or a compliance reactance. By using a multisection filter which is sufficiently long so that a wave entering it will be essentially absorbed before it has reached the far end, been reflected, and returned to the entering end, it has been possible to use imperfect types of damping for this line and still obtain, over the desired band, an essentially pure resistance at the input end.

More recently a continuous line has been developed which is much easier of practical attainment than the complicated lump-loaded filter. The recorder shown in Fig. 10 is so equipped. Fig. 14 shows calibration curves of three types of recorders.

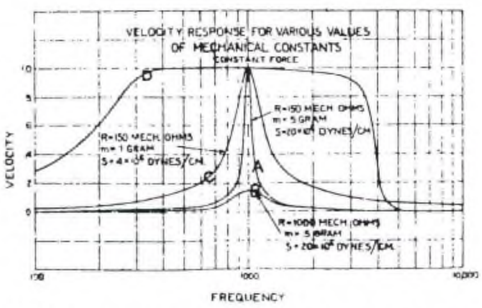


Fig. 9. Velocity response for various values of mechanical constants.

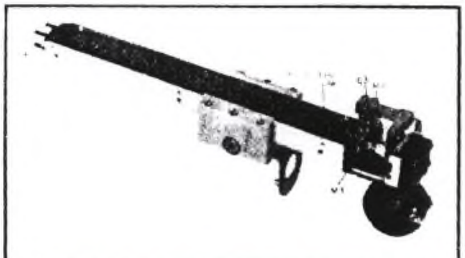


Fig. 10. Electromagnetic recorder, complete except for the bottom of the case.

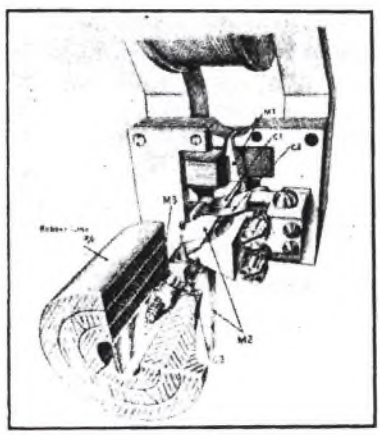


Fig. 11. Detail of mechanical filter of an electromagnetic recorder. (Lettering same as in Fig. 12.)

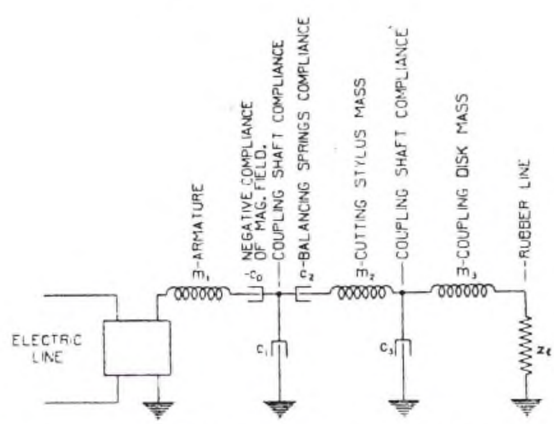


Fig. 12. Equivalent electric circuit of the electromagnetic recorder.

The compliance $-c_0$ is a mechanical quantity for which there is no simple electric analog. In a balanced armature type of structure such as that shown in Fig. 11, the action of the field on the armature, when it is at its center point, is balanced. If, however, the armature is deflected a small distance from this equilibrium, there is exerted by the magnetic field a torque tending to pull the armature further away from its center position. The value of this torque for small amplitudes is proportional to the angular displacement. It is seen therefore that this quantity is of the nature of a compliance but that the back force is in a reverse direction to that required for a positive compliance.

DESIGN OF THE REPRODUCING APPARATUS

As the analogy between the mechanical and electrical filters is more perfectly shown in the case of the reproducing equipment, the detailed quantitative description will be given in this connection. Figs. 15 and 16 show, respectively, a diagram of the reproducing system and its equivalent electric circuit. From these diagrams it is evident which units in the mechanical system correspond to the various electrical parts. As the series compliances c_2, c_4, and c_6 have been made so large that the low-frequency cutoff caused by them lies well below the low-frequency cutoff of the horn, an inappreciable error is introduced in using for design purposes formulas of low-pass filters [9]. The two formulas which will be used are as follows:

$$f_c = \frac{1}{\pi}\sqrt{\frac{1}{mc}} \tag{12}$$

where

- f_c cut-off frequency of a lumped transmission system, hertz
- c shunt compliance per section, cm/dyn
- m series mass per section, gram

$$z_0 \quad \sqrt{m/c} \tag{13}$$

where z_0 is the value of characteristic impedance over the greater part of the band range.[4]

Eqs. (12) and (13) which form the basis of the design work contain four variables, f_c, c, m, and z_0. It is, therefore, necessary to determine two of them by the physical requirements of the problem after which the other two are determined. The upper cut-off frequency f_c was arbitrarily chosen at 5000 pulses per second as a compromise between the highest frequency occurring on the record and the increase in surface noise as the cutoff is raised. The choice of the other arbitrarily set variable came after considerable preliminary experimenting and was fixed by the difficulty of obtaining a diaphragm which is light enough and has a large enough area. Hence the effective mass of the diaphragm m_3 (Figs. 15 and 16) was fixed at 0.186 gram, a value that can be obtained by careful design. The effective area can be made as large as 13 cm². For convenience let the arbitrary value chosen for $f_c = \bar{f}_c$ and the value of $m = \bar{m}_3$.

Solving Eqs. (12) and (13) for c and z_0, we get

$$c = \frac{1}{\pi^2 \bar{f}_c^{\,2} \bar{m}_3} \tag{14}$$

$$z_0 = \pi \bar{f}_c \bar{m}_3 \tag{15}$$

also

$$z_0 = \frac{1}{\pi c \bar{f}_c}. \tag{16}$$

In order to obtain the low value of mass mentioned,

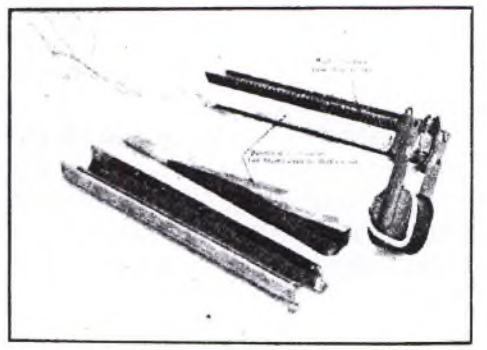


Fig. 13. Electromagnetic recorder using lumped loaded termination. The method of furnishing dissipation to the lumped loaded line is shown.

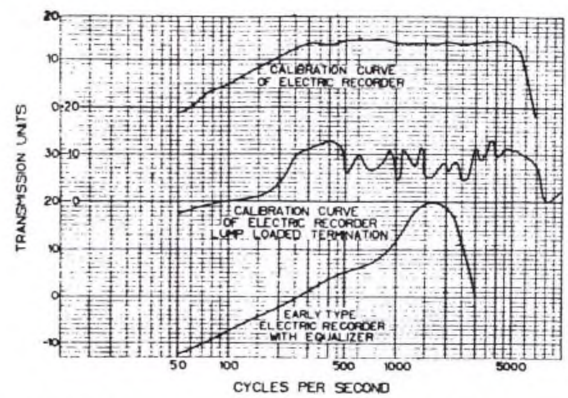


Fig. 14. Calibration curve of three types of electromagnetic recorders.

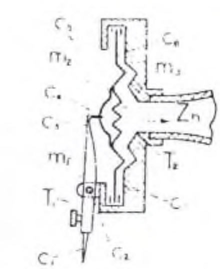


Fig. 15. Diagrammatic sketch of mechanical system of phonograph.

[4] z_0 may be called nominal mid-shunt or mid-series impedance. Their actual values in the transmission band being, at any frequency f_c,

mid-series $= z_0\sqrt{1-(f/f_c)^2}$

mid-shunt $\dfrac{z_0}{\sqrt{1-(f/f_c)^2}}$.

with a large enough area, it was necessary to make the diaphragm of a very stiff light material. An aluminum alloy sheet 0.0017 in. thick was chosen and concentrically corrugated as shown in Figs. 17 and 18. These corrugations are spaces sufficiently close so that the natural periods of the flat surfaces are all above $\bar{f}_c$. To ensure that this central stiffened portion should vibrate with approximate plunger action, which is more efficient than diaphragm action, it is driven at six points near its periphery.

Reference to Figs. 15 and 16 and Eq. (14) shows that the compliance of the air chamber c_7, of the spider legs c_5, and shunt tip of the needle arm c_3 are determined. Also the mass of the spider m_2 and the effective mass of the needle arm m_1, as viewed at the point where it is attached to the spider, are determined.

The impedance looking into the system from the record is determined by the rate at which it is necessary to radiate energy in order that the reproduction may be loud enough. The power taken from the record

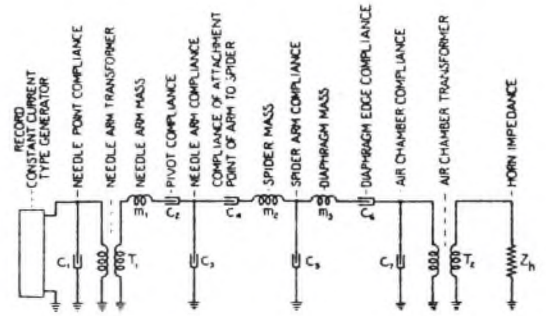


Fig. 16. Electric equivalent of system shown in Fig. 15.

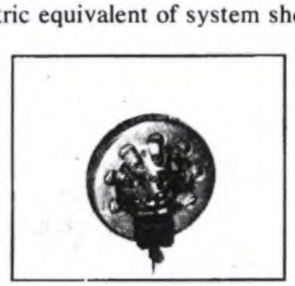


Fig. 17. Mechanical reproducing system without horn.

Fig. 18. Sectional drawing showing construction of system shown in Fig. 17.

is approximately $\nu^2 z_0$, since z_0 is a resistance over most of the band. Experiment has shown this value of z_0 to be approximately 4500 mechanical ohms.

But substituting in Eq. (13) the value of $\bar{m}_3$, and from Eq. (14) the value of c_5, we find that the impedance is only 2920 mechanical ohms. It is, therefore, necessary to use a transformer whose impedance ratio is 4500/2920. From this and a knowledge of filter structures the needle-point compliance can be determined. The value obtained is easily realized with commercial types of needles.

It will be noted that the record is shown in Fig. 16 as a constant-current generator, that is, a generator whose impedance appears high as viewed from the needle point. That this is necessary is obvious when it is remembered that, if the impedance looking back into the record were to equal the impedance of the filter system, the walls of the record would have to yield an amount comparable with one half the amplitude of the lateral cut. This would cause a breakdown of the record material with consequent damage.

The design of the system is complete, therefore, except for the resistance termination which is supplied by the horn for all frequencies above its low-frequency cutoff. The characteristics of the horn will be dealt with later. The resistance within the band looking in at the small end of the horn is GA_2, where G is the mechanical ohms per square centimeter of an infinite cylindrical tube of the same area, and A_2 is the area in square centimeters of the small end of the horn.

Let A_1 be the effective plunger area of the diaphragm (as previously mentioned, this is 13 cm²). The impedance looking back at the diaphragm is

$$z_0 = \pi \bar{f}_c \bar{m}_3 = 2920 \text{ mechanical ohms}$$

from Eq. (15), and the impedance looking at a horn whose small end area equals A_2 is

$$z_h = r_0 = A_2 G. \tag{17}$$

Substituting $A_2 = 13$ cm² and $G = 41\ \Omega/\text{cm}^2$, we get

$$z_h = r_0 = 533 \text{ mechanical ohms.}$$

This is entirely insufficient so that the air-chamber transformer becomes necessary.

To calculate the necessary ratio of areas on the two sides of the air-chamber transformer, the following formula is needed. The formula assumes the chamber to be relatively small compared with all wave lengths of the sound to be transmitted, that is, the pressure changes throughout the chamber are substantially in phase:

$$\frac{z_0}{z_h} = \frac{A_1^2}{A_2^2} \tag{18}$$

where

- z_0 impedance of primary side of transformer, mechanical ohm
- z_h impedance on secondary side of transformer, that is, horn impedance, mechanical ohm

v_1 mechanical current, that is, velocity on primary side of transformer, cm/s

v_2 mechanical current on secondary side of transformer, cm/s

F_1 alternating force on primary side of air-chamber transformer, dyne

F_2 alternating force on secondary side of air-chamber transformer, dyne

A_1 effective area working into primary side of air chamber, cm^2

A_2 effective area working into secondary side of air chamber, cm^2.

The characteristic impedance of the line on the diaphragm or primary side of the air chamber as shown by Eq. (15) is

$$z_0 = \pi \bar{f}_c \bar{m}_3. \tag{19}$$

From Eq. (17) the characteristic impedance on the horn or secondary side is

$$z_h = GA_2. \tag{20}$$

Therefore,

$$\left(\frac{A_2}{A_1}\right)^2 = \frac{z_h}{z_0} = \frac{GA_2}{\pi \bar{f}_c \bar{m}_3}. \tag{21}$$

Solving this for A_2, we get

$$A_2 = \frac{GA_1^2}{\pi \bar{f}_c \bar{m}_3}. \tag{22}$$

The equivalent of the air chamber to a transformer shunted by a compliance is shown earlier in the paper.

In applying the foregoing method of design to a practical structure, a number of design problems had to be solved. The construction of the diaphragm and the method by which it is actuated have already been described, except for the tangential corrugations constituting the series compliance. The use of these corrugations results in the value of the series compliance being practically independent of the nature of the clamping, and has eliminated a tendency to "rattle" introduced by unevenness in the clamping surfaces.

Another feature in connection with the sound box is the needle-arm bearing shown in Figs. 17 and 18. Ordinary knife-edge bearings are not sufficiently rigid as fulcrums, and the rotational reactance as well as the rotational resistance is undesirably large. A construction which has been found to meet the necessary requirements is the ball-bearing type with the steel balls held in position by magnetic pull. By making the ball-containing case of soft steel and magnetizing the shaft, it has been possible to manufacture this bearing reliably and cheaply.

The horn which has been used as a terminating resistance to the mechanical filter structure is a logarithmic one. The general properties of logarithmic horns have been understood for some time [10].

There are two fundamental constants of such a horn, the first is the area of the large end and the second the rate of taper. The area of the mouth determines the lowest frequency which is radiated satisfactorily. The energy of the frequencies below this is largely reflected if it is permitted to reach the mouth.

From the equations given by Webster [10] it can be shown that all logarithmic horns have a low-frequency cutoff which is determined by the rate of taper. If the rate of taper is so proportioned that its resulting cutoff prevents the lower frequencies from reaching the horn mouth, the horn will then radiate all frequencies reaching its mouth, and very little reflection will result.[5] It is possible, therefore, to build a horn having no marked fundamental resonance.

Since the characteristics of the horn are determined by the area of its mouth and by its rate of taper, the length of the horn is determined by the area of the small end. This area is determined in turn by the mechanical impedance and effective area of the system which it is terminating, as shown in Eq. (22). It is seen, therefore, that the length of the horn should not be considered as a fundamental constant. A paper describing the design of horns based on these principles is being prepared.

An interesting feature of the horn which has been built commercially is its method of folding. The sketch in Fig. 19 shows a shadow picture of the horn. It will be noticed that the sound passage is folded only in its thin direction, which permits the radius of the turns to be small and thereby makes the folding compact.

Fig. 20 shows the frequency characteristic of a phonograph designed as shown above with a logarithmic horn whose rate of taper and area of mouth opening place the low cutoff at about 115 Hz. It also shows the characteristics of one of the best of the old-

[5] The authors wish to express their appreciation in this connection of the work of Mr. P. B. Flanders who carried out the mathematical investigation of these relationships and to Mr. A. L. Thuras who checked experimentally the mathematical theory.

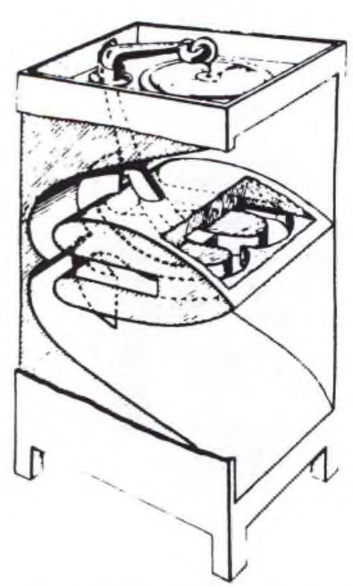


Fig. 19. Sectional view of folded horn showing air passage.

style phonographs. Curve A represents the new machine, while curve B represents the old-style standard machine.

DISCUSSION

C. R. Hanna: The following discussion applies particularly to that part of the paper dealing with the reproducing mechanism. The relative mertis of the several improvements that were made are not clearly brought out in the paper, and it is the purpose of the writer to compare the importance of the various developments.

In listening to reproduction from one of the new-type phonographs, the average person is impressed with just two things; first, the apparent greater volume of sound, and second, the great improvement in the response at low frequencies. The greater volume of sound is due partly to the fact that there are more low frequencies present, and perhaps, in a measure, to the fact that the diaphragm is one which acts like a piston, causing a greater volumetric rate of displacement of air into the horn for a given needle velocity than with the old type of flat diaphragm.

The improvement in the low-frequency characteristic of the reproducer, as described, could not have been obtained without the use of the slowly expanding logarithmic or exponential horn. The authors refer to the work of A. G. Webster in this connection. The general properties of the exponential horn were given in his National Academy of Science paper of 1919. Webster did not, however, carry his work sufficiently far to show the properties which the authors have stated in their paper, namely, that the exponential horn is a uniform radiator of sound down to a certain frequency, known as the cut-off frequency, which is determined by the rate of increase of section and the area of the large end of the horn.

The authors cite some work (as yet unpublished), by Messrs. Flanders and Thuras, in which these properties are shown both theoretically and experimentally. I desire to call attention to the fact that the paper by Hanna and Slepian on "The Function and Design of Horns for Loud Speakers" [7] showed these same properties for the exponential horn. The equation for such a horn is

$$A = A_0 \epsilon^{Bx}$$

where

A area at any point
A_0 initial area
x distance from initial area, cm
B constant which determines the rate of increase.

It was demonstrated that the cut-off frequency is determined by the relation

$$\frac{2\pi f}{B} = \frac{a}{2}$$

where a is the velocity of sound. From this it is seen that the smaller B is, the lower will be the cutoff.

The radiation characteristic of the infinite exponential horn for a fixed velocity of air in its throat was also shown in the paper by Hanna and Slepian. Fig. 21 shows this curve. The abscissas are ω/B, and the ordinates give the comparison between the exponential horn and the infinite straight pipe which is a uniform radiator down to zero frequency. The cut-off point is seen to be as stated above.

It was clearly brought out in this paper that an exponential horn could be made with much smaller dimensions than any other shape of horn giving equal performance. A comparison was also shown between a particular exponential horn and a conical horn of equal length and terminal dimensions. This is given in Fig. 22, the superiority of the exponential horn being quite pronounced. Up to this time many persons had advocated the conical horn. It is believed that this paper was the first to show the superiority of the exponential horn.

Now, taking up the matter of the final or large area of the horn, as is pointed out by Maxfield and Harrison, if this area is large enough to prevent end reflec-

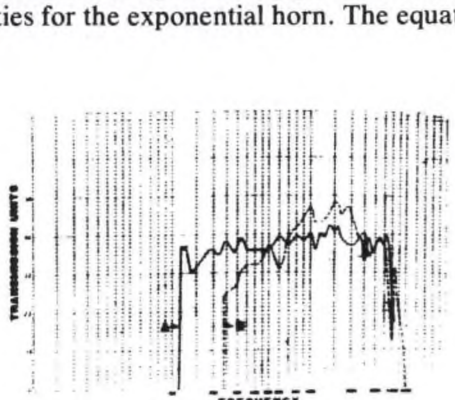


Fig. 20. Response frequency characteristic of two phonographs. A—characteristic of band-pass filter type described. B—characteristic of one of the best commercial machines previously on the market.

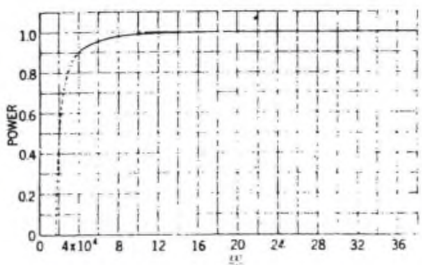


Fig. 21

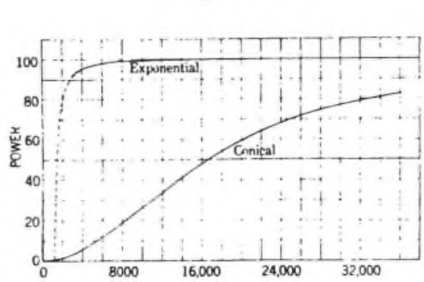


Fig. 22

tions in the range of frequencies where the horn is a good radiator along its length, a horn will be secured which has very little resonance. The curves of Fig. 23 were presented by Hanna and Slepian to show the variation of reflection with frequency and area. The curves indicate that the smaller the area and the lower the frequency, the greater will be the reflection. It is seen, however, from the curve for the largest area, that the reflection becomes appreciable only in the range of frequencies where the horn ceases to be a good radiator along its length. Hence it follows that a horn of this shape can be designed with no marked fundamental resonance.

The degree of horn resonance and the position of the cut-off frequency as indicated by Fig. 20 of the paper agree very closely with values that can be predicted from the curves of Figs. 21 and 23 in this discussion.

The very careful proportioning of masses and compliances in the mechanical system of the reproducer has played only a minor part in the securing of a more uniform frequency-response characteristic than in the older types of phonographs. The slight extension of the upper frequency range may be attributed to the accurate design of mechanical parts. It should be pointed out, however, that since the phonograph record is a constant-current (or velocity) generator, the impedance of the mechanical system does not have to be uniform over a wide band of frequencies for it to be forced to vibrate in accordance with the vibrational velocity of the record. A departure from this fact, not apparent from the electrical analogy given by the authors in their paper, is the ability of the whole arm of the reproducer to vibrate in the low-frequency ranges instead of just the diaphragm mechanisms. This may be overcome either by increasing the mass of the arm or, as the authors have done, by reducing the stiffness of the diaphragm.

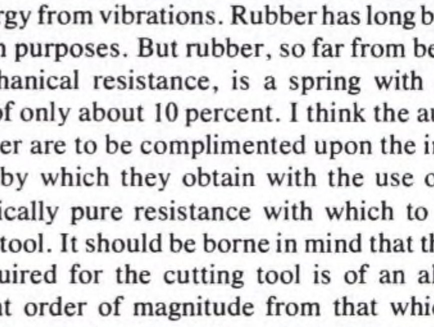
Great credit is due the authors for the design of a mechanical system which is light and resilient, enabling the needle to track the record with small reaction force (and consequent decrease in wear) at the high frequencies where the accelerations are great, and at low frequencies where the deflections are great. The big improvement in the quality of reproduction, however, is due to the use of an exponential horn whose rate of increase of section is very small and whose final section is quite large.

E. W. Kellogg: I think most of us have thought of the rocking arm, which connects the needle with the diaphragm in a phonograph, as a simple lever, rigid enough so that when the needle moves one way, the diaphragm moves the opposite direction by a corresponding amount. If we could see what is really going on during a high-frequency vibration, we should probably find that the motion was more nearly like that of a snake. Messrs. Maxfield and Harrison and their associates have accepted the wave-motion picture and based their design upon it. The most striking resultant change in design is the interposition of a flexible link, or spring, between the end of the lever and the diaphragm. On first thought, it seems like deliberately throwing away some of the available motion, but the result is quite the opposite. I refer to the spider through which the diaphragm is driven.

If telephonic currents are to be transmitted without distortion over a high-efficiency line of length exceeding a sixth of a wavelength for the highest frequencies, the line must end in a noninductive resistance of a definite value. A corresponding resistance is required in a mechanical system. In the case of the reproducing system the required resistance is obtained from the sound radiation of the diaphragm. But for the cutting tool, some other resistance must be found. In an electrical circuit nothing is easier to get than resistance, yet its mechanical counterpart is by no means easy to obtain. Sliding friction is not at all suitable. Motion in viscous fluids and electromagnetic drag, such as used in wattmeters, are true analogs. I wish to draw an illustration from the case of electromagnetic drag. An aluminum ring, weighing about 4 grams, surrounds a magnet pole, so that it is in a radial field of about 10 000 gauss. If one pushes it up and down, it feels as if it were in thick molasses. Under its own weight it settles about 1 millimeter per second. Yet if this ring is vibrated in an axial direction at 4000 cycles, its mass so predominates over the resistance that the power factor is only about 40 percent. Mechanical hysteresis is another means for absorbing energy from vibrations. Rubber has long been used for such purposes. But rubber, so far from being pure in mechanical resistance, is a spring with a power factor of only about 10 percent. I think the authors of the paper are to be complimented upon the ingenious device by which they obtain with the use of rubber a practically pure resistance with which to load the cutting tool. It should be borne in mind that the damping required for the cutting tool is of an altogether different order of magnitude from that which many of us have employed to take out the resonance peaks from loudspeaker diaphragms and similar applications.

The paper mentions methods of measuring mechanical impedances. I should be much interested to hear something further of the means used, for the problem presents many difficulties, and the results of such measurements would find many applications.

One statement in the paper causes considerable sur-



prise. The knife edge was discarded because it has too great an elastic yield, and because it brings in too much rotational friction. The knife edge, of course, must work with an initial pressure exceeding the maximum force on the bearing, due to the vibrations, and is not well adapted to stand forces in more than one direction, but, in the case of the pivot for the reproducing lever, one would expect a well designed knife edge to work very satisfactorily.

L. T. Robinson: I am in agreement with the statement of the authors that "There is therefore no distortion in the record whose purpose is to compensate for errors in the reproducing equipment." In employing so many elements, some of which can be so readily modified in performance the temptation is very strong to look only at the final result and not be too critical as to where any corrective treatment is to be administered. I hope the stand taken by the authors will be firmly adhered by them and others who are working along similar lines. In this way, any progress that has been made, or will be made, becomes permanent.

Speaking of the electrically cut record in general, we need not, for the moment, be concerned with minor details of the process. The results already obtained are so good that we may feel sure that the electrically cut record has come to stay and will place the phonographic art on an entirely new plane of excellence.

The mechanical reproducing system described by the authors is a distinct advance over former phonographs. However, I feel that full realization of the advantage of the electrically cut record will come through electrical reproduction.

One great advantage of the electrical method of reproducing is that the control of the sound volume is obtainable quite independent of the cut on the record, and the cut on the record is controllable with consideration for the best conditions for the record alone. The advantages of such separation can be learned from the paper if it is read with this point in mind.

Volume of the sound reproduced is quite important, and reproduction to be quite satisfactory must be about equal to the volume of the original sounds. A loud tone produced on a given musical instrument is quite different from a soft tone produced on the same instrument and reproduced with larger volume.

A. E. Kennelly: We have here presented to us the wonderful analogy which underlies mechanical and electrical phenomena, with mechanical phenomena interpreted in electrical terms.

We have long known that mechanical inertia was really electrical, and now we are finding that all these mechanical phenomena are primarily electrical quantities.

J. P. Maxfield: There are one or two technical questions brought out in Mr. Hanna's discussion which are of interest. The first deals with the statement that the new reproducing mechanism has a greater apparent volume of sound. In this connection, it is interesting to note that the response curves of the new and the old machines shown in Fig. 20 indicate that in the frequency region from around 800 to 2000 Hz, the old machine produced a louder sound for a given needle velocity. It will be seen, therefore, that this apparent increase in volume has been obtained by the widening of the band reproduced rather than by increasing the amount of energy radiated in that frequency band in which the old machine was most efficient.

The other point of interest refers to his statement that "The very careful proportioning of the masses and compliances in the mechanical system of the reproducer has played only a minor part in the securing of the more uniform frequency response characteristic than in the older type of phonograph." In view of the high quality which is obtained and of the commercial requirement that the wear on the record shall not be excessive, the authors do not agree with this statement. It is not necessarily true that because the record is a constant-current generator and, therefore, delivers constant current to the sound-box mechanism, the diaphragm necessarily delivers constant current to the air. If a relatively stiff, heavy, vibrating system is used, it becomes exceedingly difficult with the constant-current type of generator to obtain good quality, and if it is possible to obtain it, the wear on the record becomes excessive. A reference to Fig. 16 indicates that the first part of the system reached is the needle point which has a definite compliance. At the higher frequencies, if the impedance of the reproducing system is too high, the needle will bend instead of moving the rest of the system, and the response will be reduced thereby. Similarly, at the low-frequency end, if the diaphragm-edge compliance is too small, that is, if the diaphragm is too stiff, the whole tone arm will vibrate and thereby reduce the motion of the diaphragm relative to its ease. It is true that, so far as response is concerned, this effect can be corrected by increasing the moment of inertia of the tone arm—a method which is equivalent to increasing the mutual inductance of the transformer T_1, (Fig. 16); but if the solution is thus obtained, the wear on the record becomes excessive, and in some cases the force becomes so great that the needle will not track in the groove.

The solution presented in the paper is one in which the mechanical impedance has been made as nearly as possible independent of frequency and is of the nature of a pure mechanical resistance. The result of this type of solution is that a maximum of sound energy at all frequencies within the band is radiated with a minimum of wear on the record.

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cutterheads and lathes

A Moving-Coil Feedback Disk Recorder*

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Westrex Corporation, Hollywood, California

The distinctive feature of this recorder is the application of corrective feedback originating in the actual stylus-driving mechanism. Thus, the motion of the stylus is accurately controlled, irrespective of recording conditions, over a wide range of amplitudes and frequencies. The latest improvements are described and include a simple method of applying heat to the stylus and the use of tapered shank styli to facilitate their replacement.

RECENT improvements in disk recording have fully maintained the prestige of this important medium. No obstacle appears to prevent it from keeping abreast of the highest standards of sound recording in other media. The advent of such techniques as the microgroove, the hot stylus, improved record materials, and recorders of superior design all have contributed to the improvement of the disk record.

In 1938 a disk recorder for vertical-cut grooves was described[1] which incorporated principles completely different from those previously employed in any type of recording device. The performance of this recorder was based on the use of electromechanical feedback, and it was the first commercial application of this novel method of controlling the behavior of an electrically driven mechanical device.

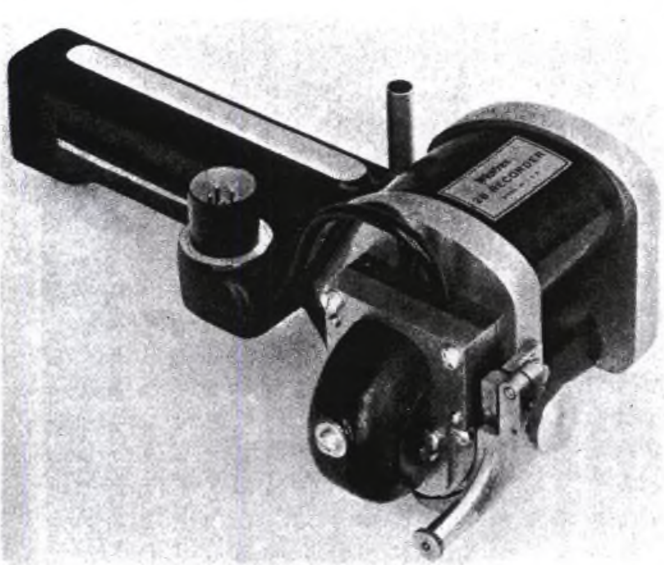

FIG. 1. Top view of recorder.

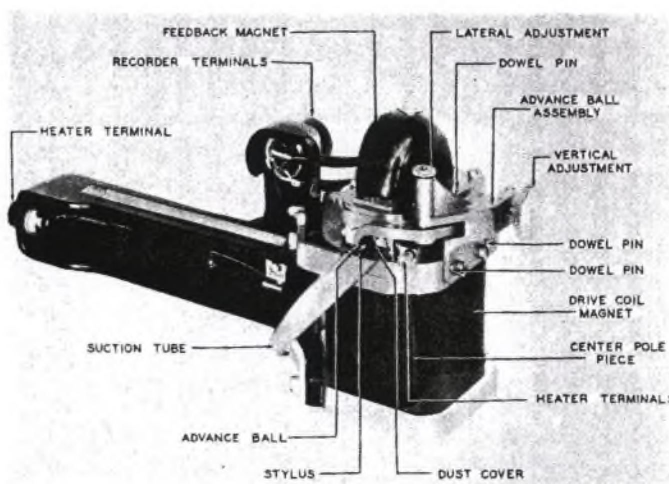

FIG. 2. Bottom view of recorder.

By 1938, inverse feedback[2] had come to be recognized as an almost ideal method of correcting distortion in electronic equipment. But its application to combined electronic and mechanical equipment was new. The successful performance of the vertical recorder led to the development of a lateral disk recorder,[3] and the latest model of this recorder will be described here. This has been designated as the Westrex 2B recorder and is shown in Figs. 1 and 2.

GENERAL CONSIDERATIONS

Feedback may be defined as a process of coupling the output of a system to its input by means which correct self-

* Presented at the Sixth Annual Convention of the Audio Engineering Society, New York, October 14-16, 1954.

[1] L. Vieth and C. F. Wiebusch, "Recent Development in Hill and Dale Recorders," *J. Soc. Motion Picture Engrs.*, **30**, 96 (January, 1938).

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generated internal distortions in proportion to the amount of feedback employed. Now, if we include an electrically driven mechanical device in the system, the final output may be made practically identical with the input if the feedback voltage originates in the final mechanical element. The same rules apply with respect to feedback gain and phase relationships, and the same benefits may be expected. In general, design problems are magnified by the presence of mechanical elements because of their tendency to "break up" into multiple modes of vibration. If the amplitude and phase relationships of the feedback voltage are of such a character as to increase the input signal, the system may become oscillatory and "sing." As a practical consideration, the entire system must be designed to include a frequency spectrum far beyond its working range. These difficulties and the general expressions for feedback are fully discussed in the references cited above.

In order to apply feedback control to a mechanical device, means must therefore be provided to generate the feedback voltage at the final output of the mechanical system. Furthermore the device should be free to operate over a greater working distance than would be required in normal use. Preferably, it should require no delicate adjustments. These are among the reasons why a dual moving-coil type of stylus-driving element forms the basis of this recorder. The inherent stability and freedom from distortion of the moving coil have long been recognized, and it has found wide application in other devices. Edison experimented with a moving-coil type of telephone prior to his invention of the phonograph.[4]

FEEDBACK AND DAMPING

The vibrating coil may be controlled over a wide range of frequencies by applying sufficient feedback voltage. This is the equivalent of a large amount of viscous damping. The concept of feedback as the equivalent of damping may appear new to electronic engineers; however, it is equivalent to ideal linear damping, but with the important difference that it merely *governs* power instead of *absorbing* it in large amounts in the control process.

The amount of power required by a moving-coil drive assembly, sufficiently damped by mechanical means to meet the requirements of disk recording, has previously discouraged its use for this purpose. With the advent of feedback and modern magnetic materials, the way was cleared for the utilization of the advantages of feedback without the dissipation of excessive power in mechanical damping. Furthermore, the physical embodiment of true viscous damping offers difficulty, and its effectiveness may be altered by temperature changes which result, for example, from the use of heated styli.

[4] Francis Jehl, *Menlo Park Reminiscences,* Vol. 1, p. 140, Edison Institute, Dearborn, Mich.

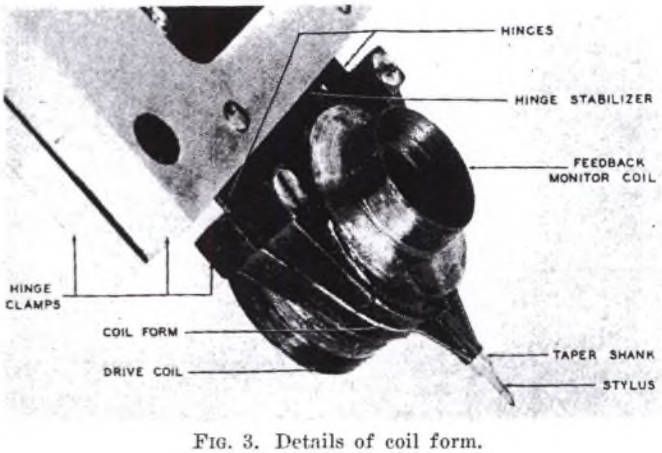

FIG. 3. Details of coil form.

Since disk recording is primarily a constant-velocity type of recording wherein amplitude is inversely proportional to frequency, it is desirable to design a recorder for constant-velocity operation over the desired audio spectrum. With this done, electrical pre-equalization may be added in a straightforward manner to conform to any desired recording characteristic. A vibrating device may be made to maintain constant velocity over a wide frequency range by resonating it at the geometrically mean frequency and providing it with sufficient damping or its equivalent.

THE MOVING COIL

The moving element of the 2B recorder consists of a drive coil connected to the amplifier output and a smaller feedback or monitor coil connected to the feedback terminals of the amplifier. The two coils are supported on a magnesium coil form which is attached in turn to the recorder proper by two beryllium copper cantilever spring hinges. These are shown in Fig. 3. The design of the coil form is intended to minimize mass and at the same time provide sufficient rigidity to avoid vibration in spurious modes. The voltage generated by the monitor coil is directly proportional to the velocity of the drive assembly throughout the usual range of frequencies. When this voltage is amplified and fed back to the driver amplifier in proper phase relationship, the driver unit must conform to it and establish a constant-velocity recording characteristic. Each coil is provided with a separate magnetic assembly to minimize coupling through the flux sources of the coils. The pole tips of the magnetic assemblies are supplied with copper slugs and rings to reduce coupling further. The center pole piece of the driver unit is manufactured from Permendeur to obtain maximum magnetic flux density. The vibrating assembly is therefore a unit whose motion is indicated by the output of the monitor coil. This not only originates the feedback control but also provides facilities for monitoring the motion of the stylus during recording. A portion of its output may be fed into a monitor channel, thus permitting the operator to observe accurately what is being recorded.

POWER CONSIDERATIONS

It may be of interest to compare the amount of power required to drive the coil at a standard recording velocity with feedback *vs* the power required with an equivalent amount of viscous damping. The mass of the driver unit, the compliance of the hinge mountings, and their mechanical resistance are analogous to a series resonant LCR circuit, since equal velocity, analogous to current, exists in all three elements. The stylus revolves about the hinge; hence the current requirement is shown by

$$i = \dot{\phi}\frac{10Z_r}{Blr}$$

where i = current in drive coil in amperes.
$\dot{\phi}$ = angular velocity of the stylus in rad/sec.
Z_r = rotational impedance in dyne–cm–sec/rad.
B = flux density in gauss.
l = length of wire on drive coil in centimeters.
r = working radius of coil in centimeters.

It can be shown that a damping factor of 8 results in a constant-velocity characteristic over a range of about 8 octaves. The value of Z_r which fulfills this condition may be determined from the resonant frequency of the coil combined with a displacement measurement of the stylus. The latter shows the rotational compliance, in rad/dyne-cm, to be given by

$$C_r = 3 \times 10^{-8}$$

and, since the resonant frequency in cycles is

$$f_r = 850$$

the moment of inertia in g-cm^2 must be

$$I = 1.2$$

Hence at resonance the mechanical impedance is found to be

$$Z_r = 1 \times 10^5 \text{ dyne–cm–sec/rad}$$

The value of *Blr* in this recorder is

$$Blr = 1.6 \times 10^5$$

Thus, for a recording velocity of 8 cm/sec, where $\dot{\phi}$ is equal to 3 rad/sec in this recorder, a current of 1.86 amperes is required and the power required by the 5.8-ohm drive coil is 20 watts. This is more power than the small coil can tolerate. Furthermore, the power will rise at higher frequencies, owing to the large amount of pre-equalization in present use at these frequencies.

In the case of feedback, the power reaches a minimum at the resonant frequency, 0.015 watt being required for an 8-cm/sec recording velocity. The increase in power above and below resonance is proportionately greater in this case than that discussed above for viscous damping, but the power remains much lower throughout the significant band of frequencies. However, in spite of this performance, an amplifier with a relatively high output-power rating is required to take care of surges resulting from the high levels which may occur at high frequencies with the pre-equalizers currently in use.

RECENT DEVELOPMENTS

The 2B recorder has an improved spring hinge arrangement. Previously there had been a tendency toward a shift in the operating axis of the cantilever spring at frequencies above 10 kc, causing sharp dips in the recorded frequency response in this range. The recent interest in recording higher frequencies has resulted in a modification which has largely eliminated these "holes." The modification consists in extending the body of the coil form to the normal rotational axis of the hinge. This fin-shaped extension revolves about an axis which coincides with the axis of the spring one-third the distance from the spring clamp to the coil form. It is embedded at this point in a compliant material which allows the hinge to flex normally but discourages rotation about any but the intended axis.

Further improvement in high-frequency response is obtained by the addition of a hollow disk of magnesium, cemented inside the form near the drive coil to increase its rigidity. A copper drive coil has been substituted for the original edge-wound aluminum coil. This was adopted because of mechanical difficulties in securing a reliable bond between the preformed aluminum coil and form. The new coils are more efficient on an ampere-turn basis and are capable of withstanding relatively high temperatures. Extreme velocities over long periods of time fail to damage either coils or hinges.

THE HOT STYLUS

Stylus heating facilities have been provided in the 2B recorder. These consist of two small terminals to which may be attached a simple heater coil energized with 6 volts from an ac source. The heater coils are designed to slide over the stylus and are held in place by the natural spring tension of their leads. This arrangement permits them to be installed or removed readily for stylus cleaning or replacement. The feedback control prevents appreciable distortion originating either from the physical presence of the heater or hum pickup from the ac heater current. The advantages of hot-stylus recording hardly need discussion today. Experiments were made with stylus heating as early as 1891 and, odd as it may seem now, the pioneer inventor who conducted them recommended a gas flame as a source of heat.[5] The ability

[5] W. Bruening, U. S. Patent 486,608 (Nov. 22, 1892).

to cut clean grooves with sharp-edged styli affords great improvement in frequency response and signal-to-noise ratios as the groove velocity decreases.

Acetate playbacks of LP recordings at 10 kc and at a 6-in. diameter show as much as 10 db improvement in noise and a gain of 16 db in signal when cut with heated sharp styli as compared with unheated, dull-edged styli[6] intended for cold use. The noise increases 16 db when the heat is removed from the sharp stylus, which indicates reductions of 16 and 6 db, respectively, for hot and cold styli. But the improvement in frequency response is even greater and is independent of heat adjustment, which illustrates the effectiveness of hot-stylus recording. No operational changes appear necessary except that the suction system must not be turned off during recording, as the resulting increase in temperature may burn chips on the stylus and these are difficult to remove. The temperature at the stylus tip is in the neighborhood of 350° F and can be measured with waxes having known melting points. With the hot-stylus attachment the coil form is modified to use tapered-shank styli, and a removal tool has been designed for their ready installation and removal. The desirable holding power of the tapered shank is as advantageous here as in machine tools and, incidentally, the same degree of Brown and Sharpe taper is used.

ADVANCE BALL ASSEMBLY

The advance ball assembly has been redesigned in the 2B recorder. This permits positioning the advance ball directly ahead of the recording area, and it is provided with an additional lateral adjustment for tracking exactly in line with the material to be removed in order to prevent scars from remaining on the record if a particle of dirt collects under the advance ball. Because of this relocation, it was necessary to locate a new suction tube alongside the stylus rather than in front of it. The use of advance balls to control depth of grooves has necessarily become more general because of the low cutting resistance offered by heated microstyli. They offer further advantages in preventing stylus damage and in permitting standing starts.

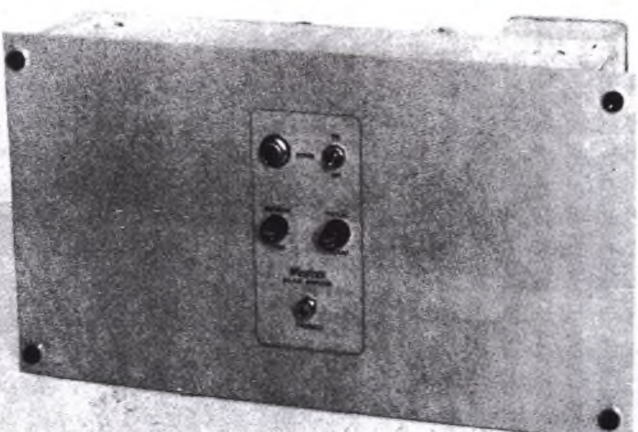

FIG. 4. Front view of amplifier.

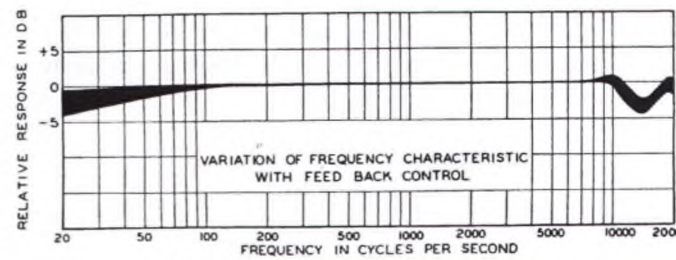

FIG. 5. Typical frequency characteristic.

THE DRIVING AMPLIFIER

A new amplifier, coded Westrex RA-1541-A, has been designed to operate with the 2B recorder. It is shown in Fig. 4. This amplifier conforms to the latest trends in regard to components, packaging, and accessibility. The signal gain has been increased 6 db and the frequency response improved. The factor of safety against "singing" without preselection of tubes has been increased. Power-frequency hum levels, measured at the monitor terminals, are at least 69 db below an 8-cm/sec recording level. The gain of the over-all feedback circuit is 57 db with maximum feedback control setting. The gain of the signal circuit is 41 db with the external feedback disconnected. An internal amplifier feedback loop originates in the secondary of the output transformer, providing a minimum of distortion at all levels below overload.

FREQUENCY CHARACTERISTIC

A typical frequency characteristic is shown in Fig. 5. The high and low frequencies may be increased or decreased several decibels by adjustment of the feedback control. When adjusted to normal the system is flat from amplifier input to reproducer output, from 30 cps to 11 kc. A dip of about 4 db centers at 15 kc, beyond which a long rise occurs, extending to approximately 28 kc, where the level exceeds that of the midrange. The feedback has little control beyond 11 kc, and the peak at 28 kc appears to be due solely to a secondary mechanical resonance. This rise in response serves a useful purpose, however, because the system may be equalized to a point well beyond 20 kc by the insertion of a single equalizer to boost the level in the 15-kc range. Recordings at 78 rpm, with subsequent reproduction at 33⅓ rpm to reduce the effects of reproducer losses, were used to establish these data, and the results agree in general with optical calibrations. With this method, experimental recordings have been made, without any equalization, which reproduced flat within ± 3 db from 20 cps to 36 kc. The presence

[6] Isabel L. Capps, "Recording Styli," *Electronic Ind.*, p. 65 (November, 1946).

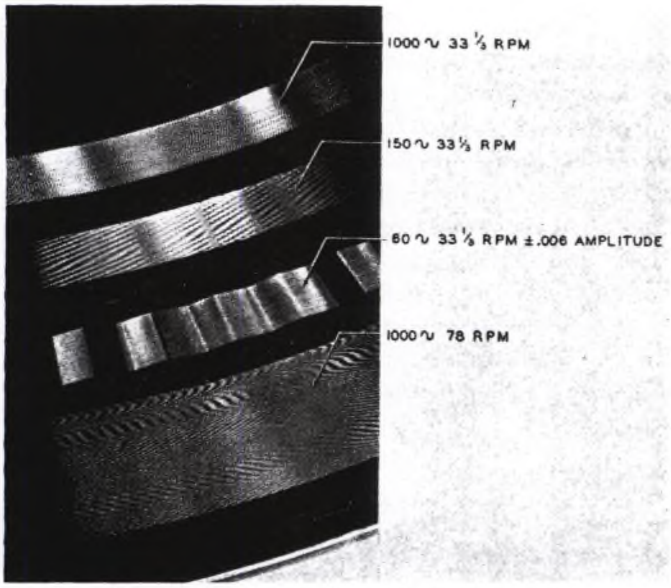

FIG. 6. Grooves recorded with square waves.

of heater wires and dust covers produces minor effects in the extremely high-frequency range. The long slender styli used with heat appear to produce no changes whatever. Furthermore, the mechanical impedance of the recorder is so high in this range, in spite of the loss of feedback, that variations in recording materials produce little effect.

SQUARE-WAVE TESTS

When square waves are recorded on a disk with a constant-velocity characteristic, they present the appearance of triangular or "zigzag" patterns. An example of square-wave recording is shown in Fig. 6. This shape results because the usual expression for square waves is integrated to the triangular form by the recording process. Conversely, the reproducing process differentiates them back to a square wave in reproduction. This latter effect may be visualized by the fact that the triangular grooves impart constant motion to the stylus with sudden changes in direction. Since constant stylus velocity generates constant reproducer voltage, the motion imparted to the reproducer generates alternate fixed voltages, hence the resulting square waves.

The recordings illustrated in Fig. 6 were made with the 2B recorder and RA-1541-A amplifier combination. The 60-cps section was made on a synchronous recording machine and at a very high amplitude to better illustrate the interesting triangular wave form. Incidentally, this track cannot be reproduced by standard reproducers, as they find difficulty tracking triangular waves of much less amplitude. Possibly these waves may serve in testing the relative tracking abilities of reproducers.

Square-wave testing is becoming widely used in providing useful information, particularly in regard to devices which may overshoot or "ring" when subjected to a sudden input voltage. This applies to mechanical as well as to electrical circuits. Square waves indicate the generation of distortion during transient operation, such as with program material, which may not be indicated by other means. The results of square input voltages into the disk system are shown in Fig. 7 under the conditions indicated therein. Reproduction from the record was made with a typical variable-reluctance cartridge. For the purposes of these graphs a small amount of low-frequency boost was added to offset the generally preferred slightly drooping low-end characteristic used in recording. Without this change, the existing phase shift at these frequencies caused a slight amount of tilt in the square waves. The differences between the three reproducing conditions at 1 kc may be due to dynamic changes in the reproducer as well as to changes in high-frequency response. The 150-cps monitor view shows a slight excess of the fundamental frequency, and its reproduction, in turn, shows a slight phase shift in the fundamental. It is apparent that the over-all result is good in all cases and that the recorder and amplifier combination itself is capable of meeting very strict square-wave requirements.

AMPLITUDE DISTORTION

As previously indicated,[3] the inherent distortion of the recorder is so low, even at high amplitudes, that care must be taken in selecting a reproducer capable of measuring grooves recorded at such amplitudes. However, the following meas-

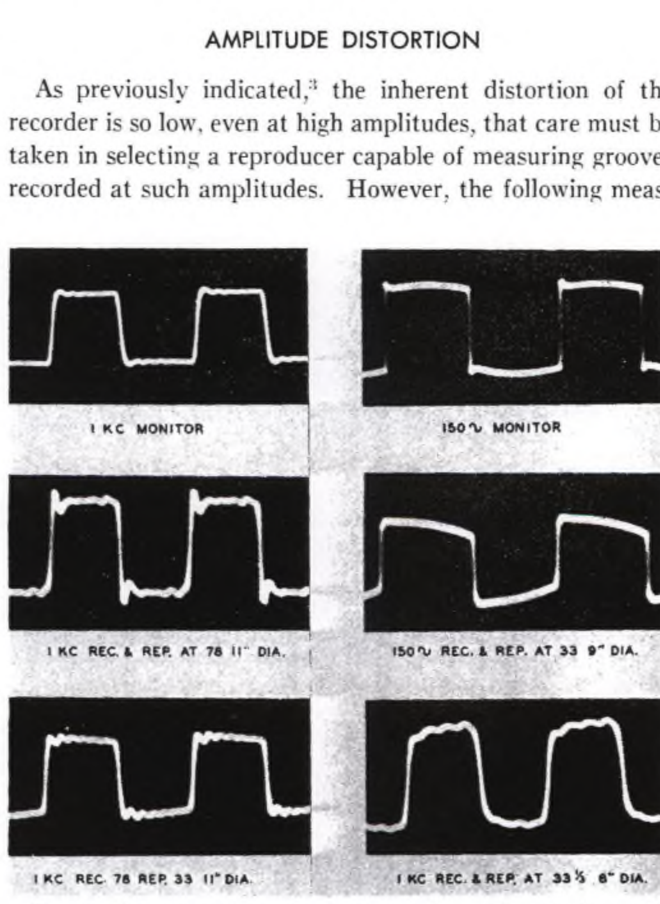

FIG. 7. Square-wave photos from monitor and reproducer signals.

Intermodulation Measurements

LF	HF	LF amplitude	Monitor output	Reproducer output
150	2 kc	± 0.00025	0.4%	0.5%
150	2 kc	± 0.0005	0.4	1.1
150	2 kc	± 0.001	0.6	1.5
150	7 kc	± 0.00025	0.3	1.6
150	7 kc	± 0.0005	0.4	2.0
150	7 kc	± 0.001	0.5	2.6
150	12 kc	± 0.0005	0.3	3.6

urements indicate very satisfactory over-all results using a standard variable-reluctance cartridge at amplitudes encountered in microgroove recordings. These measurements were made on standard recording blanks with typical pre- and post-equalizers in the circuit to conform to commercial practice. The intermodulation appears to be independent of recording diameter.

CONCLUSION

Original recordings of full orchestral music and dialogue sequences, using the latest disk techniques, have impressed critical listeners with their dynamic range, purity of tone, and extraordinary freedom from intermodulation during the most complicated passages. The quality of these recordings has created a new respect for disk recording among those engaged largely in other methods of recording. The lateral disk recorder described in this paper, in addition to retaining the operating characteristics of the earlier 2A model, incorporates new and improved design features permitting high-fidelity reproduction from 30 cps to at least 15 kc.

ACKNOWLEDGMENTS

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Recent Developments in Precision Master Recording Lathes*

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Work on the Hydrofeed® master recording lathe actually began almost 20 years ago. The increased standards of performance required to produce the microgroove LP record are easily met by this new development. Basically this lathe differs from previous ones in the use of a hydraulic-feed system, for the transverse feed, rather than a lead screw. Many other improvements are described, such as automatic pitch control, reduction of wow, flutter, and rumble to very low levels, and automatic operation throughout the operating cycle.

INTRODUCTION

IT SEEMS wise to review briefly the basic method for duplicating phonograph records, so that we may all speak the same language, for the duration of this paper, at least.

In the early days of record manufacture it was customary to cut, or engrave, a sharply defined groove in a rotating wax blank directly from the mechanical vibrations of a diaphragm. This wax engraving was called the "master recording."

For many years wax was widely used until the development of lacquer-coated discs. At the present time, the lacquer-coated discs are used almost universally for making the master recording.

After the master is cut, it must be coated with a suitable conducting material; then it is plated with 0.010 to 0.020 in. of copper, nickel, or some other suitable metal. The plated metal, of course, has a surface which is the negative image of the original master. Quite obviously, the plated metal cannot offer any improvement in surface smoothness over the original master recording. This plated metal is called a "stamper." It is mounted on a suitable backing platen and, with the aid of heat and pressure, is used to produce positive copies in a suitable plastic material.

Again the surface smoothness of the resulting finished record, called a "pressing," will not be any better than the finish of the stamper from which it was made.

The original 78-rpm shellac phonograph records contained some abrasive material which actually ground the playback stylus or "needle" to the exact shape of the groove.

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† President.

It was customary to change these needles after the playing of each record, for best results. Typical stylus forces were around 2½ oz, or 70 grams. Quite a contrast with modern practice.

The groove dimensions were determined by a stylus having a tip radius of approximately 2½ mils. The grooves were played back using a tip radius of about 3 mils during the more recent years of "quasi-standardization." Since these 78-rpm records were cut at around 100 lines per inch, simple analysis indicates that signals having a maximum peak groove amplitude of 2 or 3 mils were all that could be placed on the record without the danger of cutting into the adjacent groove.

It was customary to think of 40 db as a reasonable dynamic range for the early 78-rpm records. Since 40 db represents a 100:1 amplitude ratio, this means that the surface of the finished record groove had to be smooth compared with 1/100 of 2 mils, or 20 micro-in., or 20/1,000,000 in., whichever way you may choose to say it. Actually, as better pressing materials with lower abrasive content and the lighter-weight electrical playback systems came into universal use, the residual noise was pushed down towards 50 db below signal level, which value corresponded to about 7 micro-in. of equivalent surface finish in the groove.

Then came the microgroove development. The minimum microgroove requirements, to be competitive with 78-rpm practice, are as follows:

First, the maximum peak groove amplitude is reduced slightly to about 1 mil, with some occasional peaks exceeding this limit. Since the playback tip radius is "standardized" at 1 mil, the minimum groove width is approximately 2 3/10 mils. If we wish to retain the 50-db residual, the groove smoothness must now be less than 3½ micro-in.

In the meantime, FM transmission practice adopted in television, also, has accustomed the public to noise levels which are, in general, 60 db down from the peak levels. If we now add this on to our microgroove requirement, the *maximum* tolerable surface roughness in the recorded groove becomes approximately 1 micro-in. Good engineering practice requires, of course, that the noise contribution of the master recording lathe be considerably lower than even this figure of 1 micro-in.; after all, there are other sources of trouble in the record-manufacturing process

For a time the early microgroove records suffered from a variety of troubles; then "hi-fi" came of age and there were *more* troubles. For example, the loudspeakers in common use today are capable of transmitting appreciable output at 30 cps and even lower frequencies. The value of this increase in low-frequency transmission has long been debated pro and con. Nevertheless, many prospective purchasers of microgroove records utilize such loudspeakers in their systems and do not intend to use "rumble filters," either! In addition to the loudspeakers, better pickups, amplifiers, and microphones have all extended the lower limit downward.

The demand for longer playing time in smaller space has already created a new speed, 16⅔ rpm. Until recently, this speed was used chiefly for "talking-book" records. Recent developments announced by the Chrysler Corporation offer a new "hi-fi" 16⅔ rpm record player as optional equipment in their 1956 line of automobiles.

In order to obtain fidelity at 16⅔ rpm comparable to that which is possible at the higher speeds it will be necessary to reduce the playback stylus to ½ mil. Fortunately, most good microgroove records can be played with such a stylus, since the masters were cut with a stylus having a 2/10-mil tip radius. Recent developments in the pressing of records permit better "fill" at the bottoms of the grooves and make more practical the use of a ½-mil playback stylus.

It is probable that the maximum recorded amplitude may be reduced to perhaps ½ mil for sake of longer playing time at 16⅔ rpm. If this is to be done, then either we must accept less than 60 db of dynamic range or reduce the maximum tolerable groove finish to ½ micro-in.

Therefore, to sum up current practice and probable future trends, it is now necessary to provide less than 1 micro-in. of surface finish in the recorded groove. In the future, even this requirement will have to be exceeded.

CONVENTIONAL RECORDING MECHANISMS

Figure 1 shows the basic recording mechanism in common use today. It consists of a lead screw which is usually driven from the same shaft as the turntable through a system of gears, belts, or other means. Regardless of how the lead screw is driven, some type of nut having a suitable thread must engage this screw to drive the cutter across the surface of the master blank.

In the finest master recording lathes, it has been customary to machine, grind, and lap these lead screws with great care to produce the finest possible surface finish. This,

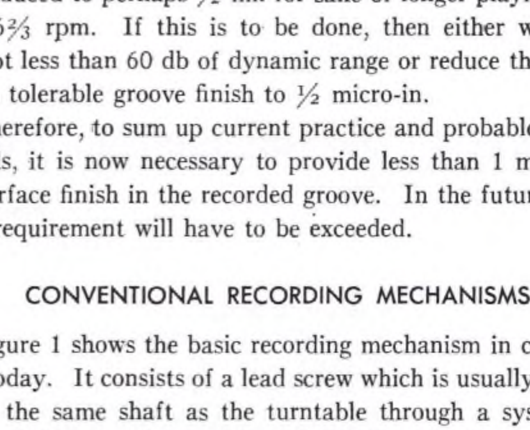

FIG. 1. The basic disc-recording mechanism in common use today.

of course, is quite necessary to minimize rumble. Occasionally, mechanical filters have been employed to attenuate further any residual rumble. The cutoff frequency of such filters has been set at the lowest value deemed practicable.

In other words, the lead screw is supposed to impart lateral motion with a dc component only—no ac component. The better the surface finish on the lead screw, the less chance there is of an ac component. In practice there is a limit to the fineness of the surface finish which can be put on a lead screw and maintained there with normal wear during recording.

It can be seen that any residual ac component of lateral motion resulting from lead-screw surface irregularities will be engraved or cut directly into the master and will be heard as rumble during playback. If these ac components are below the reproduced audible range in level, they will still be troublesome if they coincide with tone-arm resonance or overload the preamplifier and become modulated onto the desired signal.

BASIC PRINCIPLE OF THE HYDRAULIC-FEED SYSTEM

In an effort to overcome some of the inherent limitations presented by lead screws the hydraulic type of drive system shown in Fig. 2 was developed. The gauge shown in the diagram at the left of the tank is the pressure-indicating instrument. A static pressure of 150 lbs is maintained in the sealed tank. This pressure is communicated directly to one side of a piston which, in turn, is connected directly to the cutter. A special adjustable valve is provided between the other side of the piston and an open container, so that the opening of this valve will result in controlled leakage of the hydraulic fluid. As the fluid leaks past this special valve, the piston and its associated cutter will travel across the record blank. Uniform motion will result if the valve is properly constructed to assure laminar or turbulent free passage of the hydraulic fluid and if the carriage which supports the cutter encounters uniform friction. It is not too difficult to make this friction rather uniform, so that the resulting lateral motion is quite smooth and stable.

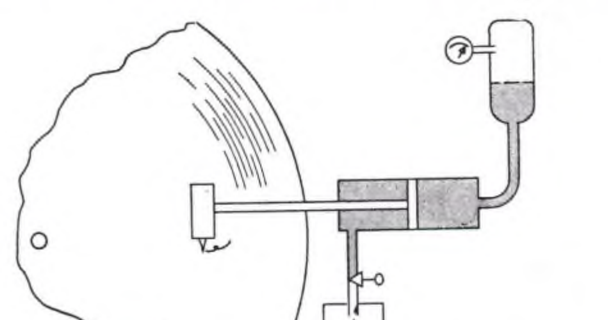

FIG. 2. Principle of the Hydrofeed hydraulic-feed recording mechanism.

Accurate machine tolerances on the valve parts have permitted use of the present recording lathe down to 600 or more lines per inch at 33⅓ rpm. The author is convinced that further refinements in the manufacture of these special hydraulic valves point to the practicability of cutting records at 1000 lines per inch in the near future.

Examination of the surface finish of recorded grooves made by means of the hydraulic-feed system reveals a substantial reduction in rumble—both audible and absolute. In fact, the author's company decided to enter the playback-turntable field in order to make available playback equipment capable of benefiting from the improved cutting made possible by this new recording lathe. With proper adjustment of the recording lathe and the playback turntable, dynamic ranges of over 70 db have been obtained.

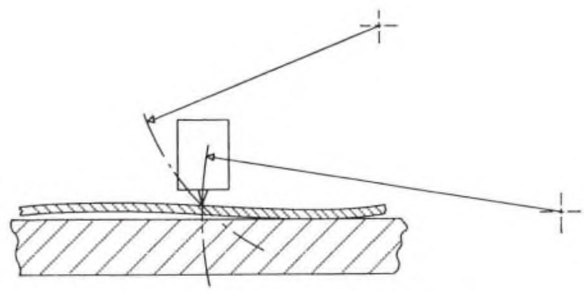

FIG. 3. Cutter pivot systems.

In order to make these measurements a direct-coupled oscilloscope was used, together with an FM type pickup having response to below 10 cps. Actually, the pickup had a tone-arm resonance at 7 cps which increased the apparent residual rumble appreciably. In order to facilitate these measurements, the surface-hiss noise was disregarded. Actually, 70 db represents about the limit of our measuring equipment. If we use this limit to estimate the equivalent surface roughness of the recorded groove, it appears that we have less than ⅓ micro-in. Of course, this measurement must of necessity include all the links in the chain. It is reasonable to assume, the author feels, that only a fraction of this very low residual figure is to be attributed to the recording lathe.

The matter of "wow" or once-around speed variation, was given quite a bit of long-overdue attention. Figure 3 illustrates two methods of arranging the pivots for vertical motion of the cutter. All recording lathes which the author has seen use a pivot system having an axis which is not in the same plane as the recording surface. Because of slightly warped blanks, this displaced axis gives rise to a component of cutter travel longitudinal with respect to the groove.

In our new master lathe—which we call the Hydrofeed®—

the pivots are located in the plane of the record surface. This arrangement gives rise to no "first order" longitudinal motion; hence, it contributes substantially no "wow."

The surface of the recording table has been lapped to a total runout of less than ½ mil. Vacuum hold-down of the master blank is provided through a series of special grooves in the turntable and a hollow turntable shaft.

In order to center the blank accurately, a special tapered center pin automatically centers the blank and then retracts slightly after the vacuum is "turned on," to prevent upward pressure from dimpling the surface. To match this accurate centering, our playback turntable has a special expanding collet spindle which automatically accommodates the normal range of center-hole variation.

MECHANICAL DETAILS OF THE SYSTEM

Figure 4 shows the cutter-bar assembly and reveals in detail how the placement of the vertical pivot axis is arranged. Several other features may be noted in the photograph:

1. The solenoid for electronic adjustment of depth of cut. This solenoid operates against a spring, so that power failure and similar occurrences cause the cutter to lift automatically, in order to prevent damage to the stylus.

2. The oil dashpot for damping the vertical motion of the cutter bar.

3. The felt advance "shoe" which supplies some additional vertical damping to help maintain uniform depth of cut.

4. The end of the suction hose for "chip" removal. (This is just visible at the extreme upper right-hand side of Fig. 4.)

5. The hydraulic cylinder. (This is visible between the lathe ways at the right of Fig. 4.)

Figure 5 gives an overall front view of the Hydrofeed lathe. The multiple-belt drive for the turntable is apparent. Speed change is effected by means of a stepped pulley and a multispeed motor. Electronic drive is optionally available, since almost all public power sources contain enough phase modulation to introduce random wow when used directly.

At the right-hand side of Fig. 5, are three indicating instruments. The one on the left reads 150 lbs of tank pressure on the hydraulic system. This pressure changes only very slightly during the operating cycle because the volume of the accumulator tank is large compared to that of the cylinder. After the recording cycle has been completed, a manually actuated hydraulic pump returns the fluid from the open container back up into the cylinder, forcing the piston, and consequently the cutter, back to the starting position.

The two other meters shown in Fig. 5 read the current in the stylus-heating coil and the current controlling the depth of cut, respectively.

In Fig. 6 the Hydrofeed lathe is seen diagonally from above.

Figure 7 shows the latest model of Hydrofeed lathe and

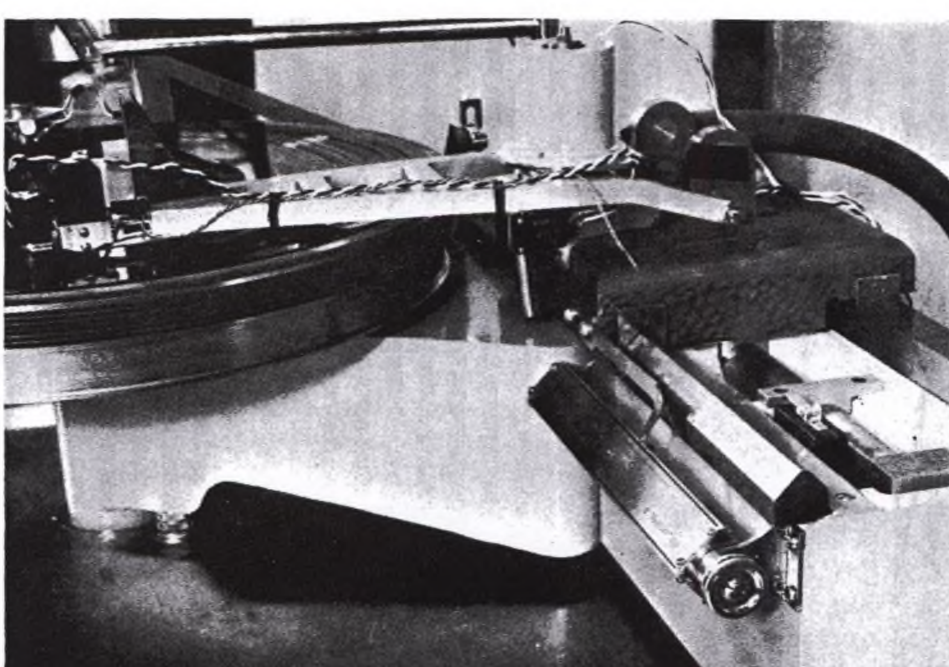

FIG. 4. Cutter-bar assembly of the Hydrofeed lathe.

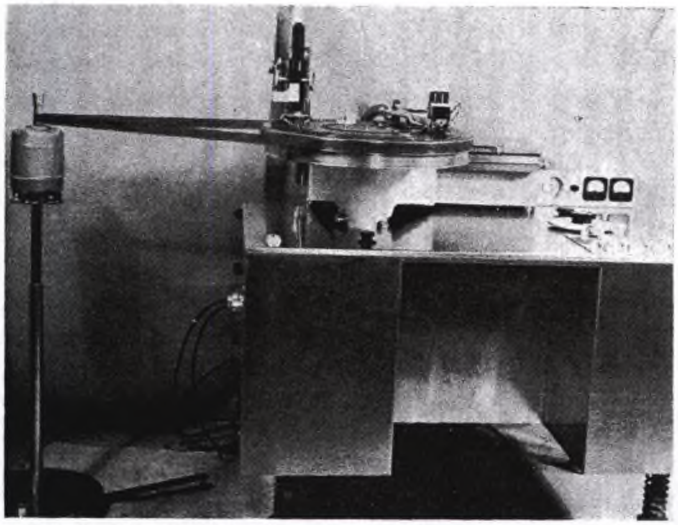

FIG. 5. Overall front view of the lathe, showing the multiple-belt drive for the turntable.

some associated electronic equipment for effecting automatic pitch variation as program level changes. Visible at the lower right of the figure are six knobs which are connected to special adjustable hydraulic valves. These hydraulic valves are all connected in parallel and are turned *on* and *off* electrically by means of six solenoid valves.

OPERATING ROUTINE

Operation is as follows: The button at the extreme upper right is pressed momentarily and the turntable begins to rotate, the vacuum system starts up, etc. After normal speed has been reached, the next button in the upper row is pressed momentarily to start normal feed. The next button in the row is then pressed to operate the fast-spiral valve so that the cutter is moved rapidly over to a point a short distance from the start of the record. Then the small knob visible just under the turntable is turned to increase the current in the depth-of-cut solenoid in order to start the cut; this knob also actuates a switch which turns on the heater current for the stylus. Next, the third button from the right—the one just under the fast-spiral button—is pushed to make the slow-spiral or lead-in groove. When the pointer on the scale of the lathe bed reads zero, the lead-in button is released, and the tape machine is started remotely by means of the last button at the left of the top row.

Since these buttons are arranged in the logical sequence of their normal use, it is very easy to operate them consecutively. If a symphony, say, is being recorded, whenever a

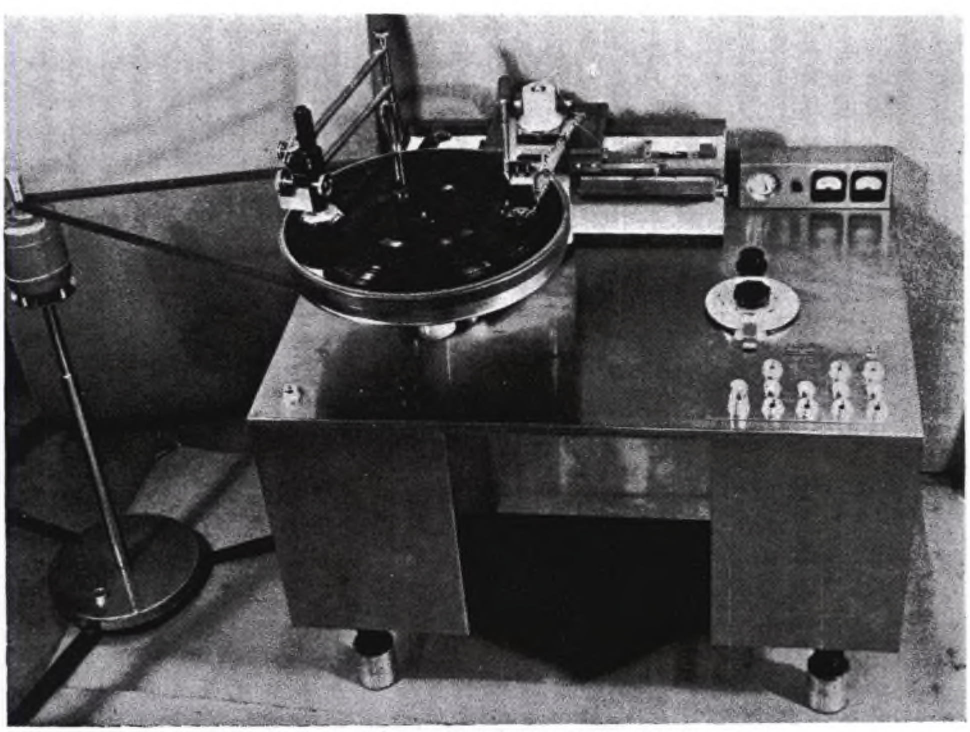

FIG. 6. The hydraulic-feed lathe seen diagonally from above.

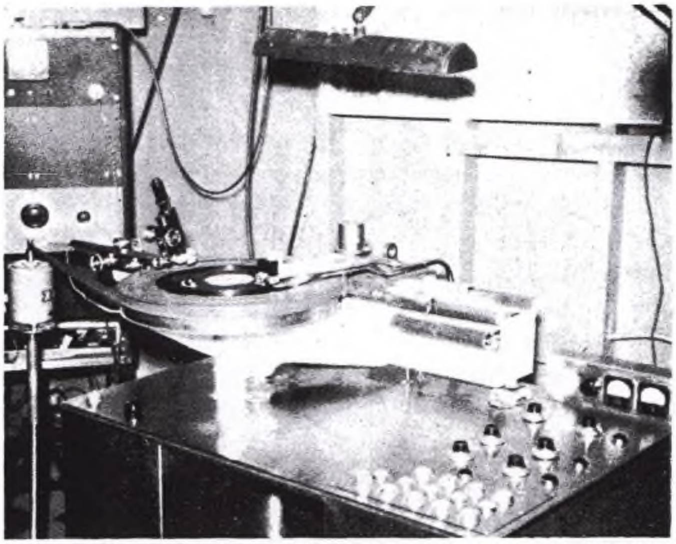

FIG. 7. The lathe with associated electronic equipment for automatic pitch control.

pause occurs between movements, a momentary pressure on the slow-spiral button will put a space on the record. A special connector is provided so that this can be done automatically, by means of an electrical contact, directly from the master tape. A piece of conductive adhesive tape is applied to the rear of the leader between movements at the point where the space is desired. This results in automatic spacing between movements. When the end of the tape is reached, the fast-spiral cut can be done manually—by pressing the upper fast-spiral button—or automatically by means of another set of electrical contacts applied to the active side of the "end leader" of the master tape. When the cutter carriage contacts the end stop, the lathe can be made to produce a locked spiral if the operator desires. Reset back to the starting point is manually initiated by pressing the extreme left-hand button. This button is surrounded by a large cup to prevent accidental reset during normal operation of the lathe.

AUTOMATIC PITCH VARIATION

Automatic pitch variation was accomplished in the first model of this lathe by a servo-operated hydraulic valve which varied the pitch automatically in response to information derived from an advance head on the tape. This system suffered from many troubles. First, any servo containing a reversible motor and a position-sensitive indicator is subject to hunting if it is made too responsive. Because of this speed-of-response limitation, it was necessary to secure information considerably in advance of the normal playback head to allow ample time for the motor to actuate the valve sufficiently to accommodate the sudden fortissimo passages common in symphonic music. Isolated cymbal crashes were particularly annoying in this respect.

To overcome this limitation of the servo-operated valve, the multiple-valve system was adopted. Tests on this improved system indicate much less lag in response. Further, the associated electronic gear is much simpler and less expensive. The exact electronic system used to operate the solenoids will not be described here, since patents covering them have not yet been issued. This much can be said, however: They operate on the basis of the equivalent peak groove amplitude rather than average amplitude or "quasi-average" amplitude. Optional means are provided for predicting accurately in advance where groove echo is likely to occur and for taking immediate action to prevent it from happening.

LUBRICATION

A certain fundamental philosophy has proved helpful in attaining smoothness of operation in this lathe. Oil films which we have tried as lubricants have been found unsatisfactory, since a recording lathe may stand unused for days and then be expected to operate immediately. No oil film seemed adequate, so we have used non-metallic self-lubricating bearing surfaces. Such relatively new combinations of materials as Nylon and polished steel, Teflon and polished steel, etc., make excellent bearing surfaces exhibiting greatly reduced rubbing noise. Even the lowly and inexpensive combination of grey hard fiber and polished steel is useful for some of these bearings. We recognize that such statements may come as heresy to established machine-tool makers but how often have they been required to cut surfaces smooth to less than 3/10,000,000 in.?

ACKNOWLEDGMENT

The author would like to acknowledge the help of his immediate associates and many of his friends, who have made numerous contributions during the development of the new Hydrofeed recording lathe.

A New Stereo Feedback Cutterhead System*

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This paper describes the development of a high-quality stereophonic feedback disc cutter system. It points out the design objectives and briefly describes the advantages and disadvantages of various electromechanical motors. A simple single armature double moving-coil design is explored in detail showing the problems in achieving adequate frequency response, sensitivity and separation. The requirement of low effective mass of the moving armature to achieve high recorded velocities at high frequencies is indicated. Also described is the basic design of the magnetic system with four gaps.

To apply motional feedback around a stereophonic disc cutter, most known transducers have a number of disadvantages. A new unique rf transducer system is described, being displacement-responsive and free from inductive couplings to the motor windings.

Since in musical recordings high peak velocities are required for short periods of time, considerable power is required to drive an inertia-controlled armature. A new high-power (400 v amp) amplifier using ceramic output tubes had to be designed. The basic description of this amplifier and its phase corrective networks is included.

The authors show how the complete system can produce high-level recordings with low distortion and good separation over the 20 cps to 15 kc range.

INTRODUCTION

In the early fall of 1957, our thinking was directed toward the design of a stereo disc recording system. It was then rumored that both American as well as British companies were actively engaged in developing such systems, but by most people these remarks were met with skepticism. Little did we know that only a little more than a year later the entire industry would accept stereo disc as a reality. After preliminary investigation, we went into active development of a new stereo disc recording system.

DESIGN OBJECTIVES

The following objectives were set up for the stereo cutter system:

(*A*) To record amplitudes up to 6-mil peak to peak, at frequencies from 20 to 500 cy, and velocities in excess of 30 cm/sec between 1 and 7 kc, and in excess of 14 cm/sec from 7 to 15 kc.

(*B*) To meet the objectives set up in (*A*), with total harmonic distortion not exceeding 1%, including all harmonic components to 15 kc.

(*C*) To obtain a frequency response flat within $\pm$ 3 db from 20 cy to 15 kc and separation in excess of 20 db.

(*D*) To meet points (*A*), (*B*), and (*C*) in both stereo channels as well as the lateral plane.

(*E*) To maintain stability of level, frequency response, and distortion independent of lacquer loading temperature, or aging.

(*F*) To provide a simple, rugged unit with a minimum of parts and no critical adjustments so as to make possible repairs and replacements in the field.

(*G*) To supply a complete system, complete with amplifiers, equalizers, monitoring and switching facilities in one package, ready to be fed from an audio source.

DESIGN

Type of System and Basic Design

In order to meet the objectives set forth in the previous paragraphs, the two known principles of cutter operation were investigated.

The moving iron system was looked at first, but it is known that moving iron systems are essentially in unstable equilibrium as the actual motion of the armature will always be more than predicted by the change of flux. This condition becomes further aggravated at large amplitudes, producing a great deal of odd harmonic distortion, principally third. In order to reduce this distortion, very stiff mechanical systems have to be used in addition to long gap length, which adds another form of distortion caused by the near saturation flux densities in the armature. These reasons

* Presented at the Tenth Annual Convention of the Audio Engineering Society, New York, October, 1958.

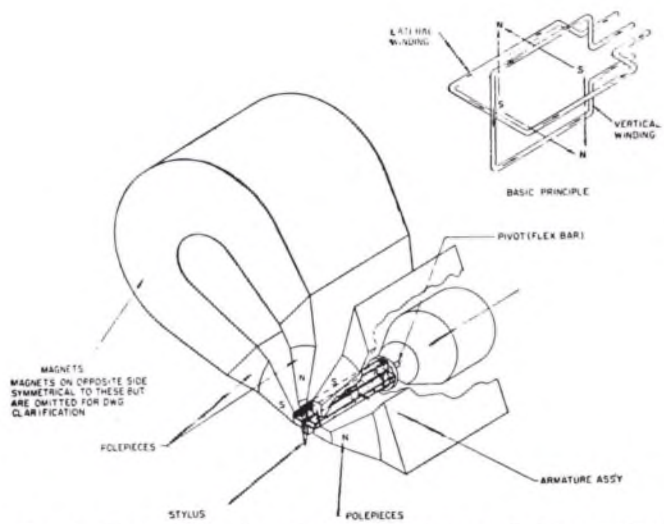

FIG. 1. The construction of the pole pieces and the armature placed between the magnetic poles. The upper right corner shows the basic operating principle, the arrows symbolizing magnetic flux lines between the pole pieces. The arrows inside the conductor show the direction of current flow. As the direction of current flow on each half of the same winding is opposite to the other half, the direction of the magnetic flux lines also has to be reversed in order to get additive motion. Two similar windings are used: one to produce vertical motion, the other lateral.

disqualify moving iron systems within the design objectives outlined.

Our efforts were then concentrated on the moving coil system. To meet objective (F), the cutterhead should have no linkages, so a single armature system was decided upon, actually a single coil form holding two independent windings for the two orthogonal channels (see Fig. 1).

Contrary to moving iron systems, the stiffness in a moving coil unit can be comparatively low and the resonant frequency is usually placed near the middle of the audio spectrum. Moving coil systems are inherently linear, assuming a homogeneous flux field within the limits of motion.

To analyze the performance of such a moving coil system, two adaptations of the basic motor force formula were derived:

for the inertia-controlled region

$$ei = 50\,[(V\omega M)/(Bl)]^2\,Z; \qquad (1)$$

for the stiffness-controlled region

$$ei = 50\,[(D\omega_0{}^2 M)/(Bl)]^2\,Z; \qquad (2)$$

where $e =$ rms volts, $i =$ rms amperes, $V =$ peak velocity in cm/sec, $D =$ peak displacement in cm, $\omega = 2\pi f$, $\omega_0 = 2\pi \cdot 1200$ (resonant frequency), $f =$ frequency in cy/sec, $M =$ mass in grams effective at stylus tip, $B =$ flux density in gauss, $l =$ total active length of wire in motor coil in cm, and $Z =$ impedance of winding in ohms.

From these basic formulas, optimum armature dimensions were derived. Flux density in the gap was assumed to be 8000 gauss, $l = 233$ cm, $M = 0.7$ g, $Z = 5\ \Omega$ at 100 cy/sec, 5.3 Ω at 5 kc and 7.1 Ω at 10 kc, giving 6.5 v amp for 6-mil peak-to-peak amplitude, at any frequency from 20 to 500 cy; 7.4 v amp for 14 cm/sec at 5 kc; and 39.5 v amp for 14 cm/sec at 10 kc. At midfrequencies, very little driving power is required; 0.03 v amp for 7 cm/sec at 1 kc.

Within the framework of the accepted 45–45 system, one can transpose the two input signals to vertical-lateral by the well-known sum and difference method in order to operate on these components in the prescribed way, and then either re-transpose to 45–45 to feed a 45–45 cutter, or directly feed a vertical-lateral cutter. The latter method was chosen since it facilitates the design of the magnetic circuit and the armature and makes the cutting of pure lateral or vertical free of precise channel balancing.

Magnetic Circuit

The dimensioning of the armature, to provide the desired sensitivity in the mass-controlled region based on a flux density of 8–10 kgauss, defined the gap structure as 0.080 in. long by 0.125 in. wide by 11/16 in. deep. Given this four-gap structure and the space above the plane of the recording disc in which to place the magnets and pole pieces it becomes, in essence, a matter of supplying flux between two diagonally placed north poles and two diagonally placed south poles. Several attempts to do this with a single magnet failed due to the need for crisscrossing over the pole pieces to get to the appropriate pole faces, with resultant high leakage, near saturation flux densities, long path, and right-angle bends.

It is generally accepted practice when one desires high flux densities in an air gap to use tapered pole pieces so that the progressively higher net leakage flux, as one moves away from the gap toward the magnet, is fed through a progressively larger pole-piece cross section to avoid saturation and also to get down in flux density to a level at the magnet face suitable for the magnet itself, whose saturation level generally is less than one-half that of good pole-piece material.

In order to avoid the crisscrossing in the magnetic circuit, the use of two separate magnets to feed the four gaps seemed to be indicated. Then, by making judicious use of the available 180° space above the disc for tapered pole pieces and tapered leakage paths, a rather compact pole structure resulted. The pole pieces are tapered both in thickness and width and are fed from horseshoe magnets. In the design of a magnetic circuit an oversimplified but useful procedure is to substitute in Eq. (3) the known dimensions and the estimated values for the leakage factor F and reluctance factor f to obtain the value of B_d/H_d.

$$B_d/H_d = [(F/f)(A_g/A_m)(L_m/L_g)]. \qquad (3)$$

Thus, B_d/H_d defines the operating point on the demagnetization curve of the particular magnet material used, where $F =$ leakage factor, $f =$ reluctance factor, $A_g =$ area of gap face, $A_m =$ area of magnet face, $L_g =$ length of gap, $L_m =$ length of magnet, and $B_g =$ flux density in the gap.

The ratio B_d/H_d is then applied to the demagnetization curve of the magnet material (e.g., Alnico V) to obtain the

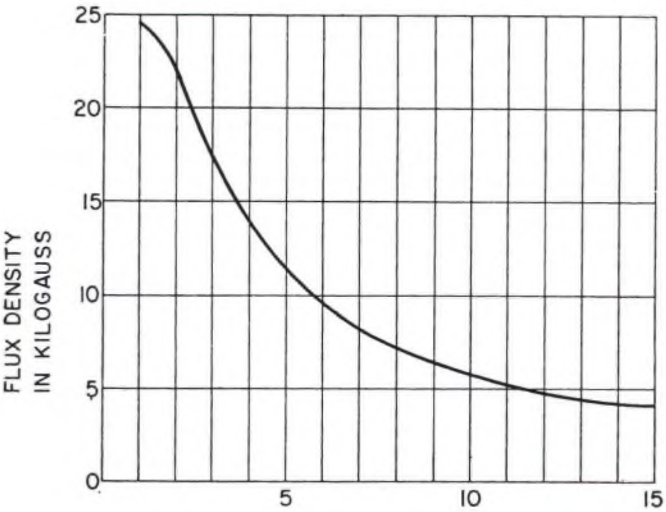

FIG. 2. A plot of flux density in the gap vs. the leakage factor of the magnetic structure illustrating the pitfalls of underestimating the leakage.

value B_d. In Eq. (4), B_d is then substituted to obtain the flux density in the gap.

$$B_g = B_d[(A_m)/(F \cdot A_g)]. \tag{4}$$

Figure 2 is a plot of flux density in the gap vs. the leakage factor for this magnetic structure and illustrates the pitfalls of underestimating the leakage. The only substitute for clairvoyance in a magnetic circuit problem is careful calculation and/or seemingly endless experimentation. We indulged in some of both.

The flux density in the gaps was predicted at 9.1 kgauss and the first model of this design gave 8.7 kgauss. There is some inequality in the two vertical gaps due to the unbalanced leakages. This inequality can readily be eliminated by the addition of a third magnet of appropriate length and cross section should this prove to be desirable, with some increase in total flux. The flux density achieved is entirely adequate to provide the desired sensitivity for the cutterhead. Figure 3 shows the magnetic circuit.

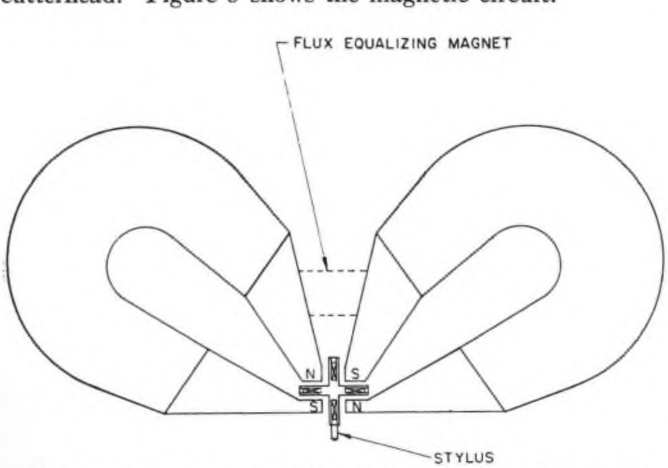

FIG. 3. The cross section of the magnetic circuit and armature.

Armature

As previously discussed, the armature had to meet certain mechanical and electrical requirements in order to conform to design objectives.

In attempting to design the optimum armature, a good number of practical difficulties were encountered. Most of the motions are small in amplitude and, being high in frequency, are very difficult to observe, so a good number of cut-and-try operations had to be followed, including the making of a large-scale approximation (4 ft tall) of the lumped constants. This indicated the importance of making the pivot as short as possible and moving it as close as possible to the rear end of the armature. Even though the basic calculations in the section entitled "Design" indicated that only moderate amount of power was required to drive the cutterhead at relatively high velocities, practice has shown that RIAA pre-emphasis as well as diameter equalization require a considerable amount of additional power at high frequencies. At 10 kc, 5 cm/sec requires only 5 v amp. Adding the normal 13.5 db of pre-emphasis increases the power required to 111.5 v amp, and adding another 6 db of diameter equalization boosts the power requirement to 460 v amp. We have to realize that this power input (460 v amp) would produce a velocity of 42 cm/sec, a velocity that can hardly be tracked with a playback stylus. Reducing the armature mass, of course, is a tremendous help, but everything possible had already been done in this department. The original armature had an effective moving mass of 6 g, the final version only 0.7 g, a reduction of approximately 8.5:1 in mass or 72.5:1 in power. Figure 4 shows the good number of experimental armatures used during development of the cutterhead; Fig. 5, the final version of armature. Figure 6 shows the frequency response of the final armature without feedback. The low mass of 0.7 g was achieved by using the lightest available materials such as magnesium and aluminum, by drilling and hollowing all solid pieces of metal, and by using an optimum amount of

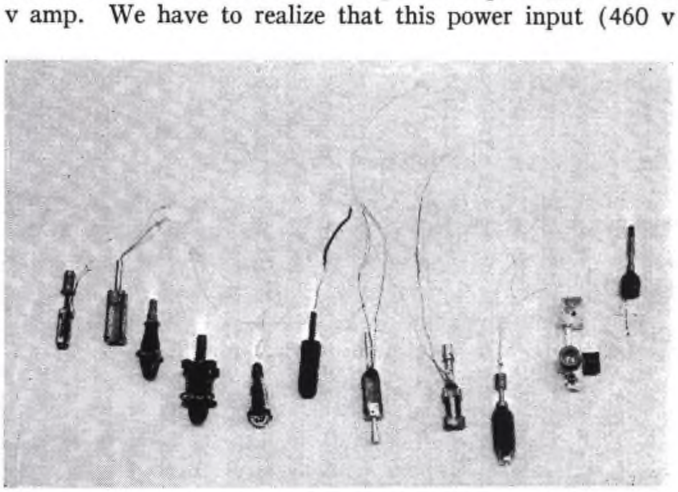

FIG. 4. A number of experimental armatures made before the final version was arrived at; the earliest is on the right, the latest, on the left.

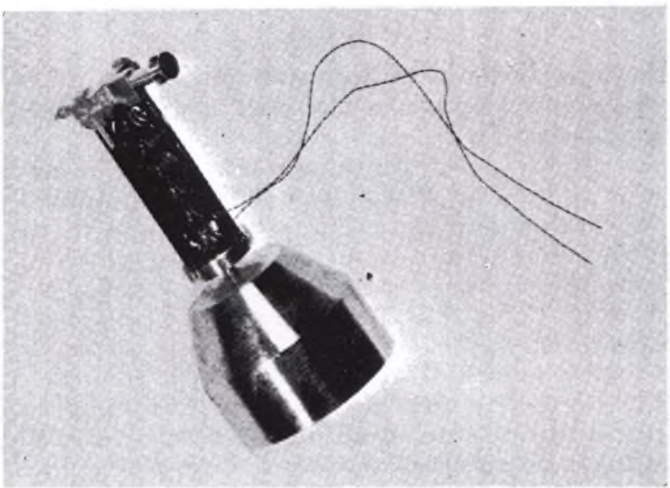

FIG. 5. The final armature. The vertical and lateral windings are visible through the holes in the bobbin. The round magnesium discs visible near the stylus chuck are feedback activators. The thinned-out section near the rear of the armature is the flex bar (pivot).

wire. Due to the high level of power dissipated in the armature, materials had to be chosen to withstand up to 500°F. Ceramic-insulated wire wound on an anodized aluminum bobbin impregnated with high-temperature epoxy fulfilled these requirements. The armatures have withstood all kinds of abuse without breakdown.

Feedback System

Before choosing the motional feedback means actually used in the cutter system, most of the usual transducers were considered and some were tried. Of the velocity-sensitive methods, the magnetic types are subject to unwanted inductive pickup from the two motor coils, low sensitivity, and are rather delicate structures; the dc capacity types, although perhaps theoretically linear, are in practice nonlinear, insensitive, subject to undesired static field pickup, dc leakage with humidity, and difficult to cable. Of the well-known displacement-sensitive methods, the rf capacity type suffers from most of the difficulties of the dc capacity type, particularly stray fields and nonlinearity. The piezoelectric types are so difficult to harness mechanically as to be not worth considering. All of the above involved the addition of some undesirable mass to the armature.

There is considerable merit in the use of a displacement responsive feedback pickup in a cutter system because the region of the audio spectrum requiring feedback the most is that region below its mechanical resonance where the cutter is stiffness-controlled, since it is only in this region that the lacquer loads the stylus appreciably, this loading being nonlinear. Since the cutter is constant amplitude in this region, an amplitude pickup can provide feedback control down to the lowest program frequencies, for example, in this case, 10 db to below 15 cy.

The feedback system used in the cutter is a novel rf system having none of the defects of previously known rf systems† (see Fig. 7, beta amplifier). It is amplitude responsive, provides high level output (100 mv/mil) with low noise (5 μv), is extremely linear, and involves no critical tuning or balance conditions. It consists, in essence, of a variable inductance balanced transducer, transformer-coupled bridge circuit, amplitude-stabilized rf oscillator, and differential detector. It is not a frequency modulation and detection system and does not depend for proper operation on the frequency stability of the oscillator or the associated circuitry. No mass need be added to the armature for the pickup feature, although for convenience we have added small discs which increase the mass by a few tenths of a per cent (see Fig. 5).

The detected transducer output is fed to a modest gain 3-stage amplifier which has its own heavy feedback, in the order of 36 db for frequencies below 500 cy (*V*503, *V*504, and *V*553). The internal feedback frequency response of this amplifier is tailored to provide an over-all feedback amplifier response, such that the system within the motional feedback loop will be constant amplitude below 500 cy and constant velocity above 500 cy.

In addition to the motional feedback, a small amount of selective over-all electrical feedback is used to control the response of the system above the audio band.

Amplifier

As can be seen in the circuit diagram (Fig. 7), the amplifier is rather conventional looking with a few exceptions. New ceramic output tetrodes are used. These tubes, even though half the size of 6*L*6's, are capable of approximately 1000 w in Class B (see Fig. 8). The 4*CX*250*B* tubes in this amplifier are used strictly in Class A, giving phenomenally low distortion (see Fig. 9).

Since, in the recording application, the driving of an inertia-controlled cutterhead requires the greatest power at high frequencies, the amplifier should also have very low difference tone distortion. This unit meets such requirement quite well, having 0.8% of the second-order difference

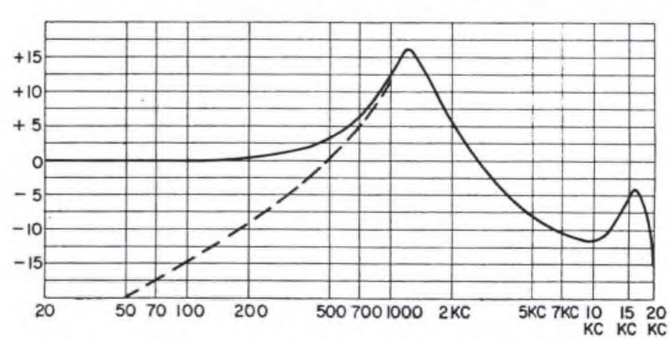

FIG. 6. Armature response without feedback. The solid line indicates the voltage at the feedback amplifier output. This constitutes constant velocity response above 500 cy and constant amplitude response below 500 cy. The broken line indicates constant velocity response of the armature.

† Invented by Norman J. Anderson, patent applied for.

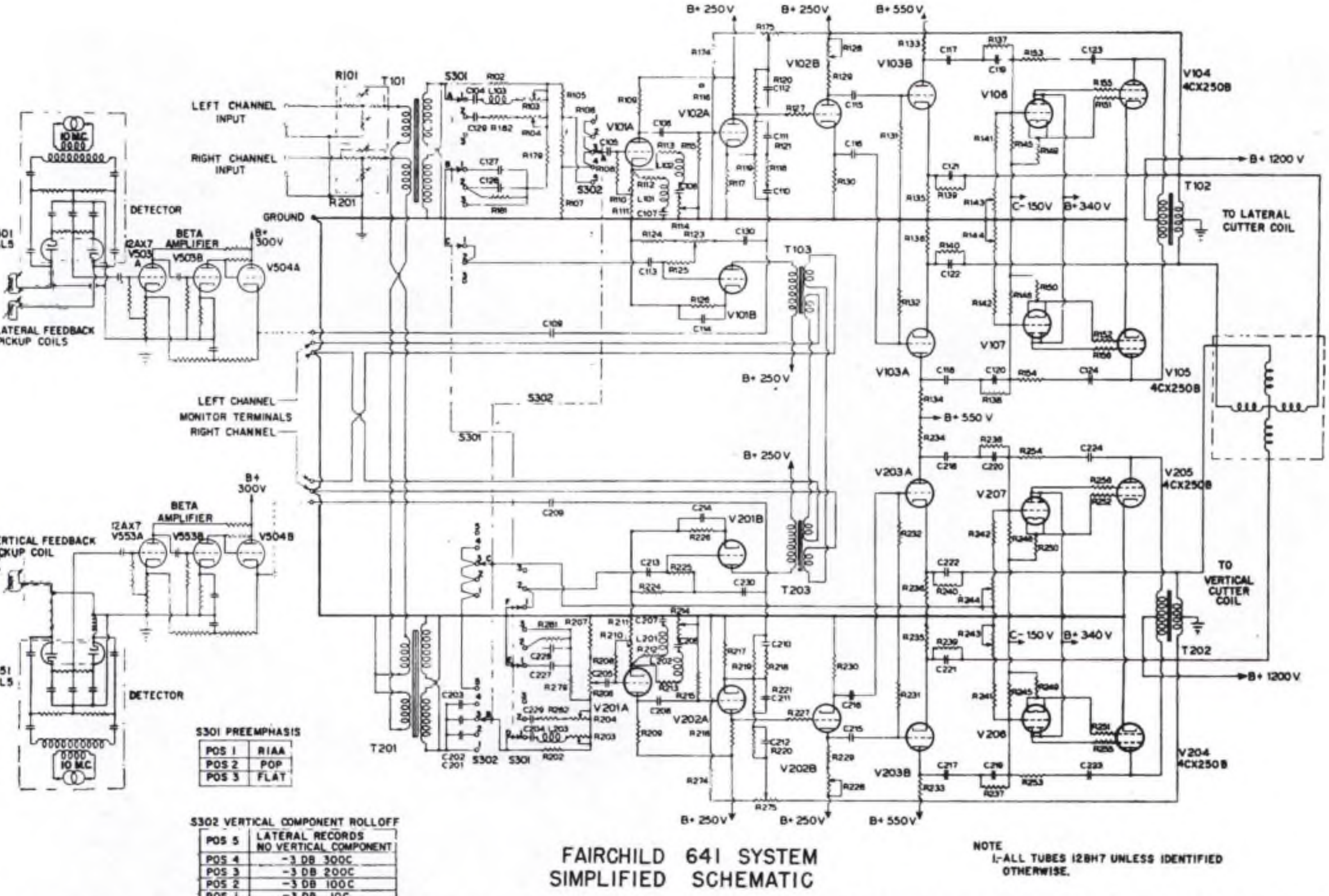

FIG. 7. A simplified schematic diagram of the amplifiers. All metering resistors, all decoupling and filtering circuits are left out for the sake of simplicity. Also omitted are the 10-Mc oscillators and the power supply. The upper right portion shows the lateral amplifier, the upper left the lateral feedback transducer and amplifier; the lower right shows the vertical amplifier, the lower left the vertical feedback transducer and amplifier.

tone at 100 w and no higher order difference tones. At 200 w, the second-order difference tone was 0.4%, the third, 0.6%.

Here is a brief description of the amplifier (see Figs. 7, 10, and 11). The input is first passed through the *R*101 and *R*201 1-db-per-step attenuator to *T*101 and *T*201. The input transformers *T*101 and *T*201 fill a dual function: first, they act as matrixing networks separating each stereo channel into their respective lateral (sum) and vertical (difference) components; secondly, these transformers provide the necessary stepup. The secondary of *T*101 now supplies lateral information, the *T*201 secondary vertical information. The secondaries, through a switch, *S*301, feed preemphasis networks, position 1 being RIAA curve, position 2 POP curve, and position 3 FLAT curve. The second switch shown, *S*302, provides low-frequency roll-off of vertical components as well as deactivating the vertical amplifier for cutting lateral records. This monophonic position (position 5, *S*302) also increases the gain of the lateral channel by 3 db (*R*105, *R*106, and *R*107) to correct for the difference of the lateral component level between monophonic and 45° stereophonic groove ($\sqrt{2}$).

In order for a vertical-lateral cutting system to have good separation between the 45° channels, the amplitudes as well as the phase angle of vertical vs. lateral channel must coincide within close tolerances. To achieve this, several screwdriver adjustments are provided: *R*206 provides balance adjustment, *R*114 and *R*214 adjust 10- to 15-kc region, and *R*175 and *R*275 the 4- to 7-kc region.

From the control grids of *V*101*A* and *V*201*A*, both amplifying channels are identical and only lateral channel will be described.

The *V*101*A* is a plate-loaded stage with relatively large cathode resistors, *R*110 and *R*111, being bypassed by frequency discriminating components *L*101, *L*102, and *C*108. This network is used to compensate for cutterhead deficiencies at high frequencies. The *R*114 can be adjusted if changing cutterheads is necessary. The *V*101*A* is coupled to *V*102*A*, and *V*102*A* is again a conventional plate-loaded amplifier with motional feedback (*R*119, *R*118, and *C*110) from the beta amplifier and electrical feedback (*C*111, *R*121, *R*120, and *C*112) returned to its cathode. The respective *RC* networks are included to compensate for phase at high frequencies. The *R*175 adjusts the amount of electrical

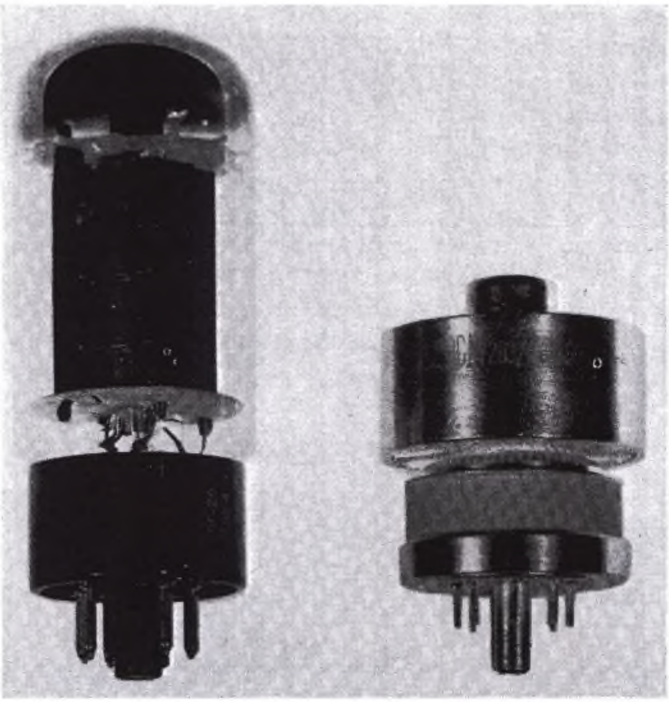

FIG. 8. The $6L6GA$ tube compared to the $4CX250B$. Even though less than half the size of the $6L6$, the $4CX250B$ has 10 times the plate dissipation of a $6L6$, and 20 times the output power capability of a $6L6$.

feedback and thereby influences the amount of motional feedback as well as cutterhead response at mid-high frequencies. It was found desirous to control the electrical rather than the motional feedback in order to retain correct gain and phase relationships.

The $V102A$ is directly coupled to the grid of $V102B$ split-load phase inverter. The variable resistor $R128$ in the plate circuit of $V102B$ is used to balance the drive to the next stages. This potentiometer is adjusted for minimum intermodulation distortion. The $V102B$ is capacitor-coupled to the grids of $V103A$ and $V103B$. The auxiliary feedback network is returned to the cathodes $R135$, $R139$, $C121$, $R136$, $R140$, and $C122$. The $V103A$ and $V103B$ ($12BH7$) are coupled via low-frequency phase shift networks to the grids of the cathode follower drivers $V106$ and $V107$ ($12BH7$).

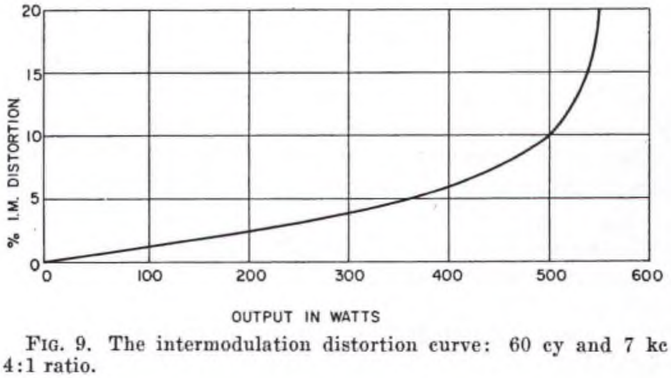

FIG. 9. The intermodulation distortion curve: 60 cy and 7 kc 4:1 ratio.

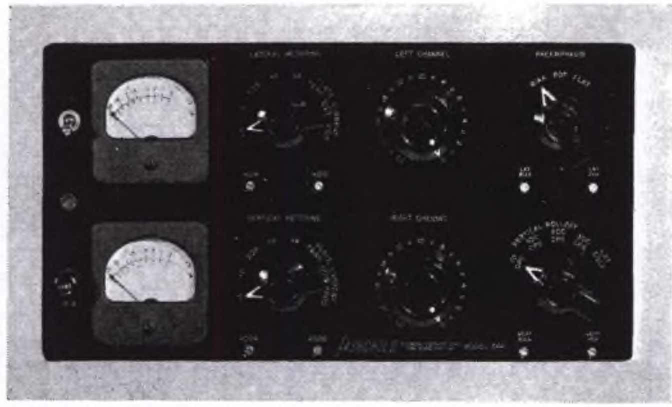

FIG. 10. A front view of the power amplifier showing the metering, attenuation, and equalization facilities.

A second phase-correcting high-frequency feedback loop is fed back to the grid of $V106$ and $V107$ from the plates of the output tubes. The cathode followers are directly coupled to the control grids of the output tubes $V104$ and

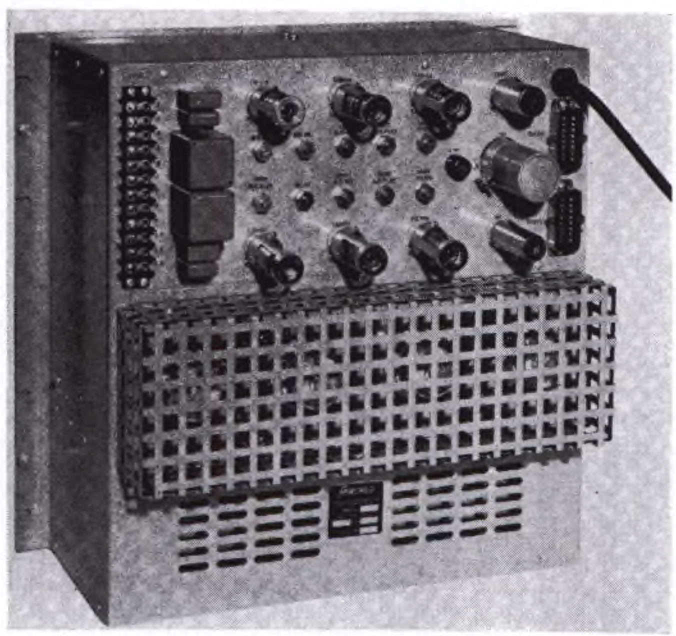

FIG. 11. The rear view of the power amplifier. The $4CX250B$'s are hidden under the perforated cover because of their exposed anode carrying a high voltage and operating at a high temperature. The input and output transformers are hidden inside the chassis.

$V105$ ($4CX250B$), the $4CX250B$'s being transformer-coupled ($T102$) to the cutterhead coil. The output tubes are cooled by a small centrifugal blower and dissipate approximately 150 w each. Since they have a plate-dissipation rating of 250 w each, they have demonstrated their remarkable ability to withstand severe overloads without damage.

To provide continuous motion monitoring while cutting, the beta amplifier output is fed through $C130$, $R123$, and

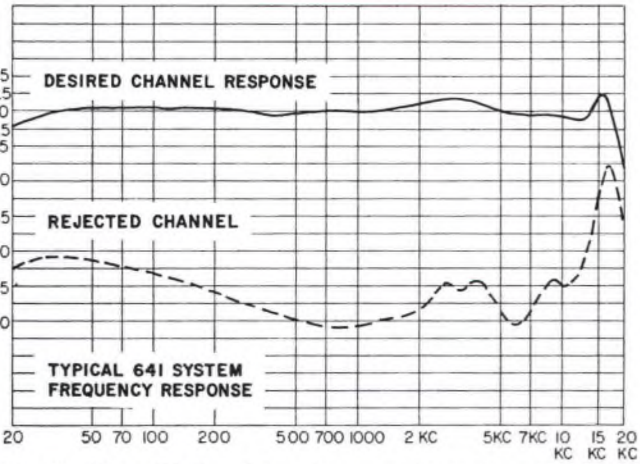

FIG. 12. The solid line shows the complete system response of each channel, the broken line the response of the rejected channel. These curves indicate the frequency response of $\pm$ 2.5 db from 20 to 16 kc and a separation in excess of 20 db below 10 kc.

*R*125 to the monitor tube *V*101*B*. A section of the pre-emphasis switch *S*301, positions 1*D*, 2*D*, and 3*D*, de-emphasizes the audio returned from the beta amplifier (*RC* networks *R*125 and *C*113). The monitor circuit output transformers also have to recombine the lateral and vertical information to the left and right channels, and this is done at the secondaries of *T*103 and *T*203.

CONCLUSION

Since the cutter system is comprised of components described separately in this paper, it is only normal to conclude by considering the results of these components working together. Figure 12 shows the over-all frequency response of the cutter system. This is well within the objectives set forth. All other objectives were also met in a practical unit, including separation in excess of 20 db up to 10 kc, and even above 30 db at midfrequencies (see Fig. 12). Above 10 kc, the separation ranged from 18 to 10 db.

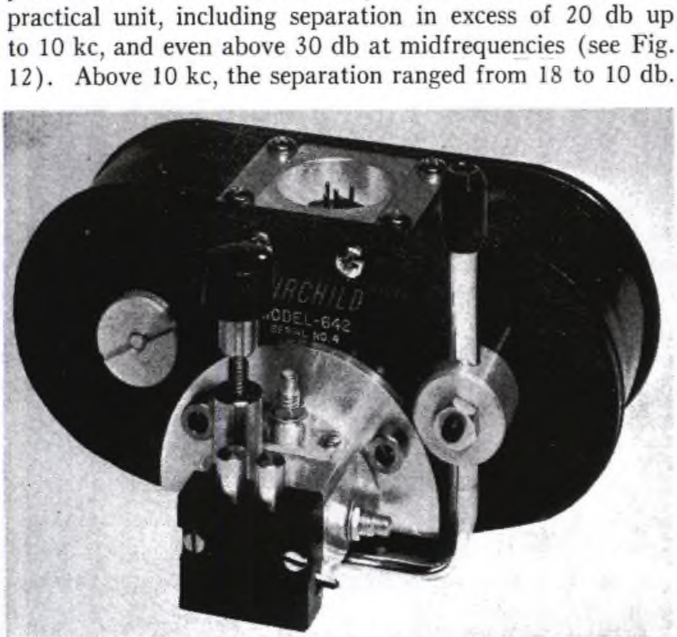

FIG. 13. The final completed version of the cutterhead.

This, however, in no way detracted from excellent aural separation. On the other hand, all of the velocities were met with over 6 db of power to spare, representing the margin of 150 w to take care of unusually high peaks. Figure 13 shows outside appearance of completed cutterhead.

The prime reason for disc recording is not the recording of sine waves but the complex wave form of sound. In the final analysis, a system of this complexity, involving both the cutter and the playback system, can only be judged by a thorough subjective listening test.‡

The authors hope that this system will materially advance the art of stereo disc recording.

ACKNOWLEDGMENTS

The authors wish to thank all members of Fairchild's engineering staff, in particular Mr. Erling P. Skov, Mr. George Alexandrovich, and Mr. John R. Flint for their contributions, both theoretical and experimental. We would also like to thank Dr. Rudy Van Gelder of Hackensack, New Jersey, and Mr. Milton T. (Bill) Putnam of Hollywood, California, in making available their facilities and evaluating the practical aspects of our test results.

‡ Therefore, whenever possible, we tried to confirm our measurements by listening to the recordings and $A-B$ testing them against the original source.

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Norman J. Anderson was a senior engineer specializing in airborne receiving equipment at the Aircraft Radio Corporation from 1934 to 1950. He then served as vice president of the Branson Corporation until, in 1955, he became a consultant. His recent consulting activities have included work on stereophonic reproduction for Fairchild Recording Equipment Corporation and phase measuring equipment for North Atlantic Industries, Inc.

Rein Narma was born in Tallinn, Estonia, where he received his education in electromechanical engineering. He emigrated to the United States in 1951, and worked as a development engineer at Rangertone, Inc., designing film and tape recording equipment. As one of the founders of Gotham Audio Development Corporation, he contributed to the development of disc and tape recording amplifiers. He has also designed several complete multichannel studio facilities. Mr. Narma joined Fairchild Recording Equipment Corporation in 1956 as Chief Engineer. There his activities have included basic development, and supervision of design and manufacture of disc recording and reproduction equipment for studio and home use. Currently he is Vice President in charge of Fairchild's Professional Products Division.

The Westrex 3D StereoDisk System*

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Technical appraisal of existing systems indicated requirements for improved response, increased efficiency, greater stability, and better power handling capabilities. A redesigned cutter and amplifier embody these improvements and incorporate practical operational advantages, typical of which are self-aligning styli and front-panel amplifier adjustments. Plug-in equalizers associated with cutters ensure constant velocity with cutter interchange. The new amplifier is compatible with new and existing cutters.

INTRODUCTION

SINCE the introduction of its StereoDisk System in 1958, Westrex has from time to time undertaken improvement studies with the aim of increasing flexibility, consistency, efficiency and a reduction of dependence on the technical capability of the user. The most recent improvement study resulted in the 3C StereoDisk Cutter and is described in a paper by Frayne and Davis.[1]

Impetus for changing the present system came from appraisal by the record and related industries as related to operating environment, input function and handling requirements. The corresponding study phase was followed by an improvement design effort, with improved performance and retro-fitting of the 3C Recorder as principal objectives. For reasons of economics, it was decided to modify existing equipment rather than to develop an entirely new system. The results are embodied in the 3D StereoDisk System incorporating such disk cutter improvements as:

1. New torque tube; 2. Quick-change, self-aligning stylus holder; 3. Large-diameter stylus; 4. New torque tube support spring to optimize compliance; 5. New drive coil support springs ribbed to reduce resonances; 6. Twice the drive coil impedance to reduce heat; 7. Special high temperature wire and insulation; 8. 25% greater sensitivity; 9. Quick change heater wire clips; 10. Nearly consistent response characteristics for all 3D Cutters.

Before attempting to discuss the new 3D Cutter, the theory of operation of such cutters[1] will be reviewed. The recorder shown in Fig. 1 contains two coil assemblies, one associated with each channel. Each comprises a drive coil and feedback coil located in annular gaps in separate pole pieces. V-shaped beryllium copper coil support springs hold and position the assemblies, and by means of these springs the assemblies are constrained to have no motion other than one parallel to their axis. This motion is transmitted to the tubular stylus support member by means of wire links braced with magnesium sleeves. The magnetic gaps of the drive and feedback coils are arranged in series parallel fashion, and magnetic flux is provided to the system by a single magnet. The arrangement of magnetic paths ensures equal flux densities in the corresponding gaps. The shaded areas between the magnetic gaps indicate copper plugs or shields which reduce the inductive crosstalk from the drive coils to the feedback coils.

STEREODISK CUTTING STYLUS

Replacement or readjustment of the stylus as used in the 3C Cutter necessitates removing the instrument from the lathe, inverting it and, by using a special stylus holding

* Presented March 18, 1964 at the Eleventh Annual Spring Convention of the Audio Engineering Society, Los Angeles.

1. John G. Frayne & R. R. Davis, "Recent Developments in SteroDisk Recording", *J. Audio Eng. Soc.* 7, 147 (1959).

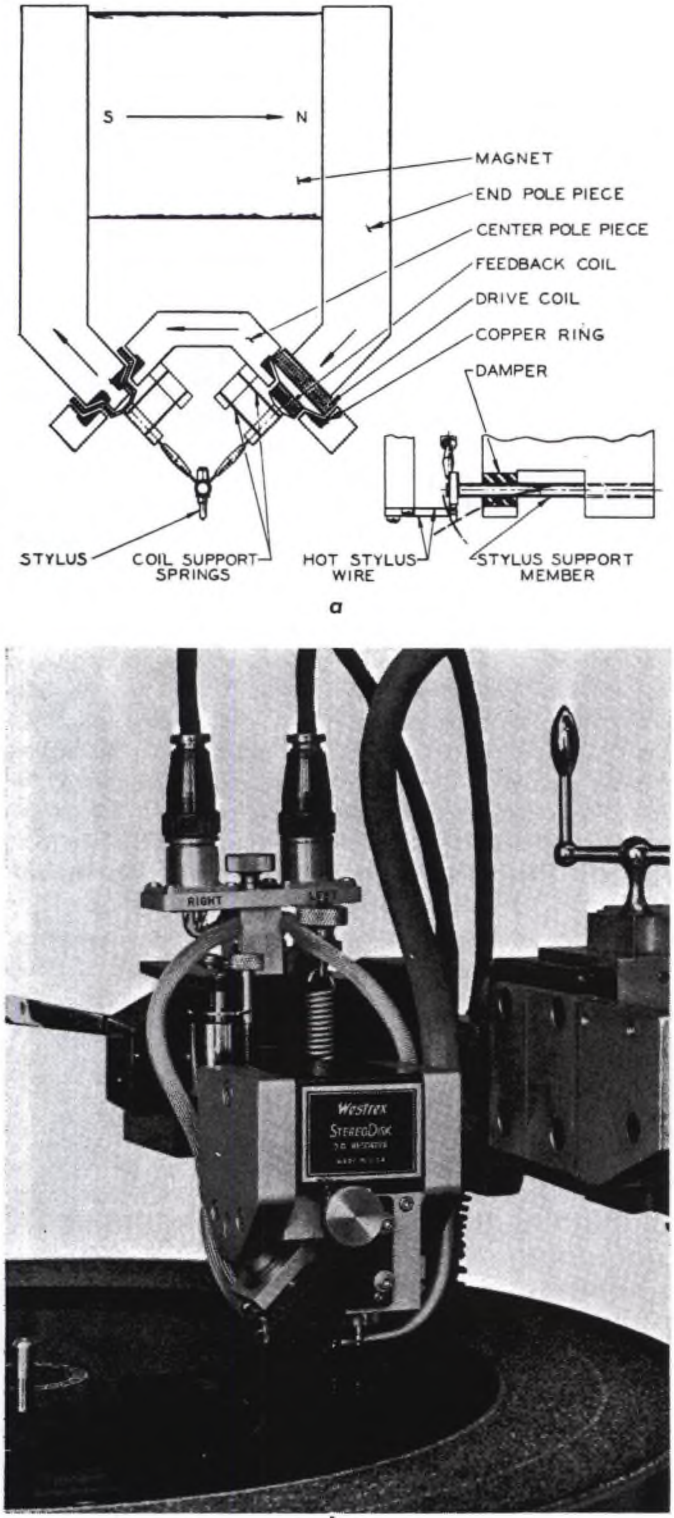


FIG. 1. Westrex StereoDisk 3D Cutter. *a*. Schematic diagram of the cutter. *b*. Photograph of the recorder in position on the recording lathe.

tool, "eyeballing" the cutting face normal to the torque tube (See Fig. 2). The sapphire stylus is cemented into a tapered brass shank which in turn is inside a tapered receptacle in the torque tube. Unfortunately, this type of removable mount is not impervious to foreign materials and accidental physical damage. Repeated manipulation of this combination invariably results in looseness, a consequence of imperfect mating.

To circumvent this problem, a new sapphire stylus and stylus holder has been designed embodying such features as automatic alignment, increased stylus diameter for strength, rigid mounting and the ability to leave the cutter mounted in the lathe. The new one-piece sapphire cutting stylus is 1.5 times greater in diameter than the previous type used in the 3C and has a flattened face 6 mil deep and 39 mil wide ground the full length of the sapphire. This ground surface serves as the cutting face and for flat location for mating to a precision mount designed to

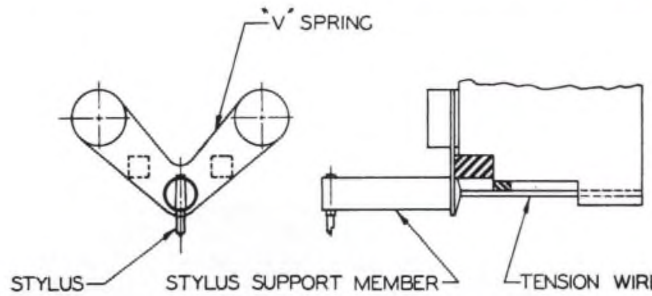


FIG. 2. Schematic diagram of the stylus support for the Westrex 3C Cutter.

give precise alignment of burnishing facets and cutting face. This 0.43-in. long sapphire inserts in a V-way at the end of a new torque tube specifically designed to accommodate the new stylus.

Torque Tube

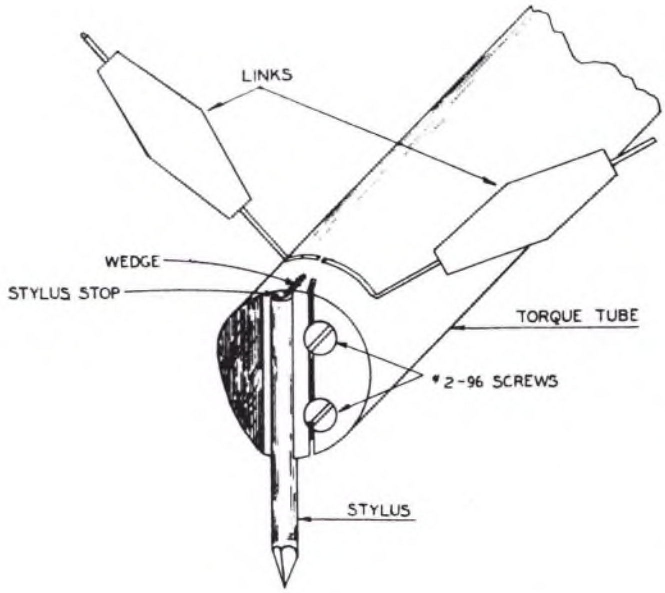
The V-mount designed to hold the new stylus is an integral part of the new torque tube. Two 2-96 screws located adjacent to the V bind on a wedge that in turn applies pressure to the stylus wall, thereby securing the sapphire in a rigid mount (Fig. 3). Accommodating the two 2-96 screws, wedge and V requires a torque tube diameter considerably larger than that used on the 3-type-series cutters. Unfortunately, an increase in size also means an increase in mass and, therefore, a material other than beryllium copper tubing, as used on the 3C, must be used since the previous 3C system already exhibits characteristics of being too heavy. After considerable investigation, hard aluminum was chosen for the new torque tube material with bonding of links and spring to the tube by means of a thixotropic epoxy.

Heater Wire

The heater wire may be quickly changed by depressing two spring loaded wire crimping clips located on the advance ball assembly mounting block.

Coil Support Springs

Drive coil assembly support springs have exhibited spuri-



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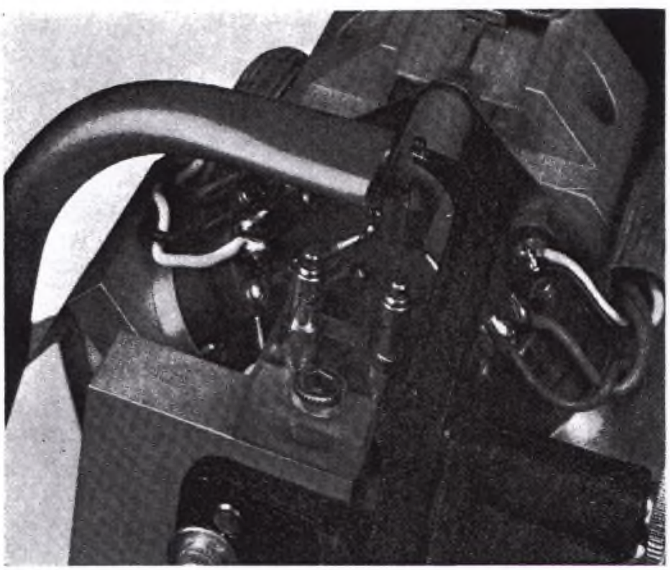

b

FIG. 3. Stylus support for the Westrex 3D Cutter. *a*. Schematic diagram. *b*. Closeup view showing torque tube, self-aligning stylus, and heater wire clips.

ous mechanical resonances in the upper frequency spectrum in 3-type-series cutters. Damping material has been placed between the two springs in an effort to reduce this undesirable effect with, at the best, partial success. Therefore, a new spring has been designed that virtually eliminates undesired resonances and standing waves. This improvement is accomplished by ribbing 80% of the straight portion of the coil-support spring, thereby forcing flexure at the support or mount end of the spring. The results have been very gratifying.

Drive Coil Efficiency

Efficiency of a stereodisk cutter is principally dependent on the mass of the system in motion, total magnetic flux encompassing the drive coil winding, dc resistance of the drive coil winding and a complex pole-zero combination of mechanical and electrical terms. While it is well beyond the scope of this paper to discuss the poles and zeros of mechanical and electrical terms in combination, the effect might be best likened to an M section of a maximally flat network with varying Q. Every effort has been made to reduce mass in all moving parts. Magnetic flux has been maximized by using the highest flux density materials available. The remaining variable, dc resistance of the drive coil, was examined closely in terms of flux reduction, a consequence of increased flux gap with an increase in turns and increased mass caused by additional turns. It is immediately apparent that since force is proportional to flux density times current times the total conductor length, doubling the number of turns would reduce the current required for a given force to half. Since heat loss in the drive coil varies as I^2R, an efficiency increase of 2 to 1 in power (or 3 db) could be realized. To accommodate a larger drive coil, the flux gap must be altered in a manner that reduces the total magnetic flux encompassing the drive coil winding. Additional loss results from fringing flux, now greater as a consequence of an increased gap. Integrating all constants and variables resulting from the higher impedance drive coil was in essence an empirical determination. Averaging test data from several 3C and 3D cutters showed a conservative increase in efficiency of 25% (See Fig. 4). This graph shows comparative I^2R losses of

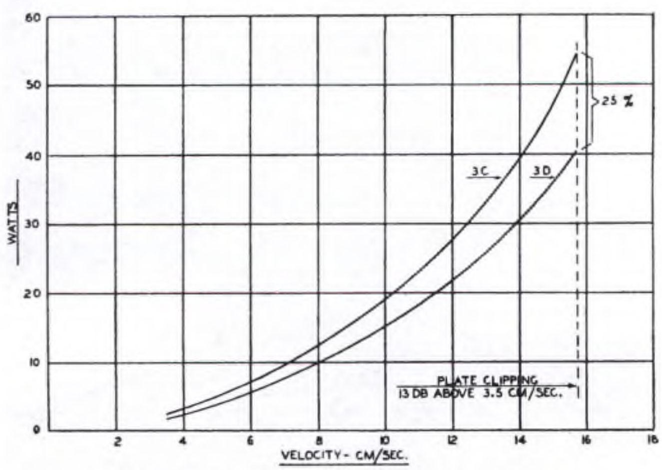

FIG. 4. I^2R losses for the drive coils of 3C and 3D Cutters.

typical 3C and 3D drive coils. The seriousness of excessive power drive on either cutter is plainly seen when viewed in terms of 40 to 55 w of heat dissipated in a mass of 0.4 g. Special high-temperature wire and varnish rated for continuous operation at 220°C for 20,000 hours is used in the 3D cutter. Many tests to prove the worthiness of this new material were made, using repeated frequency sweeps from 1 kc to 15 kc with true RIAA record characteristics and program material causing 3 db of plate clipping in the final amplifier.

The RIAA frequency sweep test was referenced to a level that would record a velocity of 3.5 cm/sec at 1 kc. The new RA-1574-D amplifier was used for all electrical tests and evaluation. Frequencies causing plate clipping power in excess of 75 w in the program material test fall within the frequency spectrum of 7 kc through 10 kc, and were subjected to the drive coil for 15 hours of continuous operation. The 3D cutter did not fail, nor did the 1 amp fuse. Examination of the drive coil after these tests showed that temperatures in excess of 300°C had been reached.

Cutter Feedback

Analyzing phase-shift and feedback characteristics of the 3C Cutter revealed a positive feedback condition from 5 kc to 12 kc (Fig. 5). This function depends entirely on

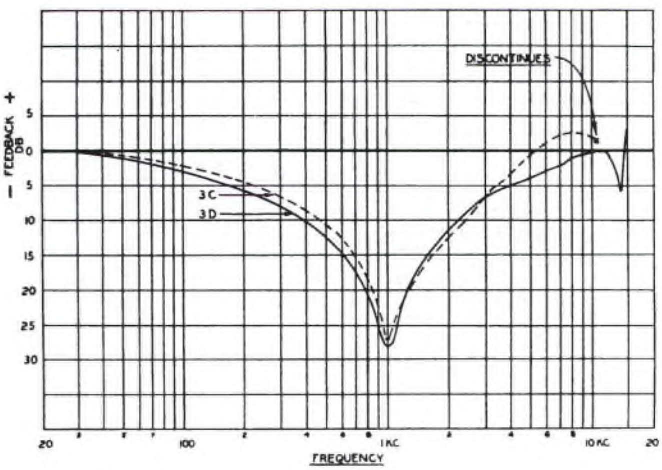

FIG. 5. Feedback characteristics for 3C and 3D Cutters.

the cutter and is independent of any associated amplifier. The effects of positive feedback as related to the 3C Cutter may be thought of as Q multiplication and phase-shifting of the mechanical resonant modes of the suspension system. Different values of feedback shift the phase and Q of

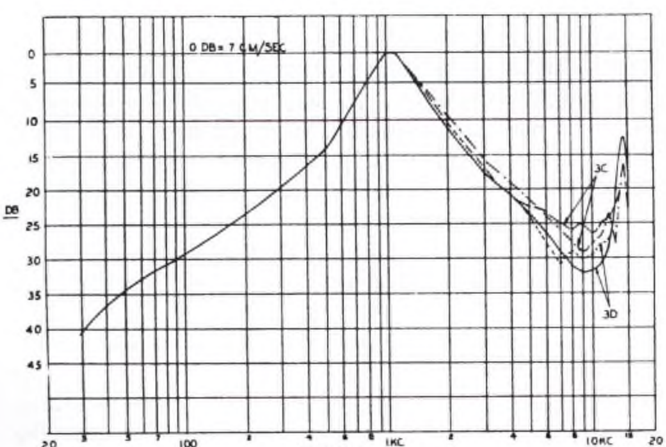

FIG. 6. Resonance characteristics without feedback for 3C and 3D Cutters.

the mechanical resonances, resulting in a frequency response characterized by hills and valleys that do not appear as a family of curves but rather are extreme in magnitude and irregular in position (See Fig. 6). After considerable testing and data integrating, a condition approximating a fourth-order effect was found to result from improper shaping of the copper rings that control inductive crosstalk between the drive coil and feedback coil. After modification of the copper rings, within the limitations of the existing mechanical design, feedback was made negative or unity over the critical portion of the frequency spectrum.

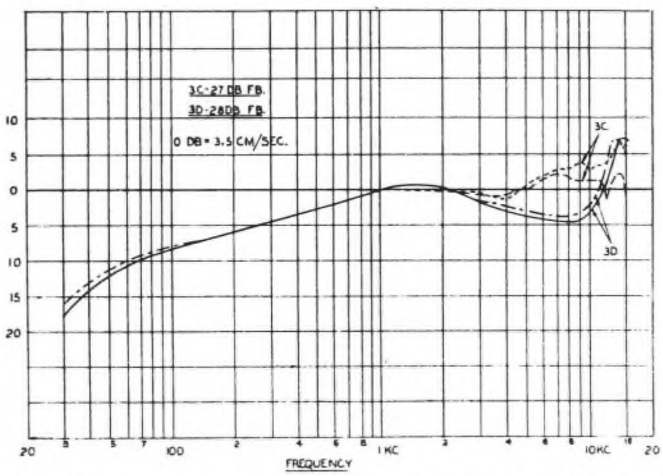

FIG. 7. Response characteristics, with feedback, for 3C and 3D Cutters.

Reference is made to Fig. 6 illustrating the response characteristics, without feedback, of the 3C and 3D Cutters for two extremes of mass and compliance. Note the similarity of extremes between the two cutters. Figure 7 shows the same cutters with feedback, 27 db for the 3C and 28 db for the 3D. The response characteristics of the 3C

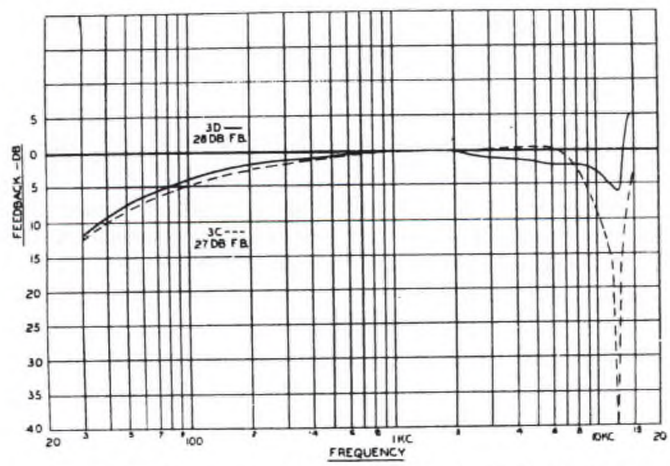

FIG. 8. Feedback coil voltage in db for 3C and 3D Cutters.

2. J. W. Stafford, "Maximum Peak Velocity Capabilities of the Disk Record" *J. Audio Eng. Soc.* **8**, 162 (1960).

3. B. B. Bauer, "The Vertical Tracking Angle Problem in Stereophonic Record Reproduction" *IEEE Trans. on Audio* **AU-11**, 47 (1963).

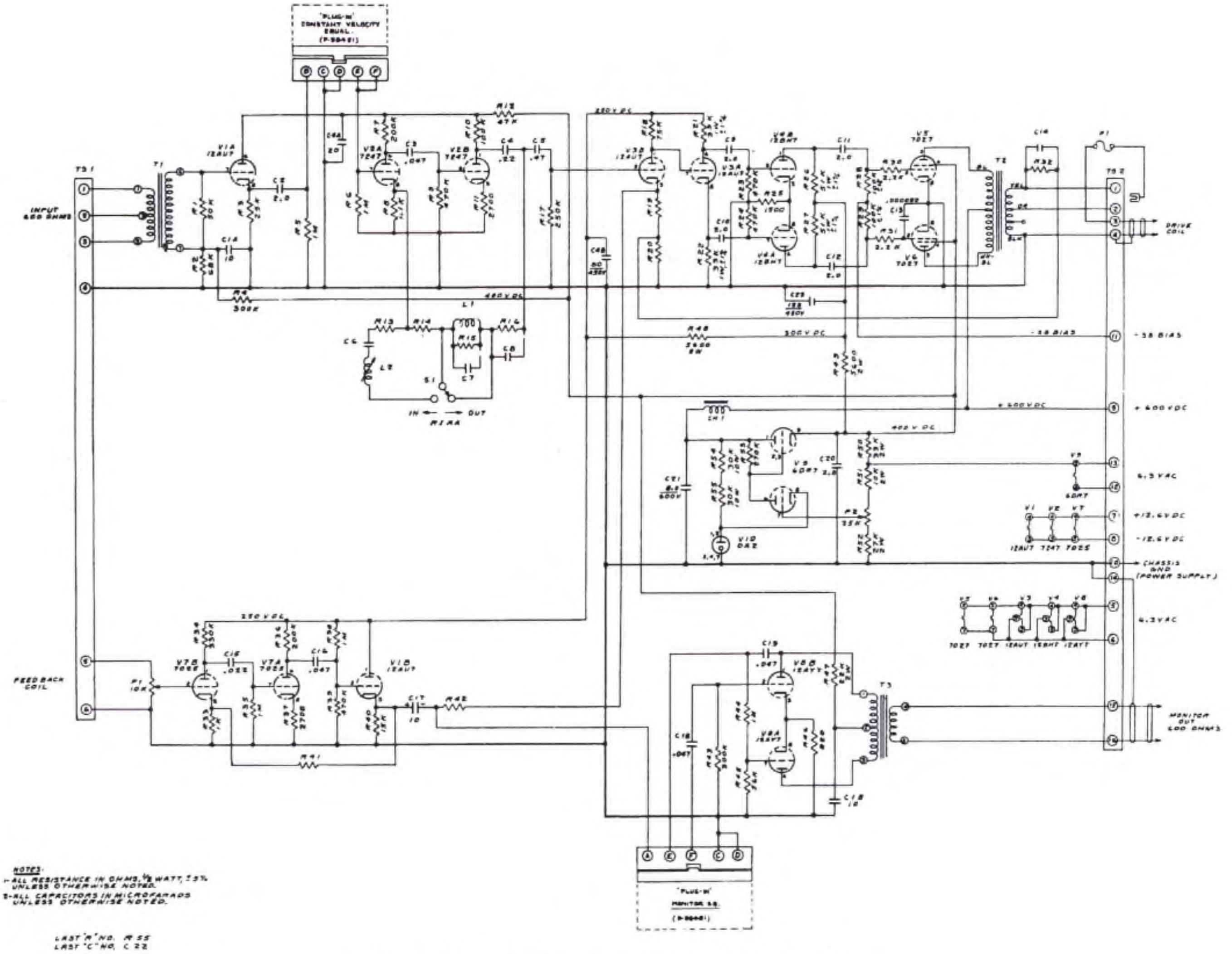

FIG. 9. Schematic diagram of the RA-1574-D Amplifier.

typify the variances expected from a positive feedback condition, while the response of the 3D Cutter with negative feedback is smooth, without hill and valley irregularities, and may be easily equalized by using simple passive networks. The drop in response of the 3D Cutter in the spectral region of 3 kc to 11 kc results from the elimination of positive feedback; i.e., the boost factor is no longer present. A side effect of the foregoing improvement is in the feedback voltage derived from the 3D Cutter. Figure 8 shows the familiar dip between 12 kc and 13 kc associated with the 3C cutter, while the 3D Cutter has no nulling and makes possible true monitoring of the feedback voltage.

RA-1574-D AMPLIFIER

Before attempting to redesign an amplifier for the Westrex StereoDisk System, desirable improvements and deficiencies of existing amplifiers that are associated in use with Westrex cutters, were appraised; the most controversial of these is the power necessary to drive the Westrex cutters.

A study of power necessary to drive a 3C Recorder to the maximum space limits of standard disk recordings has previously been reported.[2] The maximum peak power that would be required in microgroove recording turned out to be about 25 w at 5000 cps in the case of 45 rpm recording.

It should be pointed out that at 78 rpm the maximum peak power approximates 70 w, computed on the same basis. The playback stylus for 78 rpm reproduction is 2.5 mil instead of 0.5 mil as used in the above computations, so that the limiting high-frequency amplitude would be reached before the 70 w peak power level had been used. Since the limiting velocity at high frequencies is proportional to the stylus tip radius, a power reduction of 7 db results.

In view of the foregoing considerations, a power source in excess of 75 w rms is not justified.

Features embodied in the new RA-1574-D amplifier, based on the foregoing analysis and appraisals from the record and related industries, are:

1. Compatibility with all Westrex disk cutters;

2. 75 w power @ less than 1% distortion, 30 cps to 15 kc including all stages;

3. —15 dbm input level for 3.5 cm/sec;

4. +4 dbm output level from monitor amplifier when cutting 3.5 cm/sec;

5. Inverse RIAA in monitor amplifier giving ± 2 db from 50 cycles to 15 kc;

6. All circuits high level, medium and low impedance for increased stability;

7. Constant 600 ohm input impedance to amplifier;

8. "RIAA-IN," "RIAA-OUT" switch mounted in front panel to allow equalization for constant velocity;

9. Plug-in card for precise constant velocity correction for individual cutters;

10. Plug-in equalization card for monitor amplifier;

11. Noise at least 80 db below 75 w;

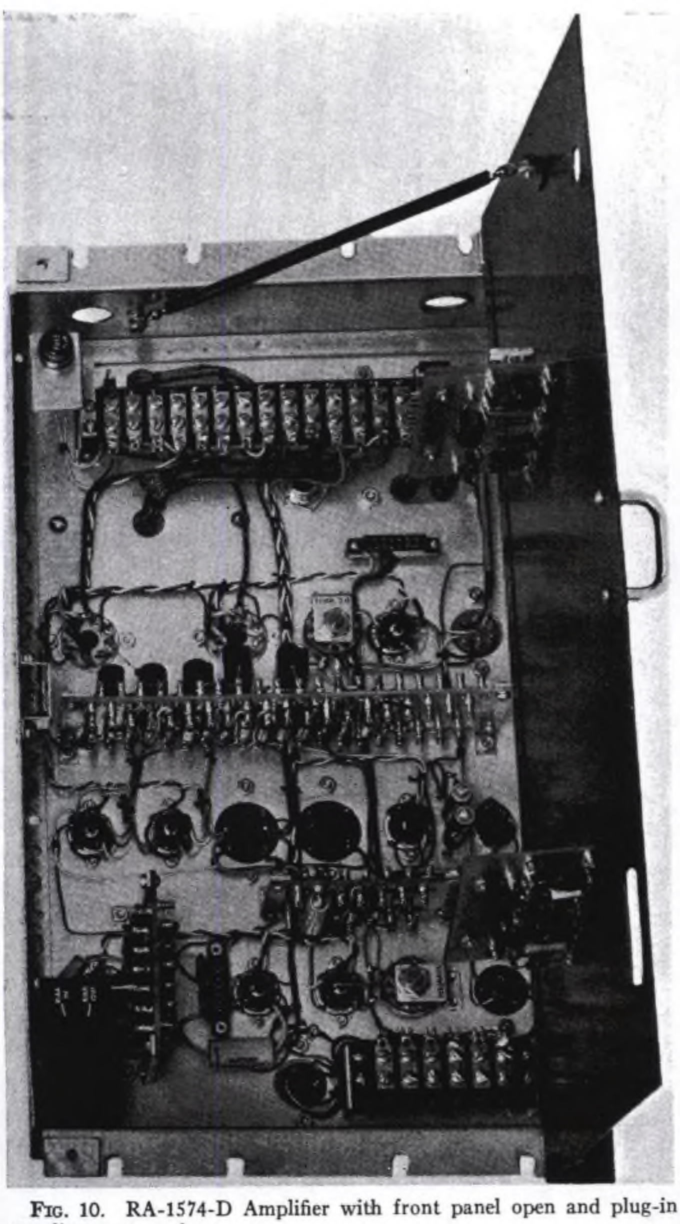

FIG. 10. RA-1574-D Amplifier with front panel open and plug-in equalizers removed.

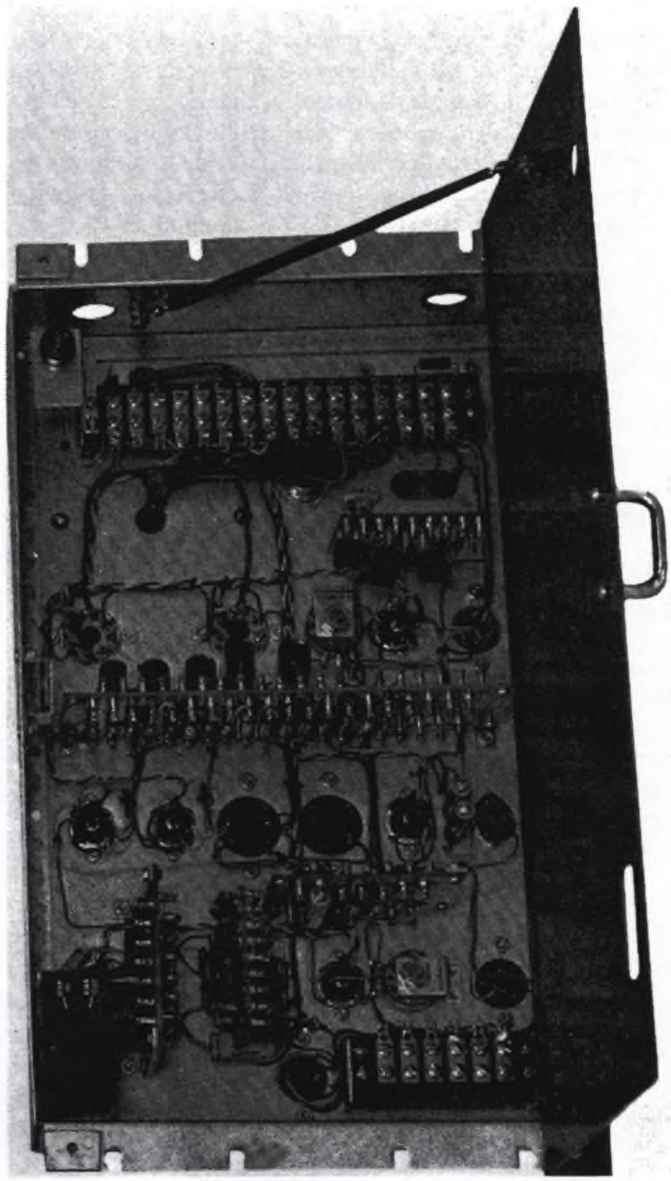

FIG. 11. RA-1574-D Amplifier with front panel open and plug-in equalizers in place.

12. Hinged front panel for quick access;

13. All adjustments on panel side for convenience;

14. Regulated power supply for all stages and final amplifier screens;

15. Regulated bias supply for the final amplifier;

16. Solid state rectifier;

17. Output impedance for new and old cutters.

In an age of micro-miniaturization, transistors and exotic design, it may seem strange that Westrex has retained vacuum tubes and spacious layouts in the design of the RA-1574-D Amplifier. However, this retention of the old has a very good reason. The Westrex monaural and StereoDisk systems are used throughout the world, operated by personnel with varied experiences and language barriers. Material procurement necessitated by component failure is often from foreign local stock, generally not of the calibre, value or tolerance specified in the original equipment.

It is then that the type of material used, ease of repair, straightforward design, margin of design, cost and modification flexibility without the fear of generating instabilities become especially important. In order to reduce the cost of the RA-1574-D amplifier, redesign incorporates the existing RA-1567 Power Supply, output transformer, choke and input transformers from the RA-1574 amplifier.

Description

This redesigned amplifier functions like the previous type but has many additional features not presently available in amplifiers used with the Westrex record cutters. As envisioned, the RA-1574-D Amplifier may be used with any Westrex cutter, past, present or future. This flexibility has been realized by excluding all compensated variables from the amplifier and consolidating them on plug-in cards. High and low-output impedance may be selected by a quick-change jumper. All controls and plug-in cards are accessible from the wired side of the chassis and covered by a hinged panel that opens to expose 90% of the chassis. A series voltage regulator provides constant pressure to the screens of the final amplifier tubes and all stages within the amplifier. Large open-loop gain with heavy feedback ensures a low dynamic impedance, gain stability and phase shift reduction in all stages of the RA-1574-D Amplifier. The schematic for the new amplifier is shown in Fig. 9.

Constant Velocity Plug-in Card

Constant velocity equalization from 1 kc to 15 kc and RIAA equalization from 30 cps to 1 kc is provided by passive networks mounted on a plug-in card (See Figs. 10 and 11). This card is driven from a constant voltage generator, V_1 in Fig. 9, which in turn receives its signal source from input transformer T_1. Termination for the equalizer network is on the plug-in card and is independent of the high-impedance amplifier input it drives (See schematic, Fig. 12).

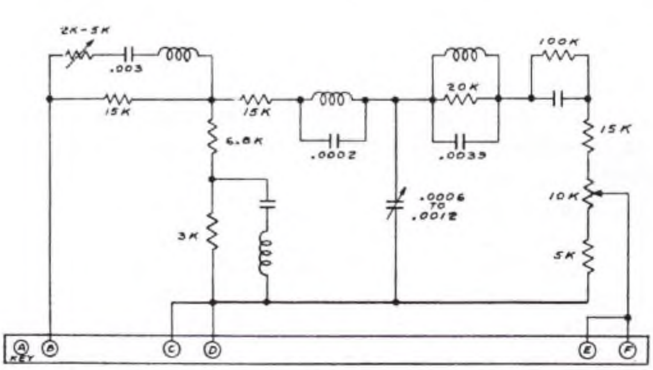

FIG. 12. Schematic diagram of plug-in equalizer card.

One plug-in card will be provided with each channel of the new 3D Cutter and equalized to ± 1 db from 50 cps to 12 kc and ± 2 db from 12 kc to 15 kc.

RIAA Equalization

For reasons of reproduction and light pattern evaluation, the constant velocity method of record cutter equalization is most desirable and has been provided for in the RA-1574-D Amplifier. When the RIAA switch protruding through the front panel of the amplifier cover is in the *OUT* position, the amplifier is flat in frequency response characteristic, and thus does not affect the constant velocity equalization provided by the plug-in card (See Fig. 13).

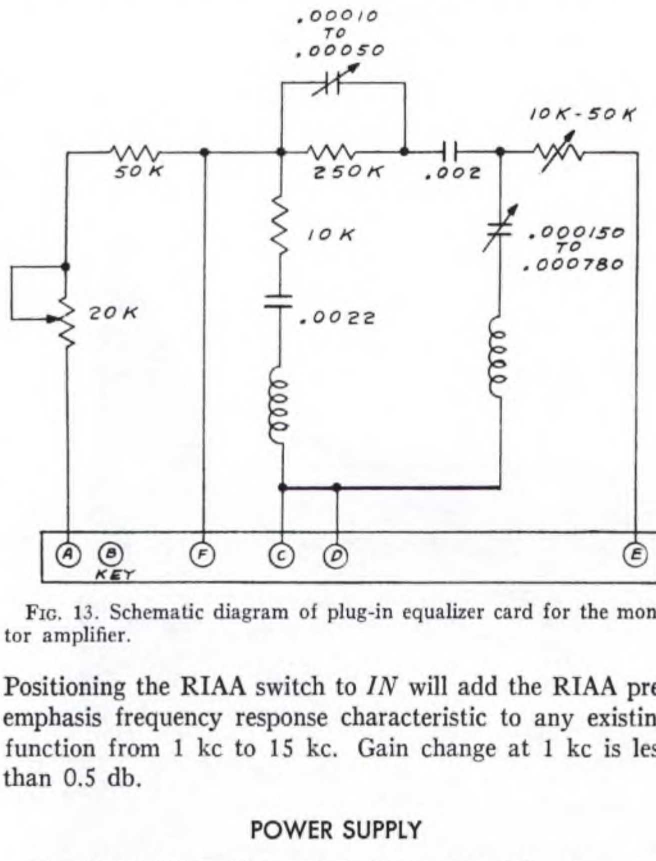

FIG. 13. Schematic diagram of plug-in equalizer card for the monitor amplifier.

Positioning the RIAA switch to *IN* will add the RIAA pre-emphasis frequency response characteristic to any existing function from 1 kc to 15 kc. Gain change at 1 kc is less than 0.5 db.

POWER SUPPLY

The RA-1567-D Power Supply is essentially the same as in previous models. The interlock switch has been removed and is replaced by a potentiometer that provides bias voltage adjustment to grids of the final amplifier. The bias voltage supply is zener-diode regulated. The time-delay relay previously mounted on the RA-1574-C Amplifier chassis is now incorporated into the power supply. The design of this relay is new and greatly improved. Vacuum tube rectifiers have been replaced with solid state types, with wiring remaining as is.

VERTICAL ANGLE

Considerable thought and study was devoted to the vertical cutting angle problem. The excellent work of Bauer at Columbia[3] and Woodward and Fox of RCA[4] was reviewed, particularly to determine what Westrex could do, if any-

4. J. G. Woodward & E. O. Fox, "A Study of Tracking Angle Errors in Stereodisk Recording", *IEEE Trans. on Audio* **AU-11**, 56 (1963).

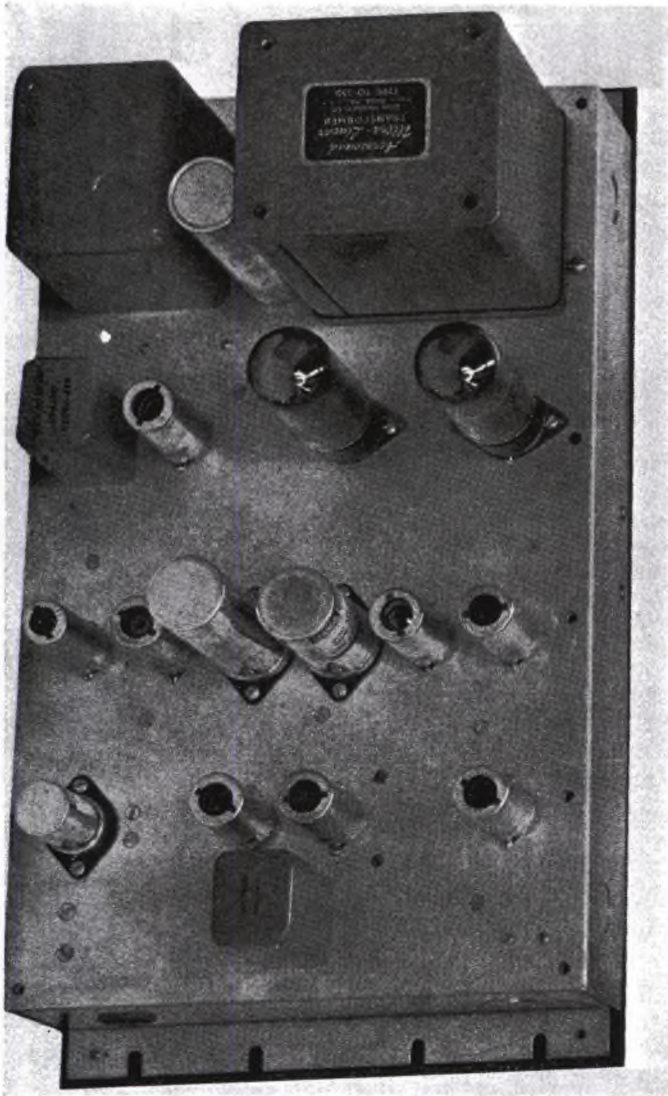

FIG. 14. RA-1574-D Amplifier viewed from the vacuum-tube side.

thing, toward industry standardization of vertical tracking angle. Preliminary tests were made. However, two facts were quite obvious from the beginning—first, standardization is primarily a problem of the reproducing equipment manufacturers and should be agreed upon by RIAA standards before changing present recording equipment specifications; and secondly, the problem of distortion *vs* effective vertical cutting angle is extremely complex, involving such factors as frequency, peak stylus velocities, groove diameter, groove spacing, depth of cut, plastic properties of the master disk, etc., and would require an excessive amount of time and effort for an analytical solution.

On the basis of this vertical cutting angle study, it was decided to maintain the 23° angle as presently established in the 3C Recorder, bearing in mind that in the future a change might be necessary.

CONCLUSION

The design objective of the new Westrex StereoDisk System was to improve fllexibity, compatibility, stability and efficiency of existing amplifiers and cutters by means of mechanical and electrical modification. Every effort has been made to reduce cost by incorporating into the new design many non-moving parts of the earlier 3-type series cutters. The RA-1567-C Power Supply, input transformer, output transformer and choke from the RA-1574-C Amplifier are retained.

ACKNOWLEDGMENTS

The authors wish to express their appreciation to Otto Hepp for his invaluable assistance in this program. Without Mr. Hepp's meticulous attention to detail, patience and exceptional ability as a mechanical designer and precision machinist, the Westrex 3D Cutter would not be a reality.

The authors also wish to thank the many professionals and laymen in the recording industry who gave of their time to advise and appraise the Westrex StereoDisk System.

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Mr. Nelson is a member of Sigma Tau and the Institute of Electrical and Electronic Engineers.

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Jerome W. Stafford, a research engineer with Litton Data Systems Division, received an A.B. in mathematics from Pomona College in 1926, and a B.S. in physics from Massachusetts Institute of Technology in 1928. His experience has been in the design and development of optical recording devices, sound recording systems, cinemascope magnetic sound printing systems with automatic electronic inspection, and associated devices. He also acted as project engineer and consultant on the Radar anti-aircraft fire control trainer and the Radar and visual bombing coordination trainer.

Mr. Stafford holds patents on a range indicating system and a comparator device, and has one pending for a photochromatic analyzer. In 1955 he received a technical award from the Academy of Motion Picture Arts and Sciences for "An Electronic Sound Printing Comparison Device." He has written several technical papers and is a member of various societies, including S.M.P.T.E., I.E.E.E. and the Academy of Motion Picture Arts and Sciences.

A Method for Raising the Load Capability of Stereo Cutters*

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The theoretical attainable velocity of a stereo cutter with feedback is derived and compared with measured values. Present designs are analyzed for mechanical, magnetic, and electrical/thermal limits of load capability. A method of hydrogen cooling has been devised to permit increased driving current (about 6 db).

INTRODUCTION

Moving-coil transducers of high quality for disk recording are designed as feedback systems. The displacements of the cutting stylus are converted back into electrical vibrations by a second coil and fed back to the amplifier driving the system. The attenuation resulting from this feedback makes it possible to place the characteristic resonance f_0 of the vibration system at the center of the transmission range (Fig. 1). At this minimum point of driving current there also occurs the smallest thermal load for the driving coils with a load of, for example, a musical program. Highest thermal load must be expected at the boundaries of the transmission range as a result of the pre-emphasis of the recording compensation, especially at high frequencies. This has been demonstrated in practice with highly compressed recordings of modern popular music. For this reason, in the following discussion, only the upper frequency range is considered for the calculation of the load limit; here the driving power is used mainly for the acceleration of the vibrating mass. On the other hand, the power required for the actual grooving of the recording on the lacquer plate can be disregarded at these frequencies. For the calculation of the maximum achievable recording level of a 45/45° stereo cutter (Fig. 2), it is best to use

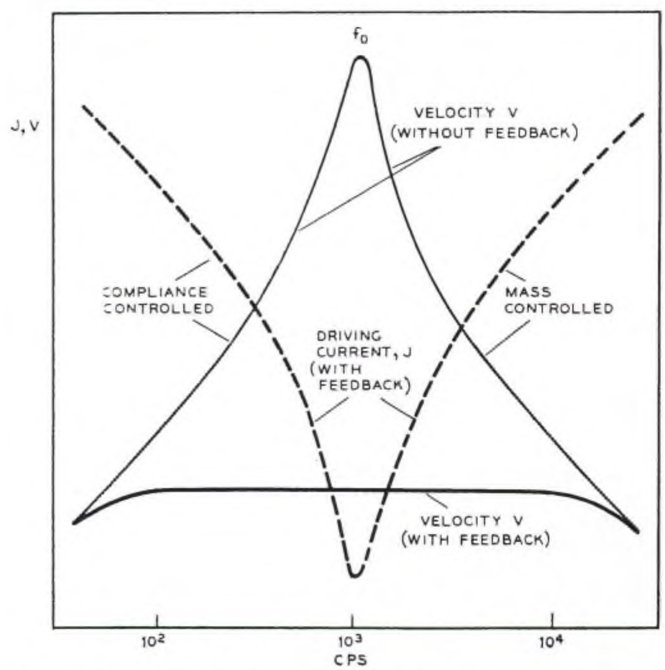

Fig. 1. Driving current and velocity curves for a moving-coil disk-recording transducer.

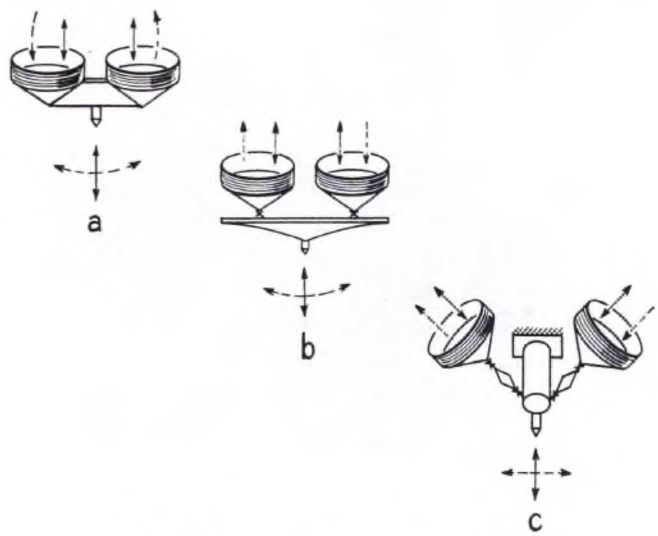

Fig. 2. Possible motions of stereo cutter coils.

* Presented October 17, 1963, at the Fifteenth Annual Fall Convention of The Audio Engineering Society, New York.

the case where both channels are modulated, i.e., both driving coils are under load. Since the systems in question are orthogonal, it is immaterial whether both coils are driven in equiphase or in phase opposition, and whether the resulting recording is lateral or vertical. For transducers which carry out a rotational motion in the lateral component and a translational motion in the vertical component (Fig. 2a, 2b), the vibrating mass is easier to determine than the moment of inertia, so that it is sufficient to consider the vertical component. For pure translational transducers (Fig. 2c), the vibrating mass is substituted for both components in any case.

CALCULATION OF MAXIMUM RECORDING LEVEL

Attainable Velocity Amplitude

The driving force of the coils is given by the formula

$$p=(B\times J\times l/9.81)\times 10^{-6},$$

where p is in Kg, B in Gauss, J in amp and l in cm. For the frequency range in question, this force must be equal to the product of the mass m and the acceleration b of the vibrating system (mass-controlled transducers). Substituting the product of the velocity amplitude v and angular frequency ω for the acceleration, we obtain

$$v\times\omega\times m=(B\times J\times l/9.81)\times 10^{-6}.$$

If the weight G is used instead of the mass m, then the velocity amplitude of the vibration system is

$$v=(B\times J\times l/G)\times(10^{-4}/\omega).$$

An electrodynamic transducer would obviously achieve the greatest speed in the mass-controlled range if it could consist of only the active conductors vibrating in the magnetic field. The coil support, sapphire mounting, etc., should have zero weight. Taking the velocity amplitude in this theoretical case as the maximum speed v_{max} and introducing the product of the length of the conductor l, the conductor cross-section area q and the specific weight γ, we obtain

$$v_{max}=(B/\gamma)\times(J/q)\times(1/\omega)\times 10,$$

where B is in Gauss, J in amp and q in mm^2. Here (J/q) is the current density σ of the coil load. The maximum speed for the copper coils ($\gamma_{Cu}=8.93$) then is

$$v_{max,Cu}=1.1\times B\times\sigma\times(1/\omega),$$

and for aluminum coils ($\gamma=2.7$)

$$v_{max,Al}=3.7\times B\times\sigma\times(1/\omega).$$

This shows that under the above conditions the maximum speed and therefore the maximum recording level depends only on the magnitude of the magnetic induction B in the air gap, the permissible current density σ in the driving coil, and the frequency. The size of the coils drops out of the calculation and is immaterial for the level to be attained.

The above assumption, however, is purely theoretical, because in practice the inactive weight, namely that of the remainder of the armature of the transducer, is added to the active weight of the coils (G_s). The sum of these weights is the aggregate weight to be moved (G). The maximum speed must therefore be multiplied by the driving weight ratio G_s/G. This is the percentage of the driving coil weight in relation to the total weight of the transducer,

$$v=v_{max}\times(G_s/G).$$

This yields

$$v_{Cu}=1.1\times B\times\sigma\times(G_s/G)\times(1/\omega),$$

or

$$v_{Al}=3.7\times B\times\sigma\times(G_s/G)\times(1/\omega).$$

An effective dynamic transducer therefore requires: *1.* a strong magnetic field; *2.* ability to operate with high current densities; *3.* a high ratio of driving coils to the total weight of the transducer (henceforth called "driving ratio").

PRACTICAL MAXIMUM VALUES OF THE PARAMETERS

Magnetic Induction

Using high-grade magnetic materials, for example Koerzit 500 for the generating magnet or a cobalt alloy of Vacuflux 50 for the magnetically intensely loaded pole pieces, even a small pole-piece diameter results in a magnetic induction of over 10,000 gauss in the air gaps necessary for the driving coils. With 7.4 mm pole-piece diameter and 0.8 mm air-gap length, a magnitude of 13,000 Gauss can be obtained in an air gap section area of 35 mm^2. A further substantial improvement is hardly possible at the present stage of technological development.

Current Density

In considering the load limit, it is useful to assume that the yield is maintained until the thermal terminal state is reached (continuous output). The maximum current density must be determined accordingly, since with highly compressed or limited recordings the average modulation is very near the maximum load. Some synthetic insulating resins and adhesives are suited for continuous temperatures of 120°C. Practice has shown that this temperature is not exceeded at current densities from 120 to 130 amp/mm^2 for copper conductor coils and 80 amp/mm^2 for aluminum. These unusually high values are reached because of the transfer of some of the heat across the air gap to the relatively large mass of the magnet structure.

The Driving Ratio

Assuming a magnetic induction in the air gap for the driving coils at 13,000 Gauss and a current density $\sigma_{Cu}=$ 130 amp/mm^2 and $\sigma_{Al}=80$ amp/mm^2 at maximum load, at a frequency of 10 kc the maximum attainable velocity amplitude is

$$v_{10,Cu}=30(G_s/G)\qquad v_{10,Al}=64(G_s/G).$$

Figure 3 shows both curves.

Known practical designs have a driving ratio of less than

50% $[(G_x/G)<0.5]$. For example, a cutter of the type shown in Fig. 2 (Teldec SX45S) employs coils whose weight is 45% of that of the transducer, so that calculation yields $v=12$ cm/sec. The measured load limit for continuous use at 10 kc for this design is 11.5 cm/sec. The calculated and experimental values are therefore in good agreement. Figure 3 might seem to indicate that a higher maximum level could

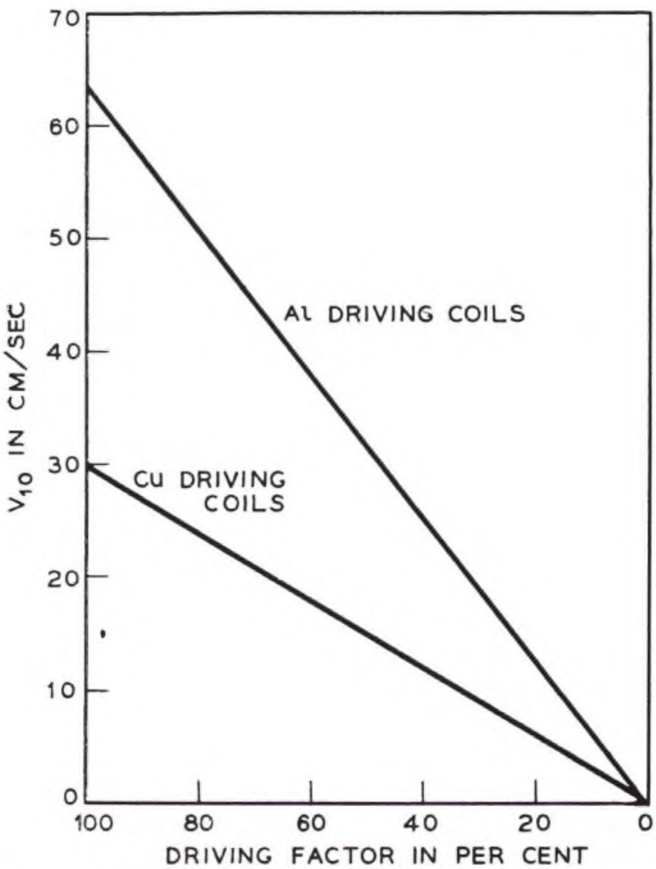

FIG. 3. Driving factors obtainable with copper and aluminum driving coils (B = 13,000 Gauss, σ = 130 amp/mm^2).

be expected with the use of aluminum wire for the coils. This is not true, however, because the percentage weight of the coil is obviously reduced by substituting aluminum coils for copper. Figure 4 shows the level gain of this exchange as a function of the driving ratio of the copper coils. It will be seen that this is advantageous only with a value of over 60%. Practical designs with a better weight ratio are hardly feasible.

As the above calculations and experimental data show, the load capability of a stereo recorder is determined by parameters that depend on the design and the materials. None of the commercial types currently on the market can exceed a speed of 15 cm/sec in continuous output at 10 kc.

A NEW WAY TO IMPROVE THE MAXIMUM RECORDING LEVEL

This speed is satisfactory for the recording of a musical program with a reasonably high level. Brief peaks beyond it are not crucial. Experience shows, however, that it is desirable to have a certain reserve margin in load rating.

For example, a partial resonance in the upper audio range has the effect of an increased driving ratio for these frequencies (Fig. 5). On the other hand, partial resonance reduces the feedback range from 8 octaves (available with cutters that have no partial resonance within the transmission range) to about 3 or 4 octaves. This limits the distortion-reducing feedback.

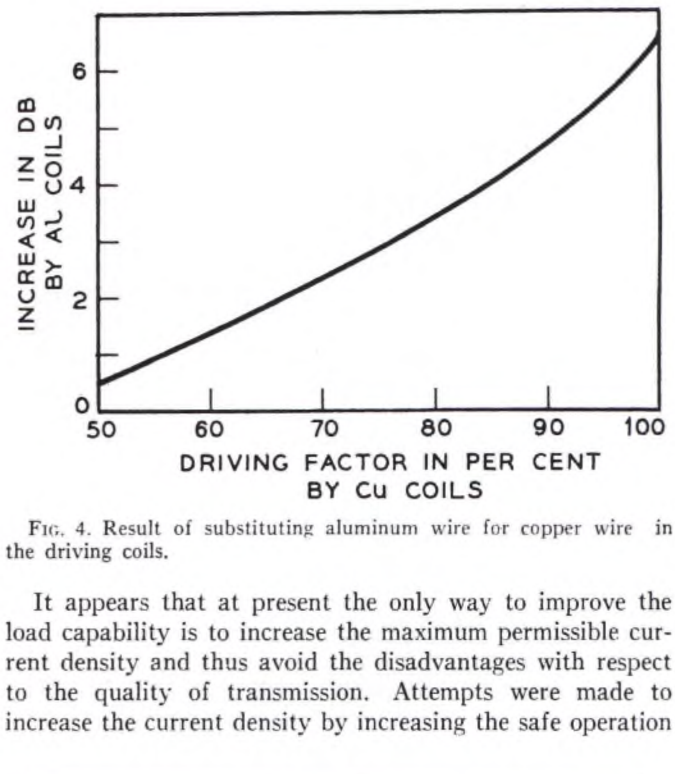

FIG. 4. Result of substituting aluminum wire for copper wire in the driving coils.

It appears that at present the only way to improve the load capability is to increase the maximum permissible current density and thus avoid the disadvantages with respect to the quality of transmission. Attempts were made to increase the current density by increasing the safe operation

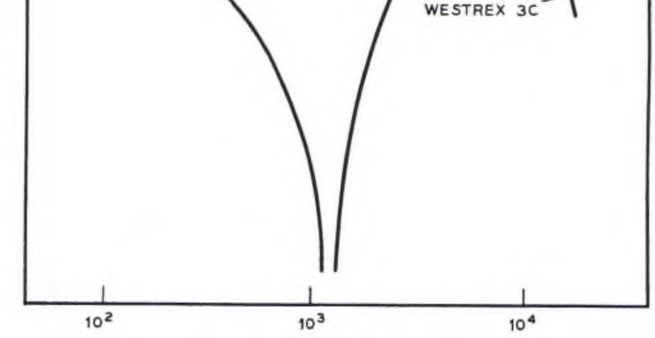

FIG. 5. Driving current curves for the Westrex 3C cutter (constant velocity).

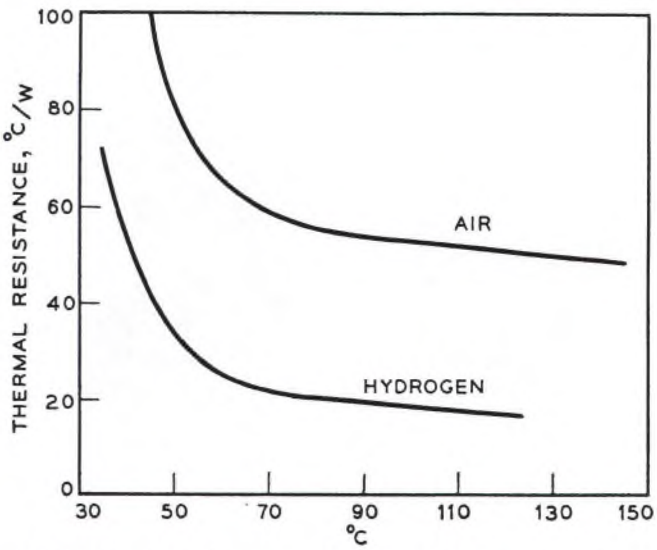

Fig. 6. Thermal resistance as a function of the temperature in the magnetic field gap.

temperature with the use of wires with ceramic insulation. Owing to the higher specific resistance of the special wires that were required, there was, however, no appreciable gain.

At the suggestion of George Neumann of the Laboratory for Electro-Acoustics, Berlin, a new approach was taken, which consists in improved removal of heat from the driving coils. This makes it possible to operate at higher current densities with the same temperature. It is known from semiconductor technology that power-transistor heat re-

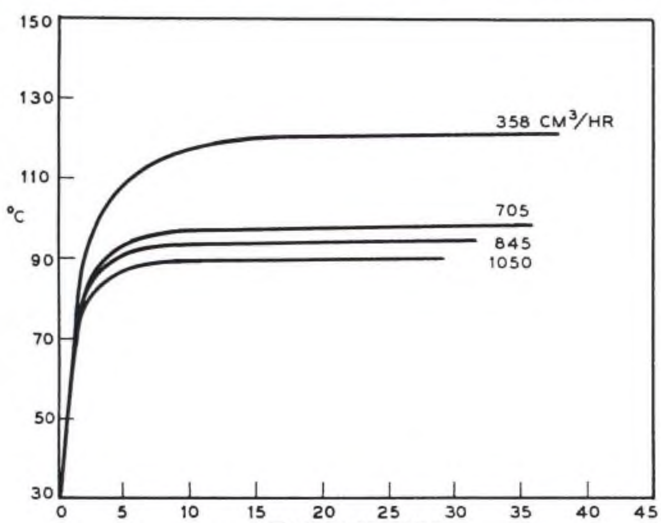

Fig. 7. Influence of the mass of the hydrogen on the temperature of SX45S driving coils.

moval can only be improved with a large cooling-plate area. As the external thermal resistance can be reduced with cooling plates in the case of semiconductors, the heat transfer can also be improved in the air gap of the stereo cutter. This can be done by introducing hydrogen instead of air, which is a poor heat conductor. Figure 6 shows the relative thermal resistance for air and hydrogen, obtained as a function of the temperature in the magnetic field gap. In view of the fact that hydrogen is very volatile, it must be continuously replenished. Figure 7 shows the flow of hydrogen through a magnet gap with total cross-section area of 10 mm^2. One litre per hour appears to be a good compromise here. The maximum level gain is shown in Fig. 8. At the

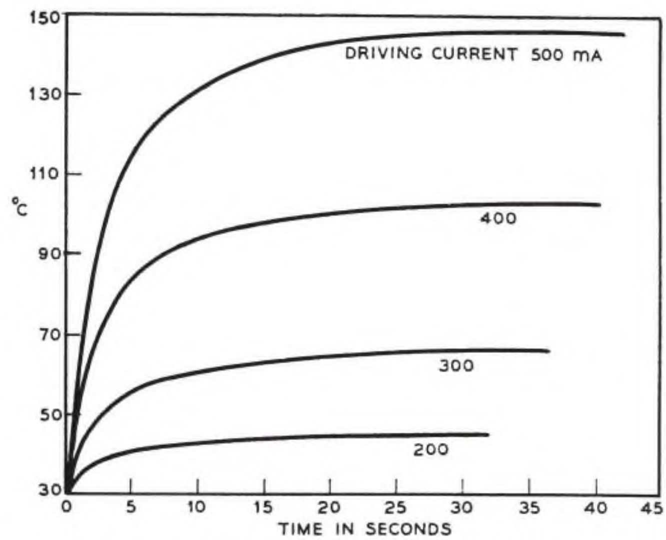

Fig. 8. Level gain with various driving currents for SX45X driving coils heated in air.

load limit a level increase of almost 6 db is possible. Figure 9 illustrates an additional advantage. The cooling is considerably more rapid at the same input power; this means that with intermittent use, the gain will be greater than the 6 db obtained by statistical measurement.

In practice the necessary hydrogen can be obtained by

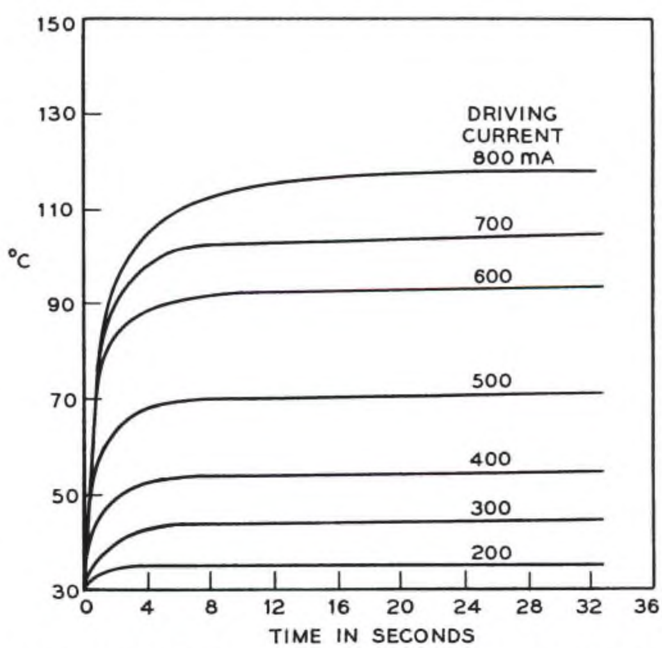

Fig. 9. Level gain with various driving currents for SX45X driving coils heated in hydrogen.

electrolysis at a small energy consumption (approximately 5 w) in the rerecording unit. It is important to purify the gas from entrailed fluid particles.† The small amount of

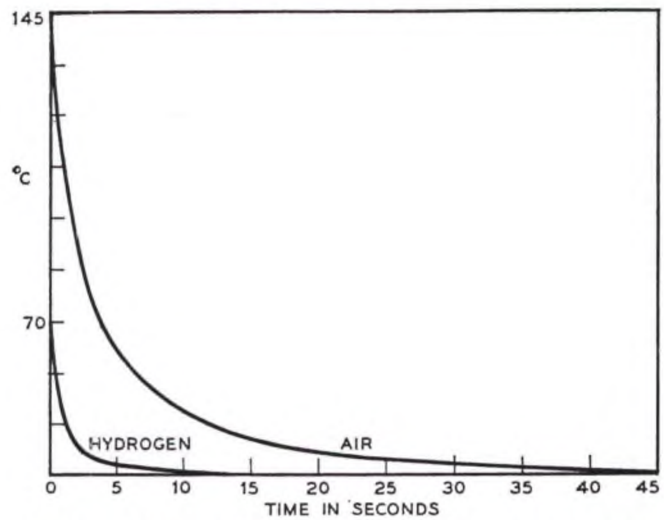

FIG. 10. Cooling of SX45X driving coils in air and in hydrogen (driving current = 500 ma).

hydrogen escaping from the recorder requires no protective measures. It has been demonstrated in practice over a half-year period that the method described here is successful.

† *Editor's Note:* It has since been suggested that helium be used in preference to hydrogen to avoid corrosive effects sometimes encountered with the latter gas when impurities or water vapor are also present due to the simple extraction process employed.

THE AUTHORS

HORST REDLICH

HANS-JOACHIM KLEMP

Horst Redlich was born in 1921 and studied engineering in Berlin, Germany. He joined Telefunken in 1943 as a research engineer, and has since worked on areals, professional tape equipment, magnetic sound recorders, and stereo records. He supervised the development of the first magnetic sound recorder in Germany for 35-mm film (in 1950, with UFA) and the first stereo records in Germany (1955-1958).

In 1951 Mr. Redlich was appointed chief engineer of the engineering branch of Telefunken-Decca (Teldec), and in 1959 he became its technical director.

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Hans-Joachim Klemp was born in 1921 in Germany. From 1947 to 1951 he was a development engineer for the Tobis Company, where he worked on sound-recording equipment for motion pictures. From 1951 to 1954 he was the leader of the Defa Division's group for magnetic sound recording on film, and in 1954 he became manager of electronic engineering for Telefunken-Decca (Teldec).

Phase-Shift Characteristics of Record Cutters and Pickups*

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Methods have been developed for measuring the relative phase-shift characteristics of record cutters and pickups. The methods and results of measurements are described.

INTRODUCTION: Audio literature is replete with data concerning frequency characteristics and channel separation of phonograph pickups, but data concerning phase are nowhere to be found. This omission is attributable to the fact that phase relationships are either imperceptible or insignificant in the reproduction of music, while any deviations from the ideal in response or channel separation are readily noticed.

Phase characteristics currently have become of interest because some systems of recording and reproducing quadraphonic programs on two-track media involve the use of additive and subtractive operations; and for these to be correctly performed, phase relationships should be accurately maintained. We have found that phase in phonograph pickups and cutters generally does not present a problem; nevertheless, it appears care to control it should be exercised in the design of future equipment.

One might add, parenthetically, that phase characteristics of cartridge and cassette tape players also are important, especially when used for two-channel quadraphonic systems. This is a topic for a future paper.

The concern with phase arises when one undertakes to record a quadraphonic program on a two-track medium so as to retain compatibility with stereophonic players. Two methods are possible; carrier or matrix approach.

In the first, the sum of a pair of channels is recorded on one of the tracks in conventional manner, and the difference is recorded on the same track by modulating a high-frequency carrier. The same operation is done for the second pair of channels on the second track. When the carriers are demodulated, they yield the two difference signals. These now are matrixed, or added to and subtracted from, respectively, the corresponding sum signals, to obtain the four original channels. It is obvious that the sum signals and the corresponding demodulated difference signals must remain in proper phase as well as amplitude relationship so that the original signals may be recovered without error. This process requires that both the phase–frequency relationships of the tracks and the phase effects of the modulation–demodulation system be precisely known.

The second approach utilizes the so-called matrixing system of encoding. A number of such systems have been proposed and only the simplest is considered here. A sum signal S is encoded by applying it with equal magnitudes, $S/2$, to both channels. During reproduction, the sum of the two channels produces an output S, while the difference produces no output. A difference signal D is encoded by applying it with equal magnitudes, $D/2$, but with opposite polarity to both channels. Then, in reproduction, the sum of the two channels produces a zero output and the difference produces D. The two signals form an orthogonal set with infinite crosstalk isolation.

* Presented October 13, 1970, at the 39th Convention of the Audio Engineering Society, New York.

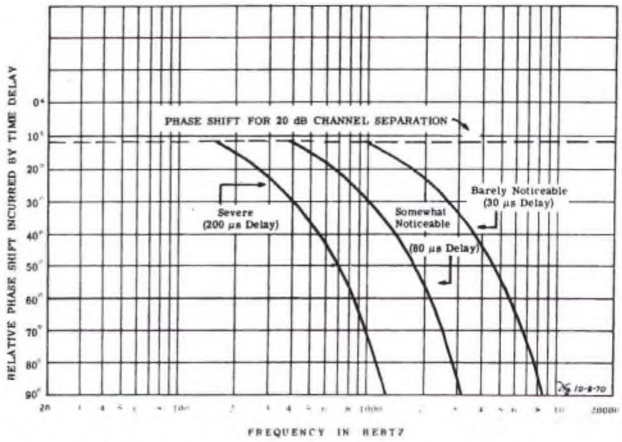

Fig. 1. Matrixed system performance change for a relative left to right time delay.

But let the relative phase between the channels be shifted 11.4 degrees. Upon application of S, the sum is S cos $5.7° = 0.994S$, while the difference becomes S sin $5.7° = 0.0994S$. The channel isolation is reduced to a ratio of the signal voltages, i.e., $0.0994S/0.994S = 0.1$, which expressed in dB is -20. It is noted that the phase–frequency relationship of the tracks *per se* is not important in this instance; only the relative phase shift between the tracks is considered.

ALLOWABLE PHASE SHIFT IN A MATRIX SYSTEM

A series of experiments was performed to determine the audible effects of relative interchannel phase shift produced by time delay in a matrixed system. Basically, a matrixed signal pair was generated from four music tape tracks and applied to a stereo tape recorder whose magnetic playback head could be skewed in order to effect a time delay. The time delay is related to phase by the equation $\phi = 360\ f \cdot t$ degrees. Both a high-speed transport and a narrow-track-width head were used so that any frequency response changes as a function of head skew would be inaudible. The tape recorder output was then rematrixed back into the original four tracks, however, with the effects of the relative channel-to-channel time delay. Quadraphonic loudspeaker reproduction of this music signal was then compared directly with the music tape.

Sound source position alterations as a function of time delay were studied for various types of music. It was found that on the average, a time delay of 30 μs introduced barely audible changes in position. An 80-μs delay produced small, but noticeable shifts, whereas a 200-μs delay produced a severe change in the position from where the sound should emanate and often caused the sound image to move between the loudspeakers.

These results are shown in Fig. 1 with reference to a relative phase shift. The curves define the audible performance change as a function of phase angle and frequency. The broken line at 11.4 degrees is the phase angle required for 20-dB channel separation. Any transmission system whose overall relative phase response (leading or lagging) falls within the lower left regions of Fig. 1 will be altered in sound source placement as indicated. In highest fidelity systems inter-channel delays should be held to much less than 30-μs.

TEST RECORD FOR PHASE MEASUREMENTS OF PICKUPS

To facilitate making phase measurements, a special record was produced. This disc permits examination of a pickup's relative phase angle (left channel versus right channel) over a frequency range of 31.5 Hz to 16 kHz. The spot frequency bands are in increments of one half octaves and have a characteristic described by a 500-Hz constant-amplitude-to-constant-velocity crossover.

The lacquer master was recorded with a minimum relative phase shift by utilizing the variable-speed turntable method in which the disc cutter sinusoidal signal frequency was fixed, and both the signal amplitude and turntable speed were varied to produce discrete bands. An r/min ratio of 16:1 created the same wavelength change on the record. Thus upon replay at a constant speed, the spot frequencies cover this 16:1 range without introducing effects of the cutter phase frequency characteristics. Three sinusoid signal frequencies were used to cut three groups of discrete frequency bands. Upon playback, these provide overlapping phase readings across the audio band. Fig. 2 describes the cutter frequencies, the lathe r/min ratio, and the playback frequencies.

Recording RPM Ratio	PLAYBACK FREQUENCY		
	Low Frequency Group 90 Hz Record Frequency	Mid Frequency Group 1.0 kHz Record Frequency	High Frequency Group 2.8 kHz Record Frequency
.175	500 Hz	5.6 kHz	16 kHz
.25	355	4.0	11.2
.35	250	2.8	8
.5	180	2.0	5.6
.7	125	1.4	4
1.0 (33-1/3)	90	1.0	2.8
1.4	63	.71	2
2.0	45	.5	1.4
2.8	31.5	.355	1

Fig. 2. Description of cutter frequencies, lathe r/min ratio for cutting phased test record, and playback frequencies at 33⅓ r/min.

PICKUP PHASE MEASURING SYSTEM

The phonograph cartridge is mounted in a 12-inch tone arm and adjusted for the manufacturer recommended tracking force. The arm bias force (antiskating) is also adjusted. The stereo preamplifier has an input resistance of 47 kΩ and is shunted by roughly 200 pF for magnetic type cartridges. A 3-kΩ load is used for ceramic cartridges. The output voltage, corresponding to the velocity on the disc, drives a stereo line amplifier. Both the dual-trace oscilloscope and the phasemeter are connected to the line amplifier's low-impedance output.

The phasemeter is an axis-crossing detector which reads the average phase difference between two repetitive signals of the same rate. It does not directly indicate which signal is leading or lagging. Hence, the dual-trace oscilloscope assists in making this judgment.

The phasemeter has an output whose low-frequency component corresponds to the average phase angle. This output is buffered by an oscilloscope dc responsive amplifier and drives a General Radio graphic level recorder.

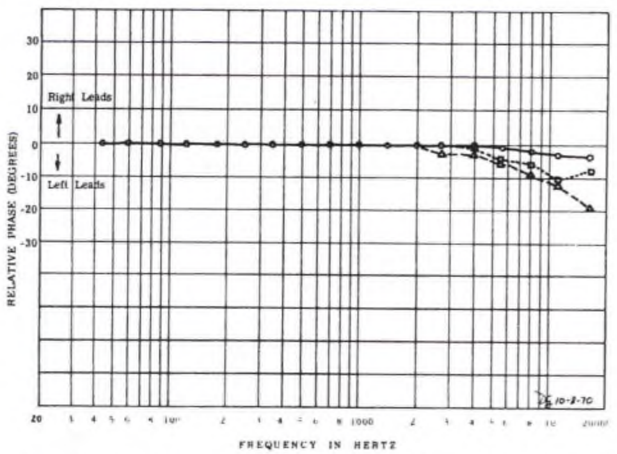

Fig. 3. Relative left-to-right phase shift of three high-fidelity magnetic pickup cartidges measured on pressing CLD-422.

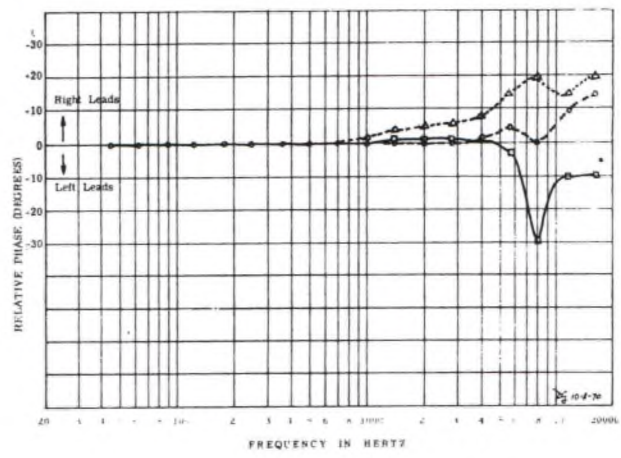

Fig. 4. Relative left-to-right phase shift of three OEM magnetic pickup cartridges measured on pressing CLD-422.

This setup permits automatic plotting of relative phase angle synchronously with a sweep frequency record, which is laterally cut.

PHASE MEASUREMENT METHOD

The left–right phase angle characteristic of phonograph cartridges is obtained by utilizing the special test record. Having been cut with a constant frequency and a variable speed turntable, the record contains no phase variation with frequency within each of the three bands. The procedure is to reproduce the record while reading the phase angle at each frequency. The phase direction, that is, the leading voltage, is observed on the dual-beam oscilloscope. The phase is then plotted at each spot frequency on logarithmic frequency paper.

Three curves are produced: 31.5 Hz to 500 Hz, 355 Hz to 5.6 kHz, and 1 kHz to 16 kHz, each in half-octave steps. Because of the overlapping points, the constant relative phase angle of the cutter at each of the three original signal frequencies can be determined and graphically removed from the combined phase plot.

Once the relative phase property of the pickup cartridge is known, the relative phase angle of the stereo disc cutter may be measured. The cutter is used to record a lacquer containing a sinusoidal sweep frequency across the audio spectrum of interest, such that a laterally modulated groove is obtained. Then, utilizing the phase-calibrated pickup, the sweep frequency record is played through the measurement system which synchronously plots the phase angle. Notation of the leading channel must be obtained from the oscilloscope. The cutter phase angle as a function of frequency is the difference between this curve and the pickup's phase response. Arithmetic subtraction of phase direction must be observed, that is, subtraction of two "leading" angles produces a lesser phase angle, subtraction of a leading and lagging phase produces a larger phase angle.

PHASE-ANGLE RESPONSE OF PHONOGRAPH CARTRIDGES

A sample of nine popular phonograph pickup cartridges was obtained for these phase-angle measurements. Three high-fidelity magnetic cartridges were chosen to represent the audiophile market, three medium priced magnetic pickups to represent the OEM high-fidelity field, and three ceramic type units indicative of the lower priced phonographs.

Fig. 3 shows the phase angle of the high-fidelity magnetic cartridges as measured on the special test record. The ordinate is the relative phase angle between the left and right channels, in which a positive angle means that the right channel leads. It is seen that there is no phase shift error at low frequencies. At the individual frequencies a phase angle of no more than 5° occurs up to a frequency of 5 kHz. One cartridge surprisingly shows less than 5° phase angle out to 16 kHz. Because of this, this cartridge was selected for subsequent disc cutter phase-angle measurements.

In Fig. 4 we see that OEM type magnetic cartridges have more phase angle variation at high frequencies than the high-fidelity types. However, below 5 kHz, the relative phase angle does not exceed 11°. Each cartridge shows the rapid phase changes associated with high-frequency stylus–groove resonance. It should be noted that the maximum angle measured is only 30°.

The relative phase-angle response of the ceramic type pickup cartridge is shown in Fig. 5. Variations in relative phase occur both at the stylus–groove resonant point

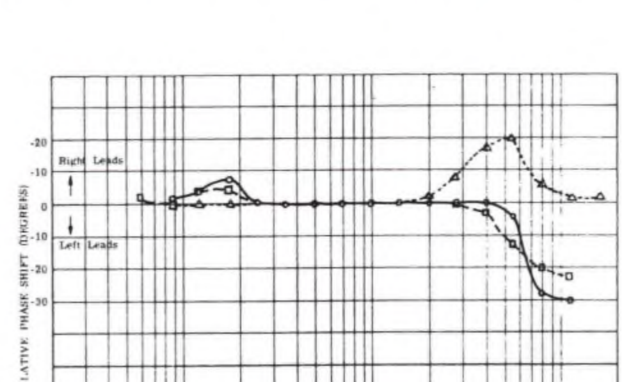

Fig. 5. Relative left-to-right phase shift of three ceramic pickup cartridges measured on pressing CLD-422.

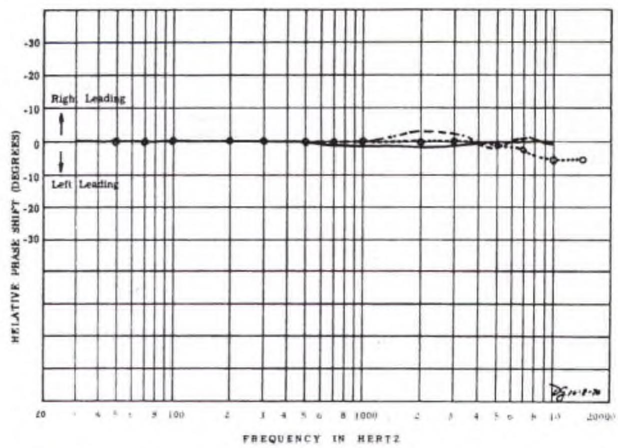

Fig. 6. Relative left-to-right phase shift of three commercial disc cutters.

above 5 kHz and the arm torsional resonant region near 200 Hz. The angles are within 10° below 3 kHz. Measurements at 16 kHz were erroneous because of large signal amplitude variations; the phasemeter limiter could not properly follow the signal of interest. Below 90 Hz, rumble components interfered with the low-level velocity signal and also produced erroneous phase readings.

PHASE-ANGLE RESPONSE OF STEREO DISC CUTTERS

Three modern commercial stereophonic disc cutters were examined for relative left-to-right phase properties as a function of frequency. Two of the disc cutters are utilized in our laboratory for research into recording phenomena. The third cutter examined is used for mastering 33⅓-r/min records in the recording studios. All are unmodified and driven by original manufacturer electronics.

The discs used for measuring the phase of the laboratory cutters were laterally modulated close to the outside record diameter with a 1-kHz velocity level of 5 cm/s rms (3.54 cm/s rms per left/right channel). Recording characteristic is described asymptotically by a constant amplitude below 500 Hz, a constant velocity above 500 Hz, with the signal velocity 3 dB down at 500 Hz.

The studio cutter on the other hand was equalized to conform with the RIAA specification, and hence the lacquer disc was recorded with this characteristic. The lateral velocity level at 1 kHz was −20 dB re 5 cm/s rms, so that high-frequency overloading did not occur. The program consisted of spot frequencies.

The phonograph cartridge possessing the lowest and smoothest phase angle response was chosen for measuring the disc cutter phase characteristics. Its phase plot (Fig. 3) introduced less than 5° at 16 kHz.

The three records were examined at one third speed, full speed, and twice speed, so that any slight frequency-dependent properties could be assessed.

The relative left-to-right phase characteristic of the three disc cutters, as a function of frequency, is shown in Fig. 6. The solid line and the broken line portray the phase properties of the two laboratory disc cutters, and the dotted curve is for the studio cutter. Each curve is made up of two measurements, the one third speed analysis and the twice speed data. Double-speed playback increases the velocity levels for the low frequencies, thereby providing better hum and rumble rejection. Playback at the slow speed transposes all frequencies into the extremely flat portion of the pickup's phase-angle curve. The phase error introduced by the pickup is now less than 1° so that the cutter response could be read directly from the data.

The largest phase angle of the three disc cutters is 5°. One cutter shows a relative phase property of no more than 2°. These few measurements indicate that a properly equalized disc cutter would have only a small phase-angle characteristic over most of the audio range.

CONCLUSION

A review of the data on pickup and cutter phase shifts in Figs. 3–6 shows that little problem is anticipated with matrixing systems using phonograph records and pickups. The phase shift for high-fidelity pickups largely remained within the 11.4° limit needed to ensure a 20-dB channel separation for the dematrixed pairs. Even the lowest cost pickups met the requirement for satisfactory unmatrixing action, albeit the sample tested was too small to draw general conclusions. As an interim measure until more data are available, it is recommended that any pickup intended for use with matrixed quadraphonic discs should first be tested for typical phase performance characteristics.

Note: The biographies of Messrs. Gravereaux and Gust appeared in the September 1971 issue and Mr. Bauer's in the June 1971 issue.

Essential Equipment for the Transmission of High Peak Levels in the Disk-Cutting System SAL 74/SX 74*†

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Today's cutting systems for the production of disk recordings feature high power capability (600 watts per channel). A circuit breaker protection system is described which shields the cutter head effectively from thermal overload as well as from excessive mechanical excursions below the resonance frequency. Furthermore the operation of an acceleration limiter is described which splits the transmission band into two separate bands using the subtraction method. These two bands can then be influenced independently, being reassembled thereafter in proper phase relationship.

INTRODUCTION: Stereo records have been cut for approximately 20 years. Although in the meantime a number of other recording media have gained considerable popularity, disk recording is still one of the most widely distributed information carriers. Every time a new recording medium has appeared on the market, a prediction of the death of the disk has followed, but the ever increasing sales figures of disk recording companies prove that the disk still is and will remain a vital medium. Intensive research and development has continued in the disk field as is the case for electroacoustics as a whole.

HISTORY OF STEREO DISK RECORDING

Over the years there have been a number of generations of cutter heads and associated amplifier systems. Initially the level capability of cutter heads was relatively low, while the constant demand for higher and higher levels on the disk itself pushed system performance to the limit. This often led to the destruction of the cutter head due to thermal overload of the drive coils. Subsequent developments in the cutter head field produced higher thermal tolerance and with it higher cutting levels. Cutter head improvements have been so rapid that we now have program material which will overload the amplifier drive system without damaging the cutter head itself. It was therefore essential that the electronics package be improved and upgraded. The first important step came about seven years ago when the driving package was transistorized. This led not only to higher levels and better cutting quality, it also resulted in great improvement in reliability and stability. The available power was 100 watts per channel, and a completely new and improved circuit breaker system was developed to protect the cutter head against destruction. In this system the temperature of the drive coils was measured using a small amount of direct current through the coil and sensed via a bridge circuit. At a predetermined maximum tempera-

* Translated by Stephen F. Temmer, Gotham Audio Corporation, New York.

† Presented March 27, 1974, at the 47th Convention of the Audio Engineering Society, Copenhagen, Denmark.

ture, the cutter head was disconnected from the power amplifiers, achieving reliable protection. This system was used in the VG 66 S cutting system along with the SX 68 stereo cutter head.

There are, nevertheless, some programs which can only be cut, using this system, with considerable compromises. The result is the development of the SAL 74 cutter drive logic with a power capability of 600 watts per channel, together with the SX 74 stereo cutter head. This system virtually can not be overloaded, either electrically or mechanically, and can cut levels that no pickup can trace. We now have a technology where the recording side is far ahead of the playback capability.

ELECTRONIC CIRCUIT BREAKER

In the beginning it was necessary to assure constant and failure-free operation of both the amplifier and the cutter head. From the electronic side a solution had to be found to prevent the 600-watt drive capability from destroying the cutter head which now could reach its critical drive coil temperature in a fraction of the time the previous 100 watts required. The circuit breaker used in the VG 66 S system was insufficient for this task, since it was too slow acting.

Another problem facing the engineers was the incredible mechanical amplitudes which could be caused by this enormous power capability, which could destroy the cutter head mechanically without thermally overloading it. The solutions of these problems were combined in the new circuit breaker SEL 74.

Temperature Determination

The most difficult problem encountered in protecting the cutter head against thermal overload damage is the measurement of the drive coil temperature. For structural reasons it is not possible to install a passive temperature-sensing element, and so the same parameter which was used in the VG 66 S, namely, the drive coil resistance, is used in the SAL 74. Since the resistance of the coil wire rises with temperature, this is used as a measure. At 25°C the resistance is 4.7 ohms, and it rises to 8.2 ohms at the maximum allowable temperature of 200°C.

It is critically important how the drive coils are connected to the direct-current measuring bridge. It should also be pointed out that the bridge direct current is obtained directly from the output stages as an offset value, since these are designed along the principles of operational amplifiers. Some of the problems are described in the following. The first requirement is the greatest possible bridge voltage deviation. This means that in Fig. 1 R1 should be identical to R2 (drive coil), while R11 should be identical to R22.

Matching the Cutter Head to the Output Stage

The second requirement has nothing to do with temperature measurement, but rather concerns the proper matching of the cutter head to the power amplifier. Since the power amplifiers can deliver their maximum power only if they are terminated in a constant load resistance, it is not possible to ignore the inductive component of the cutter head impedance. It is therefore logical to insert a network (Fig. 1) which compensates the complex portion of the cutter head resistance in that part of the bridge circuit which lies in series with the cutter head. This network is so designed as to present to the power amplifier output, together with the cutter head, a constant load of approximately 9.5 ohms (Fig. 2). For the frequencies under consideration this series network presents practically no power loss.

Accelerated Temperature Determination

A further problem in designing the bridge circuit is the fact that the very low dc voltages are superimposed by ac signals which are several orders of magnitude higher in level. It is therefore useful to define the switchoff point as the bridge's null, since at that point the ac signal voltage is a minimum. The high impedance branch of the bridge circuit was dimensioned with this problem in mind.

In spite of this design, the higher output power, and with it the remaining higher bridge diagonal voltage, requires filtering, which counteracts the requirement for rapid tripping. Even when properly dimensioned, such filtering results in a minimum trip time of 0.7 second when the cutter head is subjected to maximum power from its room-temperature condition, even though actual warm-up time is only 50 ms.

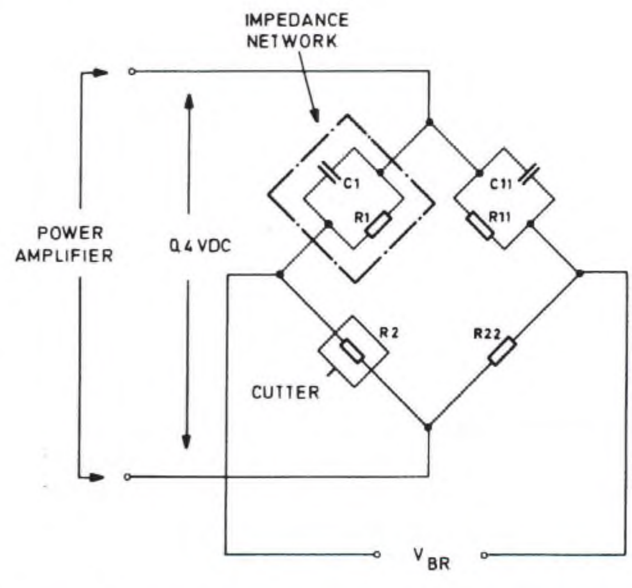

Fig. 1. Test bridge for measuring cutter head temperature.

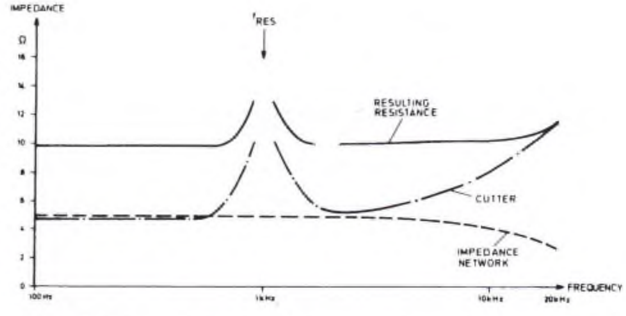

Fig. 2. Cutter head impedance, complementary impedance, and resulting load resistance.

Additional circuitry made it possible to increase trip response by a factor 10, thereby bringing it virtually in line with the actual warm-up curve, while at the same time taking into account the prior thermal load history of the cutter head. What is actually done is to calculate from the momentary cutter head current the expected temperature rise parameter without running the danger of tripping while this is going on. Switchoff only results from the actual temperature measurement and is accomplished by means of reed relays (< 1 ms) (Fig. 3).

The second safety criterium, as previously mentioned, is the maximum safe excursion. This is determined by a simple current measurement and full-wave rectification, and always leads to tripping when a current of 1 A is exceeded at any frequency below resonance.

NECESSITY FOR ACCELERATION LIMITING

A further problem is that the available power must be limited in time during the cutting process. The reasons are as follows.

Amplitudes may arise which, as was already pointed out, may produce excursions far above any that can be played back. Significantly excessive amplitudes should therefore be limited in time so that they lie well below the subjective sensibility time (≤ 10 ms).

Without the time limit on cutting power, as outlined above, critical program content may quickly raise the cutter head's temperature to its tolerance limit, thereby actuating the circuit breaker and causing an undesirable interruption of the cutting process. An adjustable power limiter in such cases serves to allow full utilization of the cutter head's thermal capabilities, and to permit tape-to-disk transfers to a point just below the circuit breaker cutoff.

Since only frequencies above cutter head resonance are of interest on the one hand, and the mathematical relationship between cutter current and stylus acceleration is frequency independent and linear on the other, a device was designed which operates as an acceleration limiter.

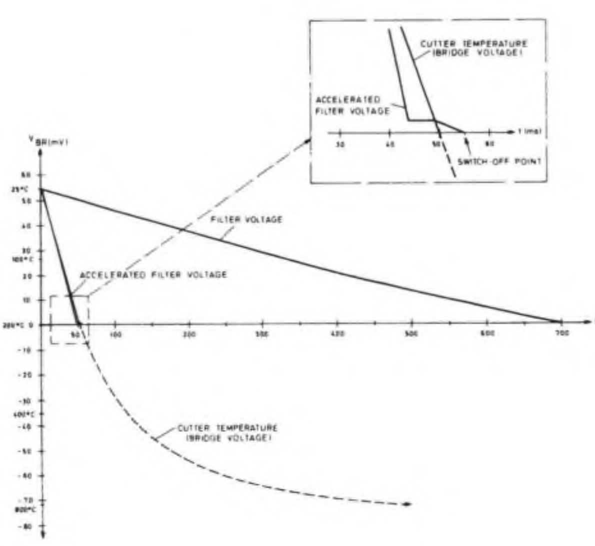

Fig. 3. Comparison of normal and accelerated temperature determination.

Frequency Band Division by the Subtraction Method

The ever increasing demands for quality in disk recording place very stringent requirements on the phase response of the system. The phase response is always then problematical when the frequency response range is to be divided into two portions, which will be treated separately and then reassembled. Problems of this sort have traditionally been solved using high- and low-pass filter networks. These, however, presented problems in the crossover area due to the different delay characteristics of the filters, resulting in poor compromise solutions. In the SAL 74 these problems have been solved through the acceleration limiter BSB 74. Division of the frequency range is achieved by first branching off the desired low-frequency portion using a low-pass filter and subtracting this from the total signal to obtain the upper frequency portion. To assure the required phase accuracy, the total signal is fed through a delay line equivalent to the phase delay of the low-pass filter before the subtracting process. After subtraction one obtains two partial response range signals which may be separately processed and then reassembled without producing any changes in the total signal due to phase delay differences of the two response portions.

The limiter operates as shown in Fig. 4. At the input to the limiter there is an amplifier 1 which serves to match impedance and level. A low-pass filter 2 follows with a crossover frequency of 4 kHz and a linear phase response in the frequency range of interest. Next follows an inverter 3 with an output of signal B representing the lower of the two frequency bands to be generated. The upper frequency band is now obtained by subtracting band B from the total signal. This has the advantage, when compared to the low-pass and high-pass method, that both portions may be reassembled without the difficulties caused by phase and level trimming. The lower frequency band can, however, only be subtracted from the higher if both signals are completely synchronous.

The total signal A is therefore fed through a low-pass filter 4, an amplifier 5, and a delay line 6 to assure that the total signal D has the same delay as the lower frequency band B. The low-pass filter serves to improve the delay line performance by suppressing all frequencies above the audio range, which also prevents their influencing the control functions of the limiter.

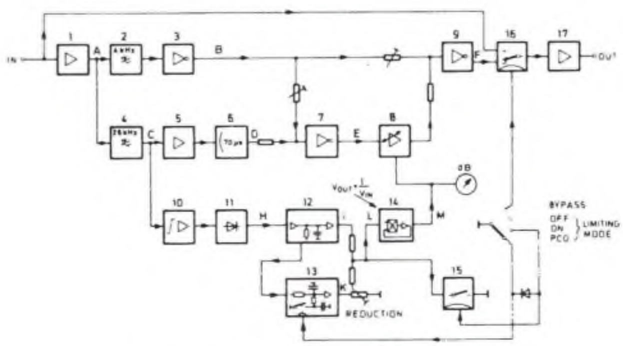

Fig. 4. Block diagram of acceleration limiter.

DC Controlled Level of the Upper Frequency Band

The addition of the inverted signal B to signal D in stage 7 produces the subtraction mentioned earlier and results in the upper frequency band E. This signal E is fed through an amplifier 8 whose gain may be varied by means of a dc control voltage, and is then recombined with signal B in stage 9, resulting in signal F, consisting of both the lower and upper frequency bands. This resulting signal F is devoid of any influence, throughout the control range, resulting from phase or crossover because of the principle applied here.

By comparison with other systems, this results in an improvement in subjective evaluation principally because of the fact that the level relationships in the higher frequencies are not disturbed (Fig. 5).

Limiter Operation

There are two ways to obtain the necessary control voltages. The signal voltage may be tapped off after the control amplifier and used for obtaining the control voltage. This backward regulating method usually applied in limiters has the disadvantage that the control function only starts after the signal has exceeded a predetermined threshold (Fig. 6). Up to the threshold point the signal is transmitted with its original steepness during the first half-wave, resulting in a dc component which can often lead to audible interference. This effect is usually heightened through the finite phase delays in the control loop. Such problems can only be avoided by using a forward-regulating system in which the branchoff of the control signal occurs ahead of the control amplifier, and which, in addition, makes sure that the control signal is fully effective before the signal passes the control amplifier. This is only possible through the use of a delay line which, in the case under discussion, already exists so that the advantages of a forward-regulating circuit can be fully realized.

To obtain the control voltage, signal C (Fig. 4) is obtained ahead of the delay line 6, but after the low-pass filter 4, so that suppressed signals lying outside the frequency range do not lead to limiting. First signal C is equalized in the amplifier stage 10 according to the required application. In this case a portion of the RIAA curve (75 μs) and the response of the cutter current (6 dB/octave) above 1 kHz are recreated. Following this is a so-called ideal rectification 11. The peak value of the signal H, which was rectified in circuit 11, is stored in circuit 12. At the output of circuit 12 a control voltage is obtained which serves to produce the full control range before the signal to be controlled reaches the gain control amplifier 8. It is, however, desirable that only a portion of the total control function become effective in the short time indicated to avoid amplifier overload, while the remaining control takes place according to psychoacoustic considerations. In the device described a time of approximately 10 ms was selected, which was achieved through the following circuitry.

The delayed component of the control voltage is obtained separately, and is available as signal K (Fig. 4) at the output amplifier 13. After the desired delay time of 10 ms it reaches the same value as signal I, which is divided via a voltage divider whose low side is connected to the output amplifier 13. As signal K increases, the voltage divider becomes less and less effective until K equals I and the full signal I is available at L.

A potentiometer between the output of stage 13 and 0 volts makes it possible to affect the degree of high-frequency limiting. A further switchable time-dependent processing of signal K in stage 13 permits so-called program controlled operation in which long-duration ($\geq$ 0.5 second) or frequently repeated control functions are recognized by an additional slow integration process, and the resulting voltage is superimposed on signal K, resulting in a slow leakoff of the high-frequency attenuation (approximately 30 seconds).

The control voltage L obtained as described cannot be fed to the gain control amplifier 8 directly, since it is a four-quadrant multiplier. It is therefore processed by a second four-quadrant multiplier 14 after which it is fed to the gain control amplifier 8.

CONCLUSION

The described components, the circuit breaker electronics and the acceleration limiter, represent essential electronics without which the advantages of the 600-watt per channel system which is now available could never have been brought to bear on the quality improvement essential in disk recording. The experiences collected over the past year, not only in disk cutting but also in control room application, indicate that the development of the BSB 74 produced a device with applications far beyond the disk cutting realm.

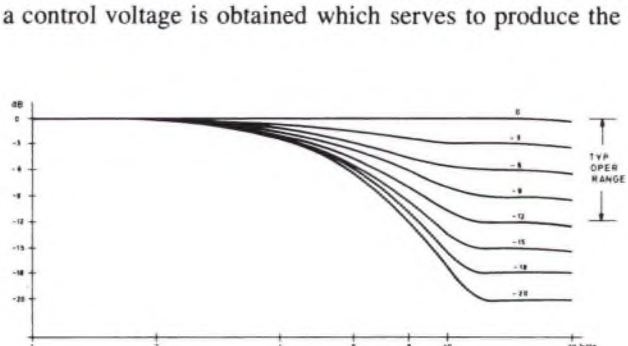

Fig. 5. Attenuation curves of acceleration limiter.

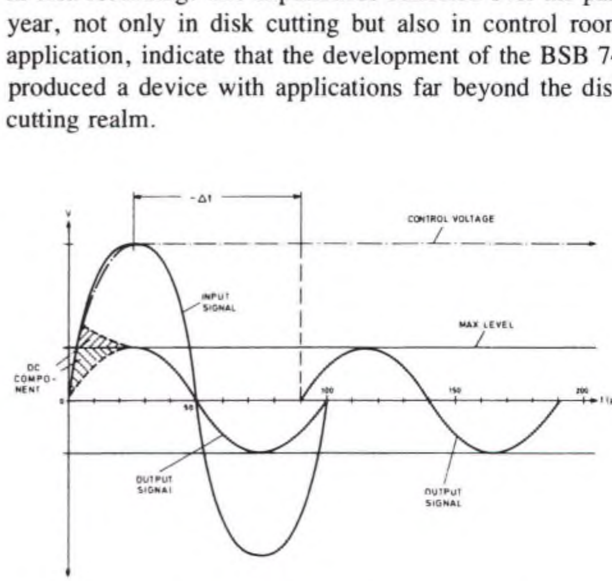

Fig. 6. Graphic presentation of limiter action for both forward and backward regulation.

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A Real-Time Digital Processor for Disk Mastering Lathe Control*

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The availability of highly reliable LSI circuitry has made possible a major breakthrough in disk recording lathe control technology. A new system for optimizing the groove-to-land ratio is described which utilizes real-time two-dimensional relationships derived from the audio waveform. This system, based on 12 years' experience with analog designs, achieves a significant advance in the time/level/skill relationships of the phonograph disk when compared to all previous cutting equipment.

0. INTRODUCTION

The idea of changing the distance between grooves as a function of the recorded signal is certainly not a new one. Columbia Gramophone Company of London obtained a patent for it in 1929 [1]. Eduard Rhein of Hamburg obtained a patent in 1942 [2], which made the control signal not only dependent on the modulation to be cut, but also on the modulation of the prior groove. Both patents exceeded the technical possibilities of their time. Add to that the time lost during World War II, and we are into the fifties before we again find patents on this subject [3]–[5]. Among these is one from Neumann [6] which describes a system that at the time was actually manufactured in quantity. The ensuing improvements within the framework of optimizing the groove-spacing control (variable pitch) were really based only on improvements in the available electrical components. In 1966 Neumann introduced a purely analog computer using discrete transistor technology [7]. The basic knowledge, using the most modern components and assembling these into a groove computer according to the latest knowledge in the field, was gathered in a paper by Braschoss and Kern delivered at the 1977 AES Convention in Paris [8]. We will show the realization of these ideas today—but let us first go back to the description of the problem.

1. THE PROBLEM

What is required is to optimally utilize the space available on a phonograph record. This means, in effect, to maintain the slowest lathe carriage travel at any given moment without thereby causing a collision between the grooves. Today's digital technology offers two completely different solutions to this problem.

2. POSSIBLE SOLUTIONS

One solution involves storing all the control signals needed for an entire record side in the memory of a computer, processing this information in a suitable man-

* Presented at the 60th Convention of the Audio Engineering Society, Los Angeles, 1978 May 2–5.

ner, and retrieving from the computer a control signal that will operate the lead screw of the lathe throughout the cutting process.

The other solution is the real-time analysis; namely, storing certain portions of the signal during the tape-to-disk transfer process for short periods of time, correlating these with other signals which are just being cut, and obtaining from these the lead screw controlling signal.

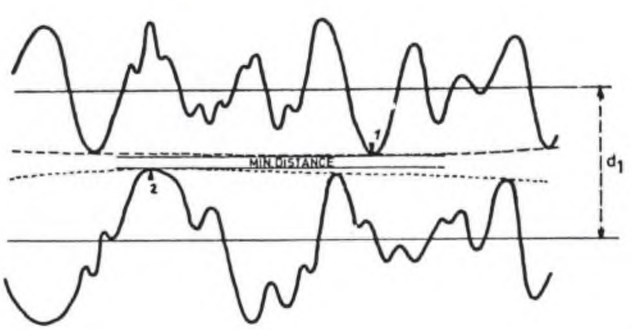
Both possibilities have advantages and disadvantages. The first solution would have the advantage of providing a readout of the exact space requirements, but it would need an additional processing step; the entering of the information. The real-time method, on the other hand, requires a sophisticated computer to permit obtaining optimum lead screw drive during the very short time that the signals are available.

We opted for the second method, for we wanted to avoid the additional, time-consuming, and therefore uneconomical processing step. It is still possible to prerun the program with the cutter up without spoiling a lacquer in cases where playing-time complications are anticipated. As indicated, the real-time analysis requires considerable computer sophistication, and one must utilize the signals carefully if optimum space utilization is to result. The problem is so complex that it cannot be solved using only a single-step approach.

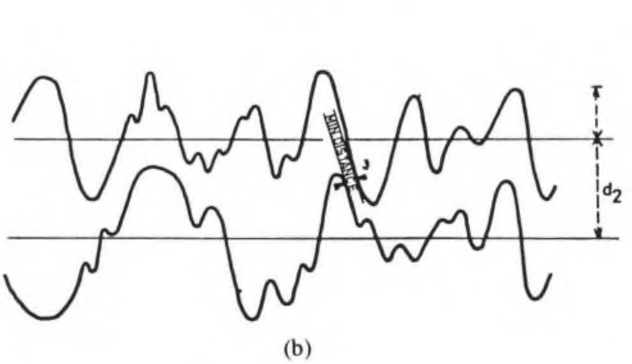
3. THE VMS 80 SOLUTION

In Fig. 1(a) two neighboring grooves are shown in a schematic fashion. Curves 1 and 2 show that the traditional pitch control, which uses the peak envelope of the signal curve, has worked correctly. Were it but possible to use the individual maximum values and their phase relationships to one another rather than the traditional peak envelope, an even closer groove-to-groove spacing would be possible. This is shown in Fig. 1(b). Today's IC components make this sort of control economically feasible. It is necessary to store the information of the left (inner groove) flank for exactly one full turntable revolution and compare it with that of the right (outer groove) flank (Fig. 2). The sampled memory, which in our unit is a digital delay, must be exactly synchronized with the angular velocity of the turntable. That is why one controls both memory and turntable speed, as well as all other control functions of the machine, from a central crystal time base. For suitable phase relationships of the signals such an arrangement permits the "snuggling" of two adjacent grooves up against each other, as the microscope picture of Fig. 3 shows. Natural sound phenomena will only seldom provide such advantageous relationships as those obtained here through the use of test signals. That is why we have searched for further possibilities of groove-space economy. The method which we developed provides for the optimum utilization of the space created for a particular modulation signal by the preceding amplitudes. This space, which the following signals can utilize, has been called the "rest space."

To better understand this rest-space utilization, it is useful to replace the continuous modulation signals with single impulses of varying magnitudes. Fig. 4 once again shows two adjacent grooves. The time between t_1 and t_1', between t_2 and t_2', and so on, is always half a turntable revolution. At the time t_1 the preview head alerts the lathe



(a)



(b)

Fig. 1. Schematic presentation of two neighboring grooves with pitch control. (a) Peak envelope method. (b) Phase recognition method.

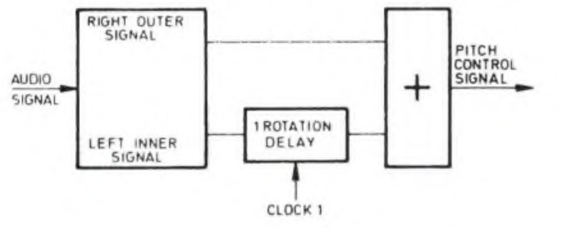


Fig. 2. Schematic block diagram of a phase recognition system according to Fig. 1. The inner flank signal is stored for exactly one full revolution and is compared to the signal portion of the outer flank of the groove then being cut.

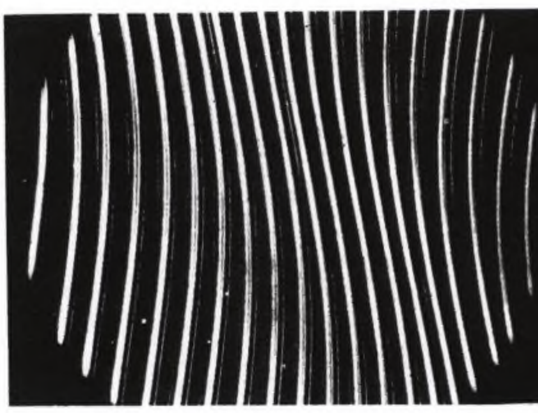


Fig. 3. Microscope groove picture of a 400-Hz signal. To show the accuracy of phase recognition, the land between grooves has been reduced to zero.

that a signal S_1 will have to be cut half a revolution later (as S_1'). The pitch control therefore has to create space for it just as the diagram shows. At t_2 there follows a signal S_2 (dotted line), smaller than S_1. This can be accommodated in the rest space following S_1 without additional pitch control. One therefore has to be sure that such signals do not produce further control signals from the computer. At time t_3 there follows a signal S_3 (dashed line), which is much larger than S_1. Here, too, the relationships can be readily seen. S_3 is substituted for S_1 as the pitch-determining signal, and the necessary space conforming to the groove shown as a dashed line is created at t_3. It is more difficult to visualize the last and far more frequently found signal relationship indicated by the dash–dot line. At time t_4 a signal S_4 occurs which, although higher than S_1, requires closer groove spacing than is needed to accommodate S_1, since it utilizes the rest space provided for S_1. However, were one to reduce pitch at t_4 corresponding to the S_4 information, then the line which would connect the foot of the S_4 signal with the foot of the S_4' signal would not leave sufficient space to accommodate S_1, and must therefore be stored until S_1 has been cut and appears in Fig. 4 as the dash–dot line.

Put in a different way, in order to utilize the rest space after cutting an initial signal, the ensuing signals must be sorted according to both magnitude *and* time relationship in order to obtain a suitable pitch-control signal. Fig. 5 shows the schematic of such an arrangement. The modulation for the transfer process comes from a tape playback machine which, aside from the usual stereo playback head which supplies the modulation, also has a second playback head, called the "preview head." It is mounted on the playback machine in such a way that the preview signals LI (for left channel, inner flank) and RO (for right channel, outer flank) are displaced by exactly half a revolution from the modulation signals. The signal identified as BP (basic pitch) is a direct voltage with which the basic pitch is set. The three signals RO, LI, and BP are added at summing point 1, and an intermediate pitch-control signal IS is obtained, which must be processed according to the various signal relationships, as shown in Fig. 4 in simplified form, to obtain the real control signal PS. In comparator 2, which is equipped with a gate, the IS is compared to the actual pitch-control signal PS. If IS is smaller than PS, the gate remains open. This circuit

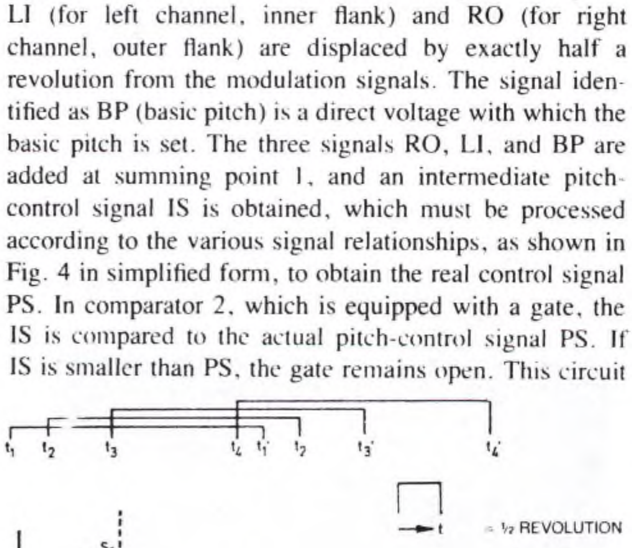


Fig. 4. Schematic presentation of two neighboring grooves with pitch control according to the rest-space utilization system. Dotted line—S_2; dashed line—S_3; dot–dash line—S_4.

prevents the smaller signals (like S_2 in Fig. 4) from influencing the pitch control. If IS is greater than PS, then it is fed via the subtraction stage 3, to be described later, into the sampled memory 5. It is here that the peak IS value, during 1/16 of a turntable revolution, is stored and held for exactly half a revolution of the turntable. The actual pitch-control signal PS is identical at any given moment with the largest temporarily stored IS signal. Referred to our impulse example in Fig. 4, this step ensures that control is always influenced by the highest instantaneous signal, here S_3. PS is furthermore fed to another sampled memory 4, which acts as an integrator and obtains the mean value of PS for half a turntable revolution, and thereby provides information about the rest space that has been created. This signal is fed to the subtracting stage 3 and provides an ever decreasing PS signal.

If one ignores the fact that we are dealing here with sampled or time-dependent processes, one can imagine that the signal fed back via integrator 4 is a feedback signal that constantly seeks to decrease the rest space.

One may ask why the seemingly constant basic pitch value BP was included in this complex control system. This becomes comprehensible when one views BP analogously to LI and RO; that is, as control signals which have the analogous task of supplying a portion of the pitch-control magnitude. In the subject circuit arrangement this is done by the computer, which means that for certain level sequences, when the rest space is to be reduced as fast as possible, the PS signal may actually become zero. This results in the carriage coming to a complete halt during the cutting process, even though a constant basic pitch was set up before the start of cutting.

Using the circuit arrangement shown in Fig. 5, the amplitude and time relationship of complex audio waves are utilized for the control of pitch. We have investigated only the horizontal component in this simplified presentation. For the lateral component of the vertical (depth) control, one basically requires a second, almost identical system. Another sampled integrator is needed for depth control. This stage, however, does not have any circuit for rest-space utilization, since depth control doesn't have any rest space in the sense in which it exists in pitch control.

By contrast to the phase recognition, the functioning of the rest-space utilization cannot be made readily visible by means of a test signal. It is typical for the microscope photos of actual musical signals to show maximum usage of available land without any improper groove kissing (Fig. 6).

4. CONTROL PANEL

Fig. 7 shows the operating panel of the machine. It has been divided into three basic sections. Starting from right to left one finds the programming section of the computer, the central indicating section, and the section in which all the operating elements have been grouped.

The main pointer instrument still shows the traditional lines per inch scale, although no longer as the main scale. The main scale is actually to be found right below and indicates in millimeters or fractional inches—the space consumed by the groove just cut. In other words, the physically proper linear value is obtained and not its reciprocal value of lines per inch.

It is a fact that the groove-to-groove spacing consists of two linear measures; the land width and the groove width. The latter is a fixed function of the groove depth. Both values may be set as basic values for the computer, even during cutting. These values are read, each on its own calibrated instrument associated with the respective control knob. The sum of both values corresponds at any given moment to the actual pitch which may be read on the main instrument. The lines per inch value obtained serves only as a comparison. The two further scales of the main instrument show the pitch during the TIME and FAST modes, here too as a linear measure. LEDs next to each scale show which scale should be read at any given moment. During the cutting process the LAND potentiometer assumes the function of the pitch knob of previous disk-cutting machines. The ability to adjust the land width without having to influence any other control signals represents a great simplification of the operation. The cutting engineer may wish to choose a very wide land for echo suppression, for example, only to return to very narrow land dimensions right afterwards without risking an overcut. He can go down to any dimensions which the LAND instrument still indicates as a finite land width.

The groove depth will likely not be touched during the cutting process. It will be determined strictly by the computer influenced by the modulation input signal.

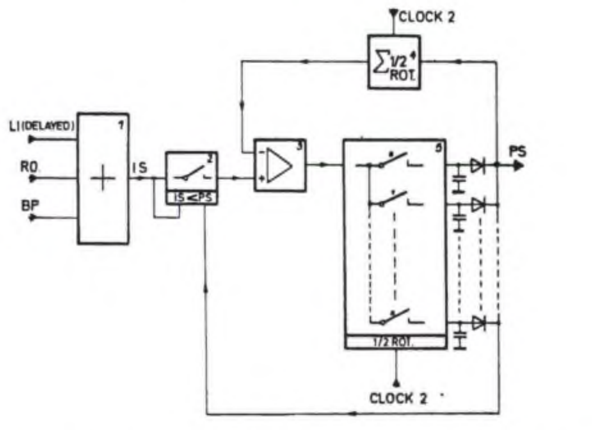


Fig. 5. Schematic block diagram for the realization of rest-space utilization according to Fig. 4. To make it easier to understand, only the lateral cut is presented. In practice, another similar circuit is needed for the lateral portion of the vertical modulation.

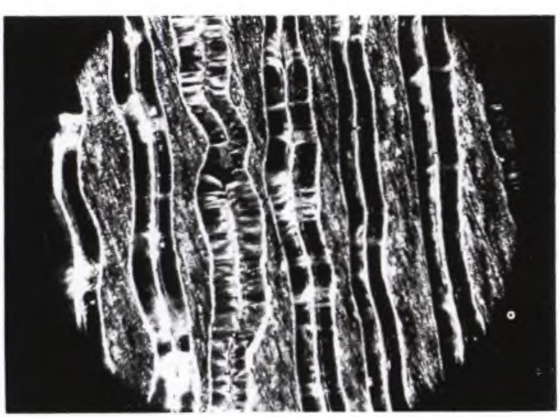


Fig. 6. Microscope groove picture of an actual sound modulation which has been cut using both phase recognition and rest-space utilization. It is typical for such pictures to show enormous packing density and the "snuggling" of adjacent grooves as far as the phase relationship permits.

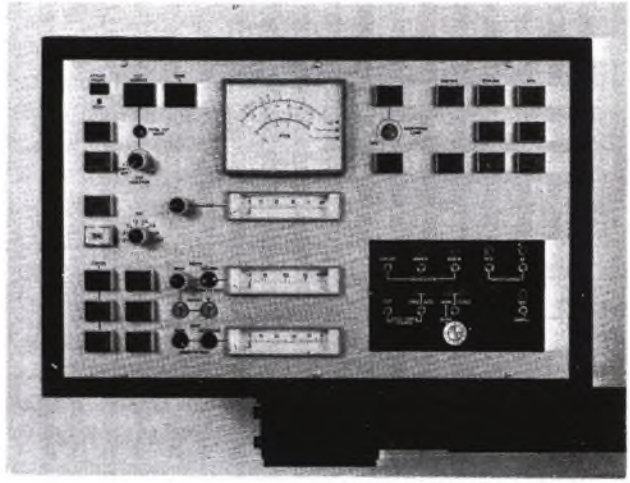


Fig. 7. Operating panel of the VMS 80 disk mastering lathe. The three meters in the center show the operation of the groove-space computer.

5. CONCLUSION

The system here described reflects the optimum analysis of the modulation to be cut on phonograph records and its conversion into signals for the control of variable pitch and depth. The extended research just completed shows a healthy respect for the methods used in the previous pitch and depth control system which has served the industry for the past twelve years. It is unlikely that with this new system the program lengths possible will be significantly greater than with the previous one, given a skilled operator. It will, however, no longer be necessary for the operator to do as much hands-on interfering with the process as he had to do until now to get the time on at the desired level. Only actual cutting experience in the field, using a wide variety of modulation, will provide the answer of just how big a step we have taken.

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styli and lacquer blanks

Practical Aspects of Hot Stylus*

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Some practical solutions to problems which arise in the use of the hot stylus technique in disk recording are presented. The author considers (1) the matching of stylus to cutter head, (2) the influence of burnishing facet size upon frequency response, distortion, and head wear, and (3) the design of feather edge styli. Procedures for setting the optimum working temperature, the elimination of resonance effects, and adjustment of cutting depth are given.

THE HOT STYLUS

The HOT STYLUS system to be discussed here is of the resistance type, and consists of #37 copper wire coiled around the gem of the stylus, and operated from a 2- to 4-volt dc supply. Approximately 6 in. of copper magnet wire were coiled on the body of the gem.

Common cutting styli were used in the initial experiments, with part of the resistance coiled on the shank of the stylus. The results obtained showed reasonable improvements in noise reduction, but too much heat was conducted to the cutter head by the aluminum shank. Exposure or increase in size of the gem solved this problem nicely, so that the resistance was coiled entirely on the jewel point.

The collaboration of a local stylus manufacturer made possible the development of (1) new types of stylus, with bigger jewel tips, (2) burnishing facets from zero to any size required, (3) special cements to strengthen the union of the jewel with the metal, and (4) the necessary degree of uniformity required between two specimens selected at random.

Unfortunately, the lack of uniformity was so great with styli obtained from reputable foreign manufacturers, that it forced the local stylus manufacturer to build a burnishing facet polishing machine which operated at the fantastic velocity of more than 250,000 rpm. This proved to be the only way to provide the very high degree of uniformity required between shipments.

Types of Stylus for Heated Cutting

With cutter heads of the plastic damping variety, such as Presto 1-D, RCA, etc., where the damping action of the blank material has an appreciable influence on frequency response, experiments performed with different types of stylus revealed the following sequence of results for a given type of master blank, which, incidentally, was Audio Master:

Capps' Anti-noise, peaked response at 6 kc with no substantial response above 7.5 kc.

Common master stylus provided a more uniform response with top limit in the 8-9 kc region.

Special types with smaller burnishing facets improved uniformly the top limit, up to the top limit of the cutter head, which in this case was a 1-D, about 11.5 kc to 12.5 kc was attained both for 78 and microgroove.

Feather edge type (zero burnishing facet) provided some measurable response at 15 kc, but the response over the spectrum, 2 to 10 kc, was full of imperfections, with peaks having the magnitude of 8 and 10 db, obviously caused by leakage of the damping action of the blank material.

This sequence of stylus types drove us to the following conclusion:

"With cutter heads employing plastic damping materials, the best compromise is to utilize a stylus with an extremely small burnishing facet, but this facet must always be greater than zero."

With cutter heads such as WE, Grampian, Orthofon, and other feedback types, where the damping action of acetate is not of primary importance, the stylus problem can be faced in a more precise manner. The experiments described below, aiming to find better stylus types, were made with a Grampian B1-AGU cutter head with the proper amount of feedback connected.

* Paper presented at the Eighth Annual Convention of the Audio Engineering Society, New York, September 26, 1956.

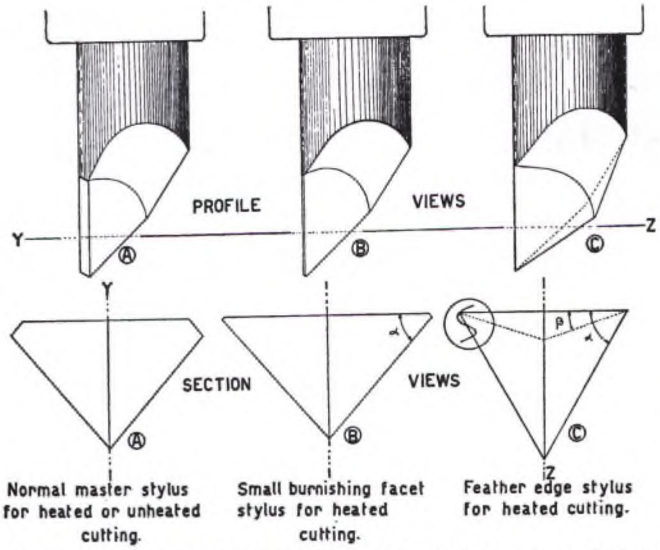

FIG. 1. Cutting styli shapes employed in the hot stylus technique. α is the angle included by the cutting facet and one auxiliary facet. The reduction of α to a smaller angle β results in a theoretically perfect feather edge stylus.

INFLUENCE OF BURNISHING FACET SIZE

Synthetic ruby was chosen for tip material, mainly because its hardness is greater than that of natural or synthetic sapphire.

In a stylus designed for cold cutting, the presence of a large burnishing facet is indispensable, but when the hot stylus procedure is utilized, the size of the burnishing facet can be reduced to zero, or almost zero, without materially affecting the noise level.

Normal styli have burnishing facets of 0.0003″ for 78 rpm normal groove. This approximate measure decreases to 0.00014″ for microgroove types. The profile of such styli can be seen in Fig. 1A.

A custom-built microgroove type was sent to us by some business friends from the United States. Measurements of the burnishing facets of these samples gave an average of 0.0001″. Test cuts revealed that even this small dimension was too large for 33⅓ rpm microgroove recording, although at 78 rpm microgroove this small facet type proved to be 100% satisfactory, within the range of audibility.

Incidentally, 78 rpm microgroove recording is a very fine way to check the top frequency of the cutter head, provided a reasonable stylus is utilized.

The figure, 0.1 mil, being still unsatisfactory, new types of stylus were made with burnishing facets as small as one-third that figure, in other words, down to 0.00003″.

With facets slightly smaller than 0.1 mil, say 0.08 or 0.07 mil, the upper frequency limit was extended, the increase being closely inversely proportional to the size of the facet; but, when the size of the facet reached a figure of 0.05 mil, curiously, this relationship between facet size and frequency ceased.

Close analysis revealed a very logical reason for this strange condition. Some element of the stylus, other than the burnishing facet, was now absorbing the vibrations of the highest frequencies. This proved to be the angle formed by one auxiliary facet and the cutting facet, as shown in Fig. 1B.

For the moment, let us call this angle α.

Some other effects had been observed before, that helped this analysis. One is the well known loss of the higher frequencies as the groove gets closer to the center of the record; the other is the limitation imposed on a 15-kc frequency at a medium-to-small diameter: that no matter how the amount of signal is increased, the recorded amplitude always remains approximately the same, and highly distorted.

New types of stylus were made with angle α somewhat reduced. This resulted in very noticeable improvements in frequency response, radius loss and the 15-kc acceptancy.

The angle α cannot be reduced below a certain minimum because for each degree that this angle is reduced, great amounts of material are removed from the tip of the stylus, thus increasing tremendously the fragility of the stylus tip, which now will break at the slightest effort, and render the stylus unsuitable for performing its functions.

A further small reduction of the burnishing facet is permissible after the readjustment of angle α, say, to 0.000035″ ± 0.000005″, our present figures.

THE FEATHER EDGE STYLUS

Parallel to the improvements made on the regular type of stylus, many experiments were made with another type of stylus, called "Feather Edge."

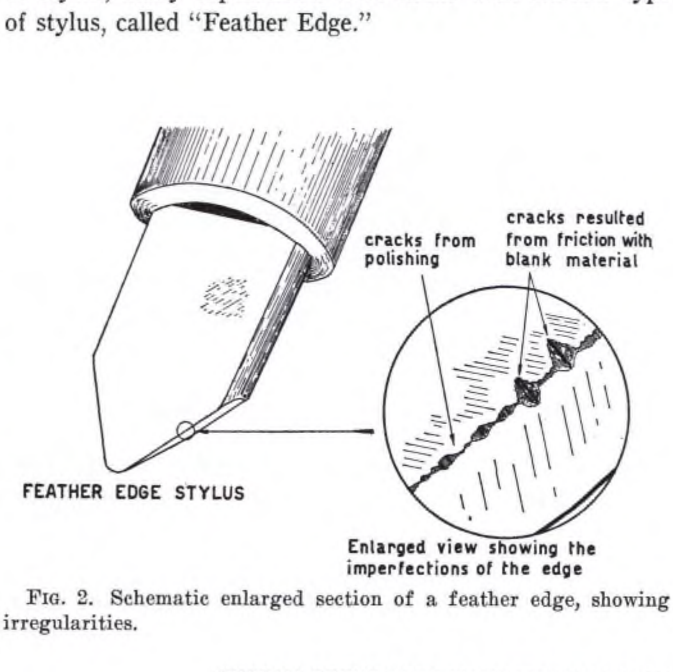

FIG. 2. Schematic enlarged section of a feather edge, showing irregularities.

This type of stylus should have theoretically a very sharp edge, but practice has shown that under high magnification, this sharp edge is not as regular as it should be. A similar situation can be observed on a razor edge at relatively low magnification. The irregularities of the edge produce noise by scratching the walls of the groove. They are due to imperfections in polishing, to scratches made by the diamond dust employed, and to cracks made by the lacquer material itself. Figure 2 illustrates these irregularities.

In order to minimize this deficiency, some modifications were made in the fundamental design of the stylus, such as an increase in the angle α (see Fig. 1C, profile view). Special polishing was also introduced. Unfortunately, increasing angle α brings back the problem discussed with burnishing facet types, that is, absorption of higher frequencies by angle α.

The section view in Fig. 1C illustrates graphically what actually happens when the sound has a wavelength shorter than the physical size of the stylus. To produce a perfect Feather Edge stylus, the angle α should be reduced to a fraction, β, of that amount, which would thus allow such a stylus to cut frequencies of 20 kc at a groove radius of 4 inches without loss.

But even if it were feasible to manufacture such a stylus, with an angle of magnitude β, this stylus would break into pieces if lowered into any other recording material than soft butter on a warm day.

DISTORTION CHARACTERISTICS FOR SPECIAL STYLUS TYPES

Distortion is inversely proportional to the frequency range reached, or, in other words, the higher the frequency engraved by the stylus, the lower the distortion.

ADJUSTMENT OF OPTIMUM OPERATING CONDITIONS FOR THE LOWEST NOISE LEVEL

This adjustment was made by the use of a 2-ohm, 50-watt rheostat, connected in series with the copper wire resistance, working with 4 volts dc supplied by a motor car battery (Fig. 3). A milliammeter connected in the circuit provided the necessary reference for the adjustment. A 0-100 ma full-scale type was used in our case. The instrument was shunted with nichrome wire until the maximum deflection with the full voltage applied corresponded to 100 ma. After that, some test cuts were made to determine exactly the best working temperature of the stylus.

Playback of these test cuts indicated that the best operating region is from 45 to 68 on this arbitrary scale for the 78-rpm, 0.003" groove, and from 42 to 56 for 33⅓, 78 rpm, and 45-rpm microgroove, and from 45 to 60 for

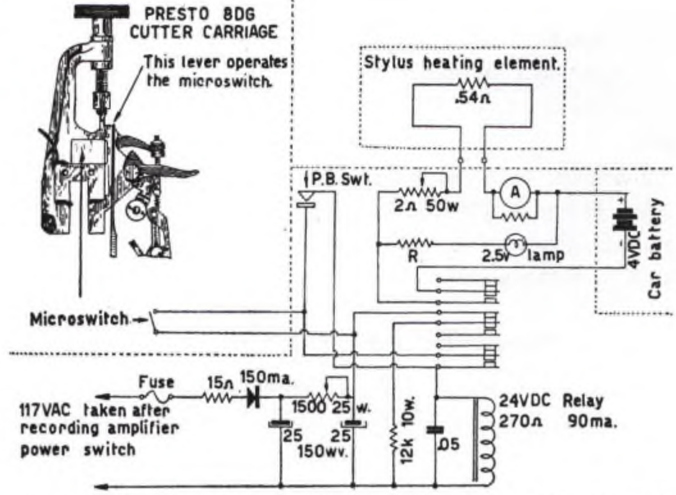

FIG. 3. Complete electrical diagram of the stylus heater, with the method used to mount a microswitch upon the carriage.

33⅓ rpm, 0.003" groove. Below these regions, noise increases gradually, and above, damage is done by the actual burning of groove walls, which also results in noise.

At even higher temperatures, breakage of the chip will occur, and the remaining part of it will accumulate around the stylus resistance, making a small fire that certainly will ruin the transcription, and the mood of the operator.

As the velocity of cutting diminishes with the diameter of the record, the temperature will vary a small amount from the outer to the inner groove, but the working region is sufficiently wide to accommodate this variation.

Room temperature also plays a small part, so the upper part of the working region should be avoided on warm days.

RESONANCE EFFECTS

Dilatation effects were also observed with this type of coiled resistance. They introduced noise in the recording, similar to that produced by a microphonic tube, although at barely perceptible levels. Total elimination of this effect was obtained by cementing the winding with inorganic cements. Zinc oxide cement, of the same type used by dentists proved effective for semi-permanent windings. A more permanent type of cement was obtained by using a combination of chemical substances in which the major ingredient was calcium sulphate.

DC vs. AC, THE GREAT CONTROVERSY

With direct current, there is a magnetic pull of the cantilever toward one of the permanent magnet poles, which although barely visible under a microscope, is enough to drive the cantilever out of its magnetic center, thus causing some nonlinear distortion similar to that produced by vacuum tubes when the bias is incorrectly adjusted.

Readjustment of the magnetic center with the operating voltage applied to the stylus provides compensation for this effect.

Obviously, with an alternating voltage, the magnetic center changes at the rate of the applied frequency, causing nonlinear distortion to both sides of the wave, or, in other words, intermodulation.

HUMAN FACTORS IN OPERATION

Since the day that heated styli emerged from the laboratory to become an equipment facility in our company, some problems arose which were due exclusively to human faults in operation. These most frequently were related to the turning off of the hot stylus switch. Although the actual stylus temperature is not excessively high, some damage can result to the magnets and to the plastic damping material (if used in the cutter) if they are exposed to the heat for considerable periods of time.

This problem was nicely solved by incorporating a microswitch into the cutter head carriage, so that a 24-volt dc relay is actuated each time the cutter head is lowered. Figure 3 shows how this is done. Double protection is obtained by connecting the ac power supply of the relay so that the power switch of the recording amplifier turns it on.

CONCLUDING REMARKS

As a heated stylus somewhat softens the lacquer material, it is advisable to adjust the depth of the cut after the stylus is turned on.

Control of the suction pressure utilized for chip removal is highly desirable.

The finished appearance of records made by following the above techniques is shiny and rainbow-like.

When styli of special type are employed, the appearance of dull stripes on the shiny surface need not cause alarm.

The hot stylus technique in its present development has enabled the engineer to further experiment with those design factors whose optimum values in the past were controlled by the signal-to-noise level barrier. They were designed to, at best, decrease the noise level, but at the expense of all the other factors so necessary for a faithful recording.

In my personal opinion, the hot stylus ranks with negative feedback in amplifiers as a great contribution to the advancement of the audio art.

Acknowledgments

The author wishes to thank here Mr. Sergio Lara Campos and Mr. J. Gomes, from RGE and from Realtone, for the many good suggestions offered, which made this paper possible. The author is also deeply grateful to Mr. Julius Postal for the presentation of this paper.

On the Damping of Phonograph Styli*

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The stylus mass and the groove compliance in phonograph reproduction give rise to a resonance at high frequency, which, even when compensated electrically, causes deleterious side effects. This resonance cannot be damped effectively by addition of series damping as it would produce excessive forces with high-velocity mid-frequency modulations. By subdividing the stylus-inductor structure into two parts connected by a semi-viscous coupler, a "dynamic damping" is obtained, which eliminates the resonance while at the same time diminishing the forces which the groove is called upon to furnish.

INTRODUCTION

ALMOST every worker in the field of phonograph reproduction is familiar with the nature of what is erroneously called "arm resonance," and which in reality is a low-frequency resonance of the arm mass and stylus compliance. The problems associated with arm resonance and methods of damping it without interfering with the proper functioning of the pickup are well understood.[1,2] At the other end of the frequency scale, however, another resonant condition is found, known by the equally erroneous name of "stylus resonance," which has not received the amount of attention it deserves. Stylus resonance in reality is a resonant condition involving the equivalent mass at the stylus tip and the compliance of the groove wall. This paper discusses the nature of this resonance and suggests an improved method of dealing with it.

STYLUS RESONANCE

The problem of stylus resonance may be placed in its practical perspective by examination of the response curves of typical "high fidelity" pickups. This is done simply, for example, by using CBS Laboratories' test record STR-120, which provides a continuous frequency-glide tone from 1,000 to 50,000 cps, synchronized with a General Radio type 1521-A recorder.[3] The response-frequency characteristic (left channel only) of five popular high-fidelity magnetic

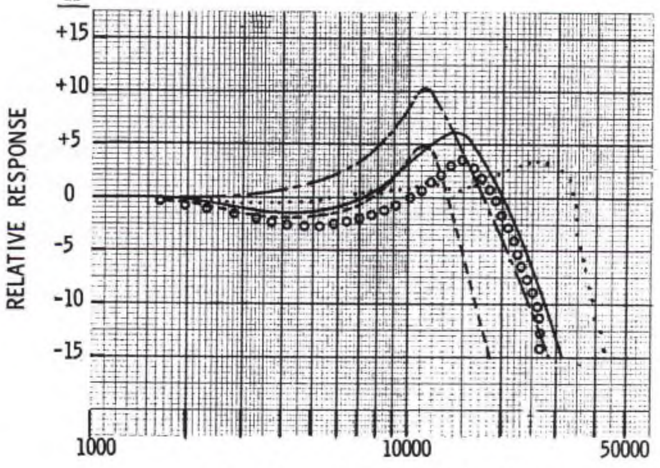

FIG. 1. Response-frequency characteristic of five popular "high fidelity" pickups, measured on an open-circuit basis (left channel only). CBS Laboratories' STR-120 record.

* Presented October 17, 1963 at the Fifteenth Annual Fall Convention of the Audio Engineering Society, New York.

1 William S. Bachman, *Proc. Inst. Radio Engrs.* **40**, 133 (1952).
2 Benjamin B. Bauer, *J. Audio Eng. Soc.* **11**, 207 (1963).

3 Benjamin B. Bauer, A. Schwartz and A. J. Gust, *J. Audio Eng. Soc.* **11**, 110 (1963).

pickups measured on an open circuit basis is shown in Fig. 1. These curves were run with a stylus bearing weight of 2½ g. It will be seen that four of the pickups exhibit resonant peaks of 8 to 10 db with resonance frequencies between 11,000 and 15,000 cps. It is not intended to imply that these curves portray an exhaustive statistic; however, our experience indicates that they are typical of what a customer might have obtained when he purchased a "high fidelity" pickup of the 1962-63 vintage.

From the resonance frequency and the groove compliance we can readily determine the stylus mass. The equations given in previous papers[4] will not be repeated here, but it can easily be verified from the resonance frequency data that the effective mass at the stylus tip of four of these pickups probably lies somewhere between 3 and 10 mg with the Q at resonance around 3. It is immediately apparent that these data do not resemble what usually is published in the commercial brochures or the high fidelity equipment reviews. This fact is simply explained: A resonant rise of 10 db, corresponding to a Q of 3, may readily be compensated by electrical means; for a typical magnetic pickup whose output is inductive this occurs naturally if it is connected across a resistance of the order of 30,000 ohm. This accounts for the usual frequency-response presentation which portrays a flat response to some 12-16 kc with a sharp cutoff thereafter.

As a practical solution of the problem, electrical compensation provides satisfactory sound reproduction to the hi-fi enthusiast. The engineer cannot rest with this approach, however, because the resonant condition is merely concealed from view while its deleterious effects remain. We will show, for example, how this resonance and ineffectual attempts to damp it may explain some of the record-wear reported by Anderson,[5] as will be shown below. Furthermore, in a recent publication, Hunt[6] offers a conjecture about a "carrier effect" which may arise from intermodulation between the random noise and high-frequency tracing distortion components and the shock-excited resonance frequency of the stylus to produce a lower frequency noise spectrum. Recognizing the difficulty of damping the stylus resonance, Hunt proposes to ameliorate this effect by diminishing the stylus mass so as to extend the stylus resonance beyond the cutoff frequency of the disc. This calls for a stylus with a mass which is at least an order of magnitude removed from present commercial practice. The response curve of the only pickup in our collection whose specifications appeared to be within even a factor of 2 or 3 from the "low-noise" conditions stipulated by Hunt is shown by the dotted line in Fig. 1. This excellent instrument, however, has a non-replaceable stylus and thus has not gained the popularity of the more rugged and convenient replaceable-stylus pickups using magnetic inductors. We now examine how the latter might be improved by suitable damping. However, the principles to be described are equally applicable to pickups using any transducer principle.

MOVING MAGNET PICKUP

As an example, the basic moving system of a moving-magnet pickup is shown in Fig. 2a. A magnet rod in cooperation with stationary pole-pieces and coils (not shown) is positioned in the cartridge by means of an elastomer ring which furnishes the stylus compliance C_s and the stylus damping R_s. A stylus holder, usually made of formed metal, is fastened rigidly to the magnet at one end and carries the stylus tip at the other end. The tip enters the groove for sensing its ondulations. It is sometimes assumed that because of this simple rigid structure the magnet follows the groove ondulations precisely, resulting in

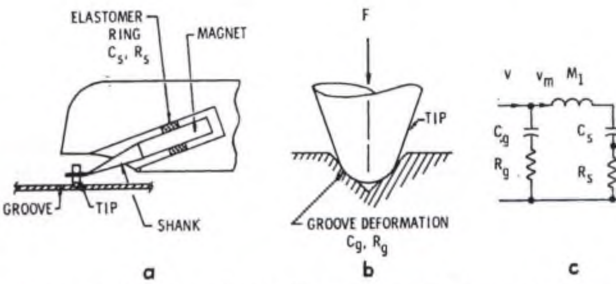

FIG. 2. *a.* Basic moving system of a moving-magnet pickup; *b.* Portrayal of groove deformation under the stylus bearing weight; *c.* Equivalent circuit of the moving system.

faithful reproduction of the recorded sound. This assertion, as was shown before, may not be realized in practice. The record groove being confronted by the small, hard stylus tip behaves more like a rubbery mass than as a rigid body; the tip actually deforms the groove wall, as illustrated in Fig. 2b, the deformation being elastic with low stylus forces but plastic with larger forces. The yielding of the groove cushions and diverts a part of the groove modulation that would normally find its way into the transducer (see the equivalent circuit shown in Fig. 2c): The modulation velocity v is diverted by the branch formed by the groove compliance and damping C_g and R_g, and then enters the branch portraying the mass of the magnet and stylus assembly, M_1, as well as the compliance and damping of the magnet suspension, C_s and R_s. C_s, of course, determines the "stylus compliance." In this equivalent circuit, we can accept the voltage generated across the damping resistance R_s as a measure of the relative magnet velocity, v_m. Also, since the compliance of the magnet suspension is a few orders of magnitude greater than the groove compliance, we can neglect it without significant error. We can also neglect the groove damping R_g which, as shown in Fig. 1, is rather ineffectual.

The equivalent circuit and computed performance characteristics of such a pickup are shown in Fig. 3. For illustrative purposes an equivalent transducer mass reflected at the stylus tip of 5 mg was chosen, which, together with a groove compliance of .048 μf (.048 $\times$ 10^{-6} cm/dyne), produces a resonant frequency at just above 10,000 cps. This mass is substantially greater than what is found in the better-class pickups, but from the data given by Hunt and the curves shown in Fig. 1 it does not appear to be atypical.

[4] Frank G. Miller, *ONR publication TM 20,* (March, 1950).
[5] Roger Anderson, *J. Audio Eng. Soc.* **9**, 111 (1961).
[6] F. V. Hunt, *J. Audio Eng. Soc.* **10**, 274 (1962).

In any event the results of this analysis can be translated into any desired values by simple transformation. The response curve of a grossly underdamped pickup is shown by the upper solid line; one might ask what damping resistance is needed to make this moving system become critically damped. This is answered by inserting into the circuit a series resistance, which for optimum damping turns out to be 350 ohm; in this manner the pickup can indeed be damped to flatness. Why this method has not been adopted by pickup designers is easy to see: With such a resistance, the force required to keep the pickup in the groove in the mid-frequency range, where stylus velocity peaks of 25 cm/sec are often encountered, would amount to 8750 dynes, corresponding to a stylus bearing weight of 9 g or, with a 50% safety factor, to 13-14 g. It should be noted that the force due to stylus-tip mass-reactance accelerated at 1,000 G (accepted as practical maximum) in this instance amounts only to 5 g. The practical solution is to be content with a damping resistance about ⅓-¼ the critical value (100 ohm), which requires but 3 to 4 g for tracking and, as was shown, produces a manageable peak which can be compensated electrically with a resistive termination at the pickup output. This sequence of events explains the performance of many present-day pickups.

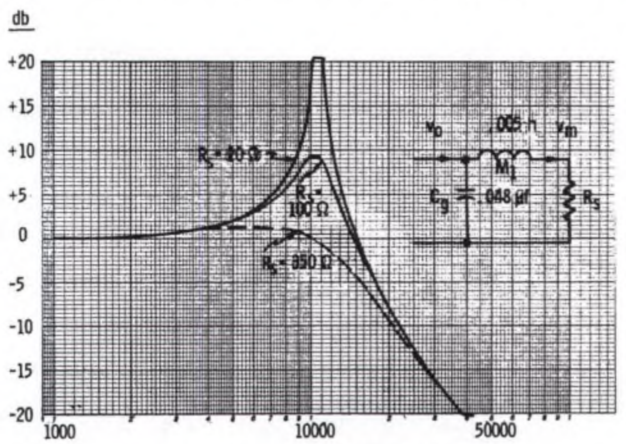

FIG. 3. Equivalent circuit and performance of the pickup with 20 ohm stylus damping, 350 ohm series stylus damping and 100 ohm damping.

DYNAMIC DAMPING OF STYLUS RESONANCE

In the paper on arm damping referred to previously[7] it was shown that the arm resonance could be damped without imposing additional forces on the groove by dividing the arm mass into two parts connected together by a damped spring. This damping method was called "dynamic damping." A similar method may be used to damp the stylus resonance with equal effectiveness, and the same name will be used for it. Dynamic damping of the stylus resonance can take on innumerable forms, but only one will be demonstrated here, by the way of illustration.

The moving magnet structure shown previously is now repeated in the arrangement shown in Fig. 4, except that this time the stylus shank, instead of being rigidly fixed to the magnet, is connected to it by a damped coupler defining a compliance C_a and a resistance R_a. The coupler can conveniently be of polyvinyl chloride, butyl rubber, or similar semi-viscous material. Examining the equivalent circuit, it will be noted that the mass element is now divided into two portions, M_s and M_m, the former being associated with the stylus and its shank, and the latter with the magnet. Also, we have added a second parallel mesh, R_a and C_a. Next, maintaining the same total mass as before, $M_s + M_m = .005$ g, we determine the optimum combinations of the two sub-masses and of the shunt elements C_a and R_a. This is done most conveniently by means of an analogy computer. The response curves obtained with three choices of mass distribution are also shown in Fig. 4. It will be seen that a favorable response is obtained with $M_s = .002$ g, $M_m = .003$ g and shunt element comprising a resistance of 125 ohm and a compliance of 1.2×10^{-6} cm/dyne. The compliance of the elastomer ring supporting the magnet in place can be neglected as long as it is chosen to be considerably larger than C_a, preferably by an order of magnitude; since it is not our intention to provide a specific structure design we exclude it from our computations.

The forces on the groove can be determined as a function of frequency by measuring the potential difference developed across the capacitor C_g. This is shown in Fig. 5. The force function with the conventional undamped stylus has the shape of a typical resonance curve with the maximum occurring at resonance, in this particular instance at just above 10,000 cps. Series damping diminishes the force at the peak, but at the same time introduces a force function at all the frequencies below 10,000 cps which, as shown before, can be quite considerable. Dynamic damping offers the lowest impedance in the operating range and thus may be expected to cause the least amount of record wear. Refering again to Anderson's paper,[8] an attempt can now be made to explain some of his results.

In his Fig. 11 Anderson shows the progressive effect of wear with a moving magnet pickup and also with an early ceramic pickup, both of which were played with stylus bearing weights of 6 and 9 g. At 6 g the moving magnet

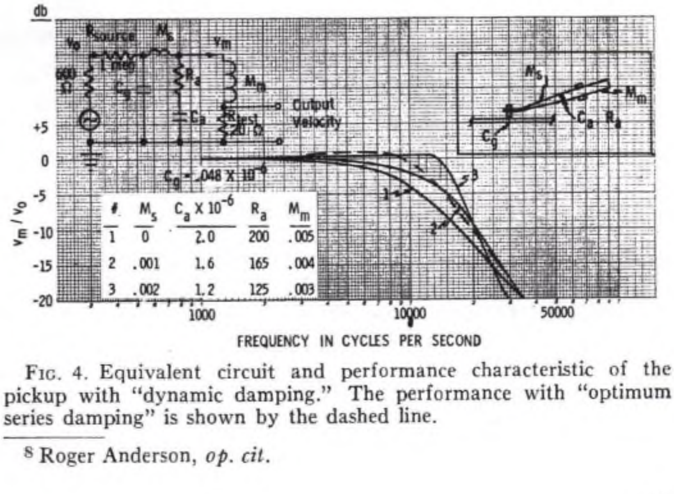

FIG. 4. Equivalent circuit and performance characteristic of the pickup with "dynamic damping." The performance with "optimum series damping" is shown by the dashed line.

[7] B. B. Bauer, *op. cit.*

[8] Roger Anderson, *op. cit.*

pickup caused an erasure of recorded modulation above 13,000 cps only, while the ceramic pickup played with the same force caused erasure of modulation at as low a frequency as 6,000 cps. It is a good guess that the ceramic pickup had a rather high stylus-tip mass and was provided with substantial damping between the stylus and the cartridge case, as was customary with the early ceramic pickups. This would create significant resistance to the groove modulations at all frequencies, causing a loss of precisely the type reported by Anderson. Interestingly, when the stylus bearing weight was increased to 9 g both pickups produced about equal modulation erasure, indicating that with 9 g

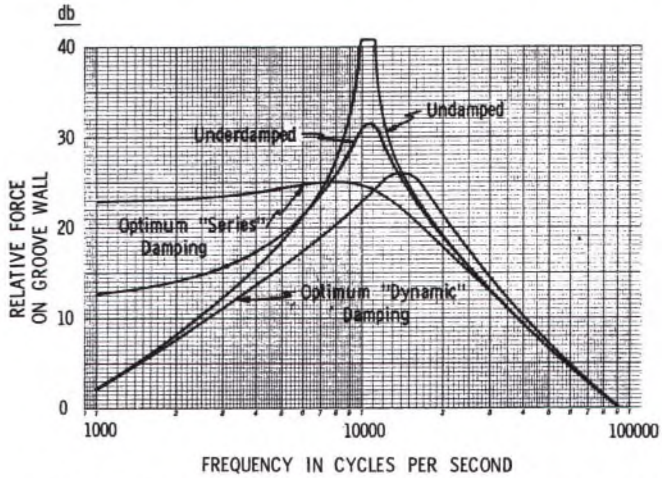

FIG. 5. Relative force on the groove for constant modulation velocity for an underdamped pickup, and also for a pickup with optimum series damping and optimum dynamic damping.

the stylus bearing weight, rather than the stylus tip impedance, constituted the predominant cause of wear. One can also hypothesize that wear could have been greatly diminished at 6 g with both pickups if dynamic stylus damping had been employed. It may be said, parenthetically, that distributed as well as lumped damping elements may be used, this approach being recognizable, for example, in the explorations of Walton.[9]

CONCLUSION

In this paper we have shown that the concept of equivalent circuit analysis and "dynamic damping" may be used to eliminate the stylus resonance and diminish the impedance offered by the stylus to the groove. This approach should be instrumental in improving future phonograph pickups. There has been a recent tendency to diminish the radius of the stylus and this, in turn, requires the use of lower stylus bearing weights; both of these factors tend to increase the effective groove compliance. To achieve an extended high frequency response with good transducer efficiency and rugged construction, the utilization of dynamic damping has a compelling attractiveness.

ACKNOWLEDGEMENT

The author wishes to acknowledge the assistance of Mr. Allen Rosenheck of CBS Laboratories who performed the analogy computations for the structures studied in this paper.

[9] John Walton, *J. Audio Eng. Soc.* **11**, 104 (1963).

THE AUTHOR

Benjamin B. Bauer graduated from Pratt Institute in 1932 and received the degree of electrical engineer at the University of Cincinnati in 1937. He joined Shure Brothers, Inc., manufacturers of electroacoustical devices in Chicago as a co-op student in 1936, became chief engineer in 1940, and vice president in 1950.

In 1957, Mr. Bauer joined the CBS Laboratories, and was appointed a vice president in 1958. He has charge of the audio, acoustics, and magnetics research in the laboratories.

Mr. Bauer is responsible for the development of a great many electroacoustical devices such as microphones, phonograph pickups, and tape recording heads. He has over 30 patented inventions in his name in the fields of audio, acoustics, and magnetic recording and has also published many papers covering a wide range of subjects in these fields.

He is a Fellow of the Institute of Electrical and Electronic Engineers, the Acoustical Society of America, and the Audio Engineering Society. He was an associate editor of the *Journal of the Acoustical Society of America* and in 1955 as co-founder and past national chairman as well as secretary of the IRE Professional Group on Audio he was recipient of the group's Achievement Award. Mr. Bauer is a member and recipient of the Recognition Award of the Eta Kappa Nu and a member of the Tau Beta Pi and Sigma Xi. He was awarded the John E. Potts Memorial Award of AES in 1963.

Design and Use of Recording Styli*

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The art and science involved in the design and use of recording styli, a new type of recording stylus, and the need for custom-made styli is described. The recording stylus is examined as the linkage between the cutterhead and record. Frequency response as a function of stylus geometry and cutterhead performance is discussed.

INTRODUCTION

SINCE 1877, when the first sound recording was made, the recording stylus has been the most used, and often misused, tool in the recording industry. With the advent of today's high standards of recording, the design of this tool has become a sophisticated science, and its manufacture requires the highest degree of craftsmanship. The demand for higher frequency response, better signal-to-noise ratio, longer playing time, and increased record durability has led the recording industry to the search for a better stylus. Because of the varying degree of emphasis placed on various demands by the individual recording engineer, a custom-made stylus is often required.

MATERIALS

Since Edison's time, various materials have been tried in efforts to produce the best possible stylus cutting edge. Over the years it has been found that corundum is the most stable, practical, and efficient material available for engraving sound waves on plastic-like materials. Corundum may be colorless, red, blue, green, yellow, or violet. Ruby and sapphire, the red and blue varieties respectively, are the most popular types. Synthetic and natural industrial sapphire is corundum (AL_2O_3) possessing a hexagonal crystalline structure and a hardness of 9 on the Moh scale and 1525 to 2000 on the Knoop scale.

There is a current illusion in some sections of the industry that ruby, or the red variety of corundum, which is used for lasers, is best for a recording stylus. This is not true. Rubies, sapphires, and all other varieties of corundum are made of the same basic raw material with the same crystal growing equipment and process detail. More than twenty-five years experience in the manufacturing of recording styli has shown that, in many instances, the colored corundum tends to mask minute irregularities in and under the polished surfaces and cutting edges. These are usually seen with clear sapphire. However, from the esthetic and sales point of view, ruby has charm.

Sapphire, because of its crystalline structure, lack of grain, and lack of cleavage planes can be ground to very acute angles and still maintain a fine edge.[1,2] This property is of the utmost importance in the manufacture of recording styli where very acute back angles are required (See Fig. 1).

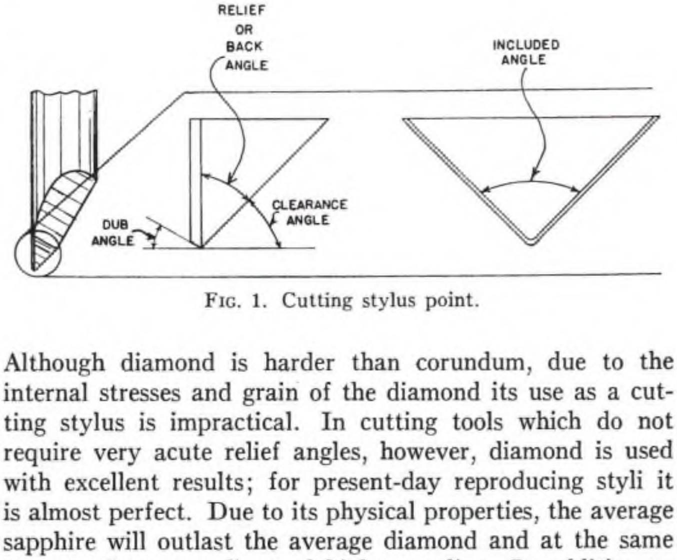

FIG. 1. Cutting stylus point.

Although diamond is harder than corundum, due to the internal stresses and grain of the diamond its use as a cutting stylus is impractical. In cutting tools which do not require very acute relief angles, however, diamond is used with excellent results; for present-day reproducing styli it is almost perfect. Due to its physical properties, the average sapphire will outlast the average diamond and at the same time produce recordings of higher quality. In addition to its physical limitations, the cost of producing a diamond cutting stylus would be much greater than that of sapphire.

STYLUS FINISH AND WEAR

There has been a great deal of discussion on surface finish of recording styli. The surface finish has been so vastly improved over the past twenty-five years that it cannot be

* Presented October 13, 1964 at the Sixteenth Annual Fall Convention of the Audio Engineering Society, New York.

[1] E. H. Kraus and C. B. Slawson, *Gems and Gem Materials* (McGraw-Hill Book Co., New York, 1941).

[2] *Properties and Uses of Linde Sapphire* (Linde Company, New York, 1958).

measured by present-day surface finish measuring devices. High-powered microscopes must be used exclusively. This indicates that the polish on the stylus measures less than one microinch. Any stylus which shows the slightest irregularity under microscopic inspection is immediately rejected before it is disc-tested for signal-to-noise ratio.

Where the wearing qualities of a stylus are concerned, edge instability is not a function of edge irregularities provided that the stylus produces a —57 db or better signal-to-noise ratio in an unmodulated groove, using 1 kc at 7 cm/sec as a zero reference. It is more likely to be related to some factors involved in the cutting process; namely, variations in lacquer composition which influence the ability of the stylus to clear itself of the lacquer material, the width of burnishing facet, the amount of heat applied to the stylus, the amount of suction used to remove the chip, and the direction in which the chip is removed. It has been found

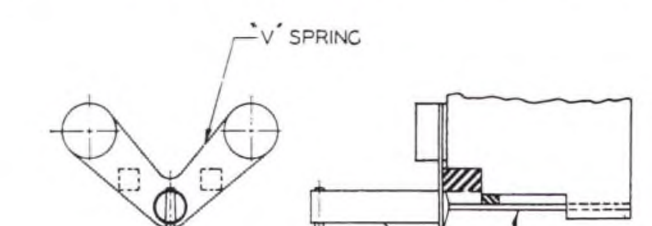

FIG. 2. Stylus suspension.

through many years of experience that over seventy per cent of the styli returned for resharpening either have aluminum or other foreign material on the tip, or are broken rather than worn.

STYLUS GEOMETRY AND GROOVE SHAPE

In the majority of records, the groove walls form an included angle of 90°. This is of the utmost importance in the case of the stereo record in the 45/45 system. Prior to the stereo record, most lateral records were cut with an 87° stylus on the assumption that after processing the groove grew to 90°. In order to maintain a 90° groove, one could, on the strength of experience with lacquer cutting problems, estimate the necessary angles on the cutting stylus. It was found that variations in the cutting and back angles of the stylus were necessary to compensate for any changes made in lacquer formulations. With the advent of the stereo record this problem was compounded many times, in spite of the availability of superior record blanks and more accurately produced styli. At the present time, the recording industry has adopted the stereo 45/45 system. By nature of the stylus' suspension in this system, as shown in Fig. 2, it must describe an arc in the vertical mode.[3] Due to this motion the cutting face of the stylus is tilted back off the vertical, as shown in Fig. 3. This tilting action results in an

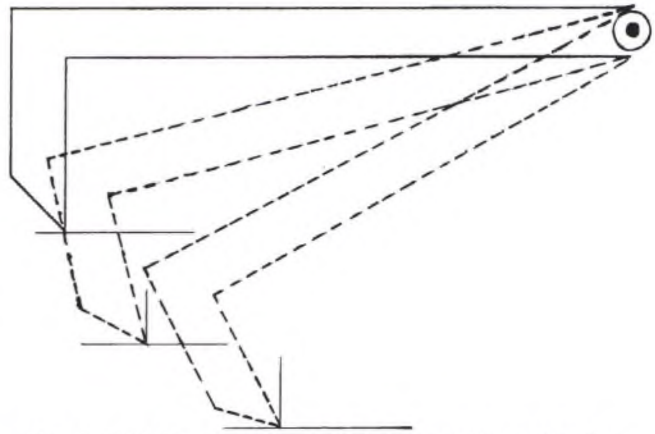
FIG. 3. Deviation from the vertical of the stylus cutting face.

increase in the included angle of the groove, as shown in Fig. 4. In view of this condition and the fact that a playback stylus rides the walls of the groove rather than the bottom, the waveshape must be distorted. This distortion increases with increased recording level. In the Westrex system, the vertical tilt angle is specified as 23°.[3] However, studies made by Woodward and Fox of RCA Laboratories[4] and Bauer of CBS Laboratories[5] have raised some controversy as to what the effective vertical tilt angle actually is in the record. This change in effective vertical tilt angle has been attributed primarily to a bending of the sapphire and lacquer memory or springback. If these were the only factors involved, the resulting effective vertical tilt angle would be more or less predictable. However, it has been found that variations of this angle are *not* predictable. Therefore, it seems reasonable to assume that other factors

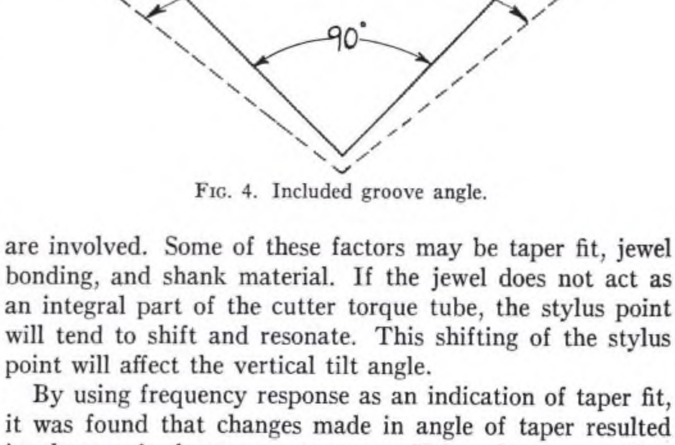

FIG. 4. Included groove angle.

are involved. Some of these factors may be taper fit, jewel bonding, and shank material. If the jewel does not act as an integral part of the cutter torque tube, the stylus point will tend to shift and resonate. This shifting of the stylus point will affect the vertical tilt angle.

By using frequency response as an indication of taper fit, it was found that changes made in angle of taper resulted in changes in frequency response. Using the cement that had been found to give the best bond, three different tapers

[3] C. S. Nelson and J. W. Stafford, "The Westrex StereoDisk System," *J. Audio Eng. Soc.* **12**, 3 (1964).

[4] J. G. Woodward and E. O. Fox, "A Study of Tracking Angle Errors in StereoDisk Recording," *IEEE Trans. on Audio* **AU-11**, 56 (1963).

[5] B. B. Bauer, "Vertical Tracking Improvements in Stereo Recording," *Audio Magazine,* 19 (Feb., 1963).

were tried, all mounted with the same jewel and inserted into the cutterhead with the same force. It was found that the response resulting from a standard production taper was essentially flat, the response from a taper with its bearing at the front end rolled off slightly at high frequencies, and the response from a taper with its bearing at the rear end peaked excessively at high frequencies, as shown in Fig. 5. From these results, it is obvious that variations

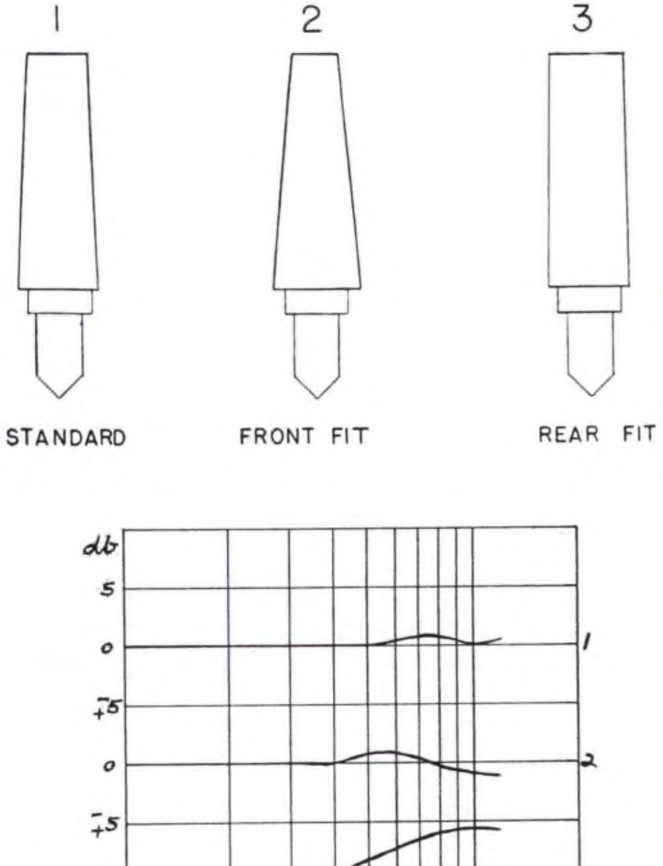

FIG. 5. Three different taper fits and corresponding frequency response curves.

in taper fit affect frequency response and may affect the vertical tilt angle.

Because of the soft nature of aluminum and the thin walls required by the dimensions of the taper, extreme care must be taken when inserting the stylus to avoid scuffing and distorting the tapered shank. Because of this it is desirable to use brass for shank material, as recommended by Westrex in 1957.[6] Proper mating of the stylus to cutterhead is dependent on the amount of wear in the tapered socket, uniformity of tapers, and the smoothness and cleanliness of both socket and shanks. In addition to the problem of

[6] Westrex Corporation, Print No. P77232, Issue 2, Revision, 1957.

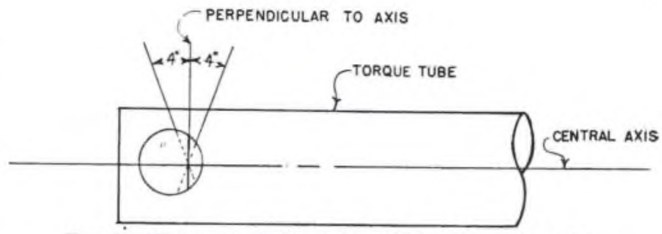

FIG. 6. Alignment of the stylus within the torque tube.

matching the taper to the cuterhead, there is the problem of alignment. It is a tedious and painstaking job to insert a tapered-shank stylus into a cutterhead and have the cutting face of the stylus fall within 4° of the perpendicular to the central axis of the torque tube, as shown in Fig. 6. For this reason, many styli which should be changed continue to be used, to the detriment of groove shape, processing, and quality of the finished record. A newly designed one-piece self-aligning stylus eliminates both the taper matching and alignment problems. A further improvement

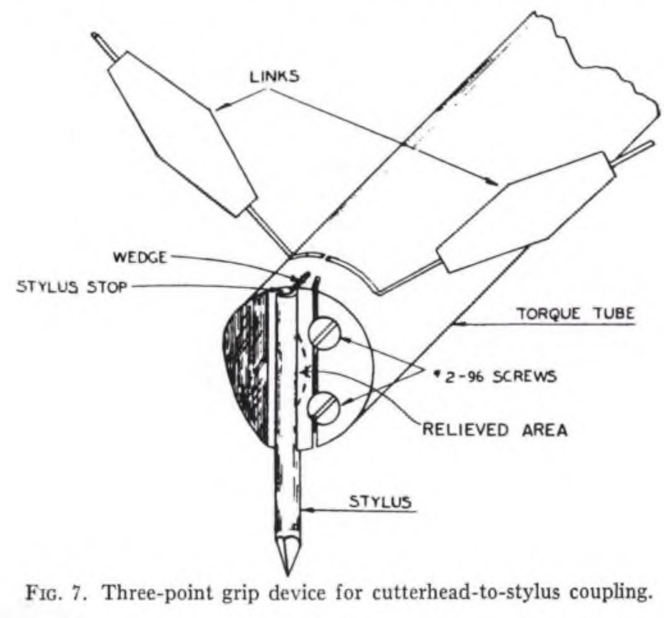

FIG. 7. Three-point grip device for cutterhead-to-stylus coupling.

in the cutterhead-to-stylus coupling could be realized through the use of a three-point holding device. This could be accomplished by relieving the center of the movable wedge at the front end of the 3D torque tube, as shown in Fig. 7.

A NEW STYLUS

In order to eliminate some of the errors caused by the vertical radial motion of the stylus previously mentioned, the new scooped stylus was created. The curved cutting face, or CAPPSCOOP† stylus, shown in Fig. 8, is a significant improvement over the present-day styli. By making the cutting face of the stylus a circular arc, it is possible to considerably reduce the deviation of the cutting face from the perpendicular to the record surface. In addition, the curved surface facilitates the removal of chip. This is evidenced by an average 3 db improvement in the signal-

† Patent applied for.

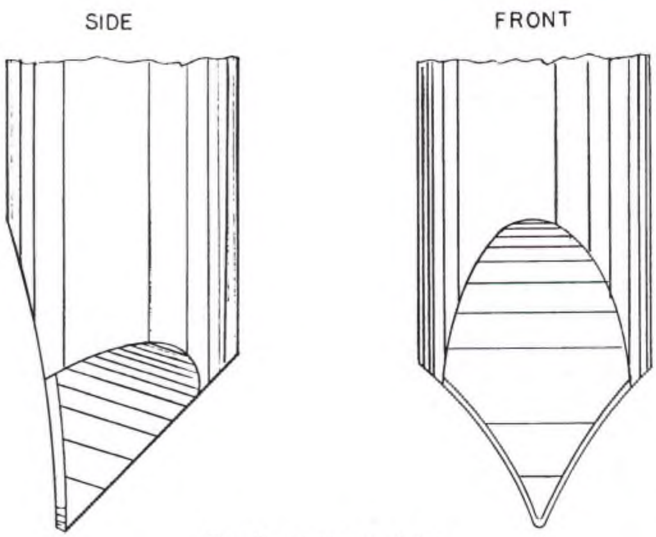

FIG. 8. CAPPSCOOP stylus.

to-noise ratio as compared with a conventional flat-faced stylus. This is achieved without loss of high-frequency response. On the other hand, there is definite improvement in the clarity of the sound, especially at the inner diameters.

A study of the groove walls produced by flat-faced styli has shown that they are not straight but are actually concave. This phenomenon is due to the cutting action of the stylus and the nature of the lacquer material. All one has to do is examine the chip removal from the disc, and it is quite evident that the cross-section of this thread is definitely not a true triangle. Of course, the chip is subjected to much more abuse and distortion than the remaining record material, but it gives a good indication of what is happening to the record groove. Because of its concave cutting edges, the CAPPSCOOP stylus compensates for the plastic memory of the disc material, thus producing straighter groove walls. This in turn results in longer stylus life, improved masters, better processing, easier molding, and improved playback tracking.

CONCLUSION

By its very nature, the recording stylus is the most reliable unit in the whole recording process. It is either up to specifications or it is not, and these specifications can be measured. However, due to weaknesses in the other components such as cutterhead response, differences in lacquer materials, oscillating amplifiers and the like, the stylus manufacturer is called upon to make any changes he can in order to correct these conditions. In some cases a design change will take care of the offending component; in others, it can at best minimize it so that the resultant masters are usable. These are the reasons for the use of custom-made styli. While it is true that many phases of recording stylus manufacture are automated, until such time as the perfect recording system is built, there will be a need for the custom-made stylus. There are no short cuts or mysteries involved in the manufacture of a recording stylus. It must be produced by exacting processes based on scientific theory in order to cope with the ever-changing problems encountered in the industry's search for the perfect recording.

ACKNOWLEDGEMENTS

The author wishes to express his thanks to the engineering and technical staff of Capps & Co., Inc., Mr. C. J. LeBel, and Mr. Donald Richter for their many helpful suggestions and discussions.

THE AUTHOR

Richard Marcucci was born in New York City in 1918. He received his education in the city school system, attending both Stuyvesant High School and the College of the City of New York. He has been with Capps & Co. since 1939 and was recently elected president of that company. Mr. Marcucci has been associated with many of the pioneers in the recording industry in both acoustic and stereo recording. He holds several patents on recording and reproducing styli.

Mr. Marcucci is a charter member of the Audio Engineering Society.

An Improved Disc for Master Recording*

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A new, improved master recording disc is being produced by advanced manufacturing facilities and techniques. The erratic quality heretofore common to lacquer discs has been eliminated. The new, flatter disc surfaces fulfill the ever-increasing demands of stereophonic and fine line cutting. Presented is a description of manufacturing improvements, quality control procedures, and comparative test data against outdated discs.

INTRODUCTION

In the late 1940's before the advent of the long-playing record, requirements for master lacquer recording discs were substantially less rigorous than at present. The discs were cut no finer than 98 lines per inch and fidelity requirements were such that flatness, cleanliness and signal-to-noise ratio could be met by leading manufacturers.

During this period, which was also before the advent of magnetic tape, the market for ordinary lacquer recording discs was larger and less demanding than that for the master discs. The availability of this substantial market gave the producers of lacquer discs the opportunity of selling material which did not meet master quality to a market where ordinary quality was adequate.

Long-playing records, variable pitch, and then stereophonic recording brought ever finer grooves and introduced demands for flatter disc surfaces, elimination of surface defects, less scoring under the advance ball, and freedom from noise and pops. At the same time the ordinary lacquer disc was being replaced by magnetic tape.

The manufacture of lacquer discs for mastering purposes was from the beginning a trying business for disc producers. Increasing quality demands caused several of them to abandon production. The few that remained were having difficulty in keeping ahead of the industry's demands for ever-improved quality. It should be pointed out, however, that this relatively small business of master disc manufacture is essential to the giant phonograph record industry.

The increasing quality demands, combined with other considerations, led Audio Devices to undertake the thorough modernization of facilities and the development of techniques capable of producing materials meeting present and future requirements. This paper is in the nature of a progress report on that program which was started in 1963.

THE NEW PROCESS

Many years of research had indicated that there was relatively little immediate advantage to be gained by changing the basic lacquer formulation. It appeared, however, that major progress could be achieved by new coating techniques leading to cleaner, flatter surfaces. The decision was therefore made to discard the French coating technique for which Audio had been exclusive licensee and to develop a program including new and superior coating techniques.

In the summer of 1963, new coating equipment was installed employing advanced coating technology. This was accompanied by a complete revamping of our manufacturing installation. Improved drying equipment, with rigid control of air flow and temperature, was installed. Manufacturing was designed to be a clean-room operation exceeding all requirements for "Design and Operation of Clean Rooms" in *Air Force Specification T.O. 00-25-203* (July 1, 1963).

In January 1964 we produced, in limited initial quantities, a superior master recording disc. Its surface was flat to the outside edge. The lacquer composition was practically defect-free and would not score under the advance ball. There was no groove tearing since lacquer would not adhere to the stylus. Groove noise measured at least 3 db down on the outside diameter. Samples distributed to a limited

* Presented October 13, 1964 at the Sixteenth Annual Fall Convention of the Audio Engineering Society, New York.

number of critical customers throughout the country were met with enthusiastic approval.

The major goal had been achieved—an improved disc could be produced. Unfortunately, the production yields of such high quality discs at this point were not sufficiently large to make the process economically practical, as is to be expected in any new manufacturing operation. The second phase, that of "de-bugging" and process improvement, has been a continuing program for many months.

To understand readily the problems that have been faced, the process in brief is as follows: The substrate, which is a high quality aluminum disc, receives a visual inspection for defects followed by a cleaning and washing operation before it can be used in the process. This specially cleaned disc is passed through the coating machine using a modified curtain coating technique, where the first face is coated. In curtain coating a carefully controlled film of lacquer is laid on the disc surface as it passes through the curtain. The disc is then dried by passage through a heated tunnel followed by a cure for a specified period of time. The process is then repeated on the second face followed by a controlled temperature cure. Finally, quality grading is done through 100 per cent visual inspection of both sides. The discs are customized as to number of holes, labels, and so on, and then packaged for distribution.

PROBLEMS AND PROGRESS

The final quality of the disc is dependent upon the extreme flatness of the aluminum substrate. Disc manufacturers must also be concerned with minute pits or pimples on the aluminum surface, which results in defects of equal degree in the lacquer coated surface, regardless of lacquer coating thickness. Finally, cosmetic effect must be considered. Since the lacquer coating is not completely opaque, certain cosmetic effects on the disc surface can be seen through the coating causing some unknowing recordists to reject the product as unsuitable. For example, the stretch leveling process to flatten the aluminum results in a grain structure which reflects as two flashes per revolution of the coated disc. In stamping the discs from flat aluminum there is a small unavoidable burr. This relatively sharp edge on one side is coated over during the process but it does cause problems in the initial washing of the material before it is coated.

The requirements for the aluminum disc are so rigid, and the market so limited, that only one manufacturer in the entire free world has produced the material. The price of this aluminum base has approximately doubled in the last ten years, while the price of master discs has gone up approximately 20 per cent. It is interesting to note that there is only one other important use for these aluminum discs; namely, double magnetic coated discs for random access use with computers. Here, six discs in a cassette sell for approximately $400. Even subtracting the cost of the cassette, this is many times the price for lacquer coated recording discs.

The aluminum disc as received is inspected then carefully washed in a detergent solution. It is dried under

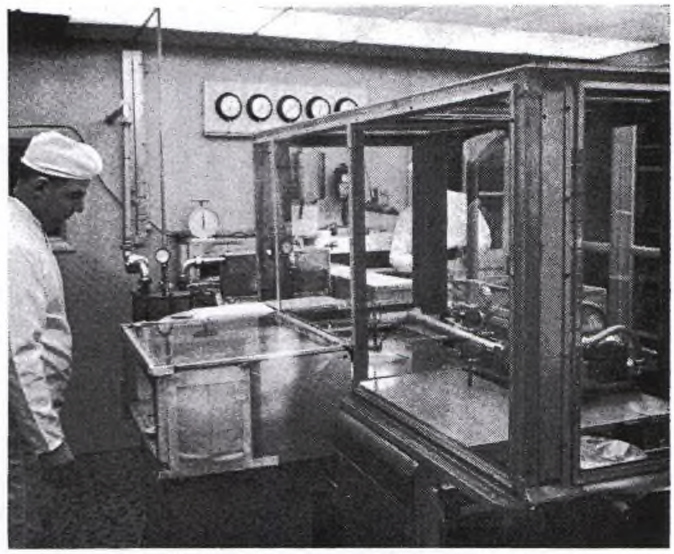

Fig. 1. Controlled conditions or white room cleanliness is essential in producing defect-free master discs.

carefully controlled conditions since all lint, dirt and grease must be removed before coating. The physical facilities are set up so that the washed disc emerges into a white room. Our program throughout the last several months has demonstrated that the cleanliness of the disc is an important factor and that correct selection and use of brushes and washing rolls will substantially affect quality and yields. After this process the disc is confined to the white room (see Fig. 1) for the remaining part of the process.

In order to meet the previously mentioned military specification for white room operation, we installed electronic dust precipitators to filter the incoming air supply for both the coating room and the drying tunnel. They are rated to remove 95 per cent of all particles three microns and larger in size. With the precipitators operating at peak efficiency and exceeding requirements of the military specification, cleanliness was still not sufficient to provide the defect-free master disc surfaces required. We found it necessary to back up each precipitator with an absolute filter rated to remove 99 per cent of all particles 0.3 micron or larger from the air. The point to emphasize here is that these filtered particles are well below the limit of visibility to the naked eye.

For those not familiar with the specifications for white rooms, these include a requirement that operators wear lint-free uniforms; in addition, the area is controlled in temperature and humidity and access to the room is forbidden to all but operating personnel. Periodically the cleanliness of the incoming air supplies is checked. The procedure followed by us is recommended by the Millipore Filter Corporation. Basically it consists of drawing a 10 ft^3 sample of air through an ultra-fine piece of filter paper by means of a vacuum pump over a 28-minute period. The filter paper is then examined under a microscope to determine the size and quantity of contaminants in the air. We have learned in the program during the last several months that absolute control over the environment is essential to the manufacturing process. Finally, the lacquer must also

FIG. 2. Master quality is maintained by actual recording and playback under studio conditions.

be subjected to the same processing care to achieve cleanliness. The lacquer is passed through a series of filters with the final filter being a polishing filter. Choice of proper filtration has required substantial development since the lacquer is a heavy viscous material.

Along with the program on cleanliness a corollary program to establish conditions for the coating operation was also progressing. The disc is coated by passage through a modified curtain of lacquer. We have found that the viscosity of the lacquer must be kept within 5 per cent of the selected coating viscosity, the temperature within 2° F, and the pressure of the lacquer entering the coating head absolutely constant. Furthermore, the speed of the disc passing through the curtain must be kept at an absolutely constant rate. The proper selection of operating points for all these variables was a time-consuming program. Now, under proper control, a disc of greatly improved flatness and controlled thickness is being produced.

The conditions of rate of air flow and temperature control throughout the drying tunnel constituted a similar program for many months to achieve optimum conditions. Too rapid drying can cause "orange peel," dimple effects and improper flow-out with loss of flatness. Too slow drying can result in too much flow-out with a magnification of aluminum base surface, as well as in a tacky disc which cannot be handled at the discharge end of the tunnel.

QUALITY CONTROL

Testing for master quality is done in a cubicle similar to a mastering cubicle in a recording studio, complete with a Scully lathe and a Westrex 3C Stereodisc Recorder. Playback equipment consists of a Grado Laboratory moving-coil cartridge mounted in an ESL-310 professional series arm and balanced for 3 g, a Model 7 Marantz stereo console, and an Instrument Electronics Model 247 Logarithmic AC amplifier voltmeter. A 200-cycle high-pass filter is employed before the meter to eliminate low-frequency equipment noise. This cubicle is adjacent to the production operation and performs continuous monitoring of each day's run. (See Fig. 2.)

Extreme care is taken in setting up to eliminate equipment error from test results. Cutter head balance is checked frequently. Stylus heat of .45 amp is used for both Capps and CBS prewired styli, since experience has shown that for our purposes no benefit is obtained from additional heat. The chip vacuum pipe is mounted behind and slightly to the right of the stylus. It was found that in this position the minimum vacuum is needed for efficient operation. Most testing is done with a pitch of 225 lines per inch and a 3-mil groove. As in making a disc master, a continuous cut is made across the surface. Occasionally, surface evaluation cuts of 400 lines per inch and 2-mils in width are made. This can be done as close as ¼-in. from the outside edge with no change in depth of cut. Playback zero reference is set with a 1 kc tone at a constant velocity of 5.5 cm/sec. Playback system noise against this reference reads —74 db through the filter.

As in the studio, the disc is visually inspected for defects before cutting. The cutter is dropped near the outside diameter and unmodulated grooves are cut to the 4-in. diameter. As the cut progresses across the surface observations are made as to uniformity of cut and scoring. The grooves of a cut disc are examined for tearing and appearance of the walls using the usual single-lens lathe microscope and also a 120-power stereo microscope.

The advance ball on the 3C recording head is a valuable additional tool in guiding the overall program by indicating the effectiveness of some of the changes in the process. Intermittent scoring can be an indication of dirt or surface contamination. Continuous scoring can be caused by other factors. For example, a disc stored for several hours at 130° F will give excessive scoring. In this case, a hard deposit adheres to the advance ball and plows through the lacquer. A loupe may be necessary to find this deposit which, until it is cleaned off the advance ball, will cause continuous scoring on all succeeding discs. Because of this condition, we recommend that lacquer discs not be stored in areas where the temperature exceeds 100° F.

TABLE I.

Diameter	Reading for new disc, db	Reading for old disc, db
12"	–68.5	–66.5
10"	–69.0	–66.0
8"	–69.5	–65.0
6"	–69.5	–66.5
5"	–70.0	–66.5
4"	–70.0	–68.5

Groove noise measurements are made at various diameters on each cut disc. Typical signal-to-noise hot stylus readings on our new disc as against our old disc are shown in Table I.

Higher noise readings on the outside diameters are caused by the increased velocity of the material by the stylus. The added noise level on the old disc as measured here is due to a more uneven surface, especially at mid-diameters, plus impurities in the lacquer. This uneveness appears as a rippled-like surface.

Some disc manufacturers have supplied discs with a surface bump or a once-per-revolution uneveness that levels out part-way in the disc. This condition introduces a swinging type of noise condition as shown in Table II.

TABLE II.

Diameter	Reading, db
12"	−66 to −58*
10"	−66 to −58*
8"	−66 to −60*
6"	−67.0
5"	−67.5
4"	−67.5

* Once per revolution swing.

We believe this to be caused by varying vertical pressure on the stylus as well as by a varying lateral rubbing action on the stylus cutting edges.

Since some recordists are still cutting with a cold stylus, we made groove noise measurements on the new disc with and without heat on the stylus (see Table III).

TABLE III.

Stylus diameter	With heat, db	Without heat, db
12"	−68.5	−67.0
10"	−69.0	−66.0
8"	−69.5	−63.5
6"	−69.5	−60.0
5"	−70.0	−57.0
4"	−70.0	−53.0

As can be seen, the hot stylus does help to some extent in keeping the groove noise down on outside diameters, especially when the surface is rough or uneven. The marked efficiency of hot stylus cutting is appreciated even more as the cutting diameter decreases.

It has been our experience, as well as that of others in the field, that stylus life has increased perhaps over 50 per cent with our improved commercial disc against our old disc. Data collected during our improvement program show that stylus life correlates directly with the degree of refinement in lacquer and air cleanliness.

CONCLUSION

The intent of this paper is to present a progress report on our new master disc program. We have learned that using the same lacquer formulation with refinements, a change in coating techniques, plus the addition of environmental controls far beyond what had been deemed necessary in the past, does result in a lacquer disc having improved flatness, cleanliness, lower noise and overall cutting superiority. We anticipate further progress as our program continues.

THE AUTHOR

John E. Jackson was born in New Brighton, England in 1923. He obtained his technical education at Lafayette College, Easton, Pennsylvania and RCA Institutes, Inc., New York City.

From 1950 to 1955 he was on the engineering staff of WPIX-TV, New York where he became master control technical director. From 1955 to 1960 he was employed on the technical operations staff of Columbia Records, New York. Following this he joined Audio Devices, Inc. at their Glenbrook, Connecticut plant. As manager of the Research and Engineering Evaluation Laboratory, he has been instrumental in the development of the new Audiodisc and numerous magnetic recording tape products.

Mr. Jackson is a member of the Audio Engineering Society and the Society of Motion Picture and Television Engineers.

Stylus Mass and Elliptical Points*

JOHN WALTON

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Elliptical points give higher measured distortion than conventional styli. An argument is put forward that this is largely due to a greater preponderance of stylus mass distortion over tracing distortion. It is claimed this theory can explain the occurrence of "modulation noise" on elliptical styli.

INTRODUCTION

THERE has recently been a trend for some of the better pickups to be fitted with elliptical styli in order to reduce the geometric distortion of tracing. However, because it has been reported that stylus mass distortion in a sense cancels tracing distortion,[1,6] it was thought necessary to assess the results of using such low stylus radii before fitting to one of our own pickups.

In order to try to find answers to the problem of relating stylus mass to stylus radius in more general terms, it was also decided to measure the effect of different stylus radii on several other pickups of *differing* stylus mass.

It will be objected that any conclusions for this latter more general case are devalued because other distortions such as those from vertical tracking error are included in the results. This is certainly so, even though these other distortions were generally only a small proportion of the total, especially where the total distortion rose away from its minimum in any particular case.

There does remain, however, an indication that the fitting of elliptical styli has, in practice, even in that minority of better pickups, not necessarily resulted in any significant reduction of distortion and can even result in its increase. A more unexpected result was the observation of certain conditions involved in the production of "modulation noise" from discs.

At the Audio Engineering Society Convention in October 1962 a paper by the present author, entitled "Stylus Mass and Distortion," showed that deformation of the recorded modulation by the stylus momentum produced nonlinearities in the upper frequency range of reproduction that were of as much importance there for the *majority* of pickups in use as those nonlinearities produced by the geometric errors of tracing. It was also postulated that the distortion due to stylus momentum could cancel this distortion at high frequencies due to tracing geometry.

These and other effects were independently analyzed and confirmed by Shiga,[1] whose paper seemed to have caused little written comment.

Shiga's paper developed expressions for these errors of reproduction which show that *a*. tracing distortion is considerably reduced by indentation of the groove due to stylus inertia, particularly at high frequencies; *b*. tracing distortion is reduced by the indentation of the groove due to tracking weight; *c*. both tracing distortion and indentation distortion are very slightly reduced at small diameters of the recorded disc due to curvature of the whole groove; and *d*. tracing distortion is modified at the upper frequency end of the range by stylus groove resonance.

Shiga's approximate expression for second harmonic distortion takes up 3 or 4 lines and his expression for third harmonic distortion is even longer.

But if we merely extract the factor for stylus mass distortion from the second harmonic expression, i.e.,

$$D_2 = \frac{a^2}{18W^2}\left(1 - \frac{r}{6}d\frac{W^2}{V_{gr}^2}\right)^2 (j\omega Z_m)^2 d \qquad (1)$$

where D_2 = second harmonic distortion, a = amplitude, W = tracking weight, r = stylus radius, d = indent depth, $w = 2\pi \times$ frequency, V_{gr} = groove speed, Z_m = impedance of arm and cartridge at fundamental frequency, V = recorded velocity, and if we find that the first term is in practice very nearly unity, and we merely use the Hertz equation[6] to express d in terms of r and W, then the expression will reduce to

$$D_2\% = -k(V\cdot f^2\cdot m^2)W^{-4/3}r^{-1/3} \qquad (2)$$

where k is a combined constant to include those in the Hertz equation.

Shiga recognizes the difficulty with the factor d, caused by the inadequacy of the Hertz equation for the gramophone stylus with its plastic range and moving indentation. Nevertheless, using his expressions, Shiga shows that about 1% second harmonic is produced by stylus inertia at 800 cps with a 5 mg stylus at 10 cm/sec peak. This would mean 25% for a 2.5 mg stylus at 8 kc. which seems to fit the facts.[6]

When these problems are combined with the variations in response and impedance resonance, as well as tracking angles, etc., in practical pickups, it is realized that any

* Presented October 14, 1965 at the Seventeenth Annual Fall Convention of the Audio Engineering Society, New York.

theoretical analysis of practical pickup behavior is not only very complicated but so far merely qualitative.

Unfortunately the complication is not obviously surmounted for the pickup designer by the process of resorting to measurement because opinions differ on the relation of these methods to audible unpleasantness, etc. (Even then, analysis of the measured distortions in relation to what *factor* one is trying to measure still leaves one with most of the *mathematical* complication.)

Again, the procedure of relying on even the best organized listening test is not good enough since all recorded conditions are not encompassed on any one agreed recording. Even if they were it is doubtful that general agreement could be reached on what are now finer points of assessment in this and other current papers on distortion (even with correctly set-up modern reproducers) since the distortions in the loudspeakers, etc., are sufficient to mask the differences.

The pickup designer cannot, however, ignore such questions as, for instance, those relating to the effect of using an elliptical stylus, and is forced to pronounce judgment to the best of his ability.

With such reservations in mind the present author believes that the measurement of harmonic amplitudes on the panoramic display analyzer is well suited to analysis of the effects of stylus radius and stylus mass as at present understood, because:

1. Within the concept that distortion is the result of amplitude modulation, the measurement of harmonic amplitudes has a basically similar sensitivity to and bears a relation to intermodulation methods: the approximate tracing distortion expression for second harmonic amplitude is given[3] by $\pi r f/V_{gr}^2$, i.e., at 400 cps

$$D_2 = 400\pi Vr/V_{gr}^2 \tag{3}$$

and the intermodulation product is given[4] by

$$I = 800\pi Vr/V_{gr}^2 \tag{4}$$

where r = stylus radius, f = frequency, V_{gr} = groove speed and V = modulation velocity.

Equation (4) applies to the 400-4000 cps SMPTE method whose widely spaced frequencies make measurement at high frequencies less possible, and 400 cps is the main modulator. Thus, the fact that it is the intermodulation produced which may cause the audible unpleasantness is irrelevant if both methods are proportionally related to the same source and effects.

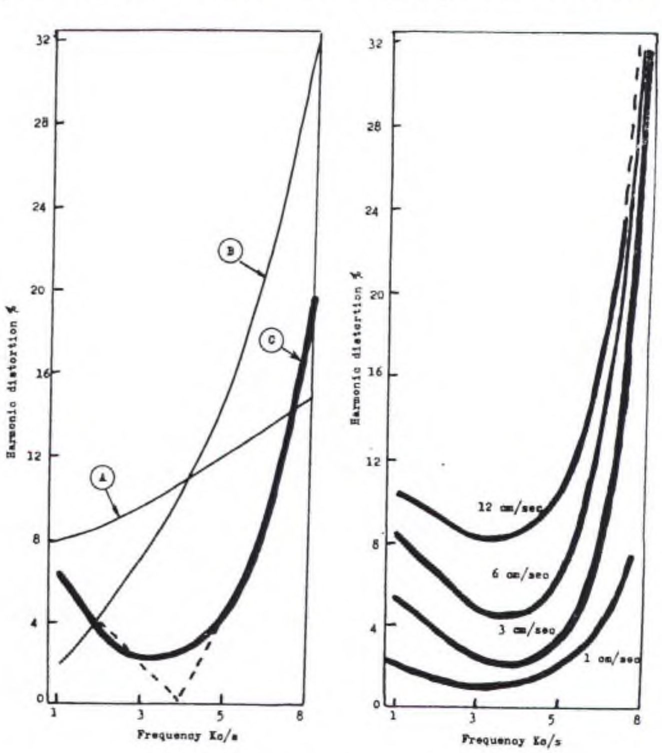
The conceptions of distortion as phase modulation[5] do not seem to invalidate the above principle for a single tone, and phase modulation in any case is an integral feature of all loudspeakers with moving cones. But if, as would appear, phase modulation is an essential feature of tracing distortion, then the corresponding theoretical figure for the intermodulation product may be larger or smaller than that given by the simple amplitude modulation theory used in Eq. (3).

For small values of distortion all methods produce answers that reduce toward zero as linearity improves, i.e., there is only a single pair of sidebands of appreciable amplitude if the phase change is less than ½ radian. This is so for most good pickups.

It would seem, therefore, that if the reproduced waveform of an appropriate range of single tones is improved or impaired, then the harmonic, intermodulation and phase modulation products will be affected accordingly. A suitable measure of linearity will indicate all three aspects therefore.

2. If one uses the SMPTE method for measuring intermodulation products it is not obvious without mathematical analysis how much of the products is due to the low frequency and how much to the high frequency. (Considering amplitude modulation only it would appear that nonlinearities due to effects at the high frequency would only be separately indicated if one measured in the region of the harmonics of the higher frequency. Considering phase modulation this is no longer necessarily true, and effects due to both frequencies can be more severely implicated in the 4000 cps sidebands. We are then concerned with the limitation of the ½ radian phase change, and *also* with the relative importance of the effects of the two frequencies, which brings us back to the starting point.)

The CCIF method of measuring the intermodulation products of closely spaced frequencies would overcome the above ambiguities, and by measuring only the difference products, it could allow of measurement to the highest frequencies. (This method has incidentally been described as



a. *b.*

FIG. 1. *a.* "Theoretical" distortion of a stylus with a mass of 3 units. *b.* Measured distortion of a 3 mg stylus. *Curve A*: tracing distortion, corrected for scanning loss. *Curve B*: stylus mass distortion. *Curve C* = *Curve A* — *Curve B*: resultant total distortion. (*Dashed line* shows deduced curve, *heavy line* the curve that would be plotted from measurements at the denoted frequencies.) All curves corrected for RIAA characteristic. Record diameter 5 in., record velocity 12 cm/sec peak, replay stylus radius .0006 in.

less sensitive,[2] and this would appear to be relevant within the above limits of phase change.)

To overcome the difficulties described in relation to the SMPTE method, and until suitable records are available for the CCIF method it was decided to measure harmonic amplitudes, considering their relevance to be vindicated.

3. The harmonic amplitude method using a panoramic display analyzer gives both an immediate analytical clue, by recognition of which harmonics are produced and in what relative proportions, and the quick reading of hundreds of combinations of level and frequency that are not seriously hampered by, for instance, fluctuations in turntable speed. The measurements include the anomalies of the cutting process itself, as in the case of the pickup user. It seems that the CCIF method may not necessarily do this.

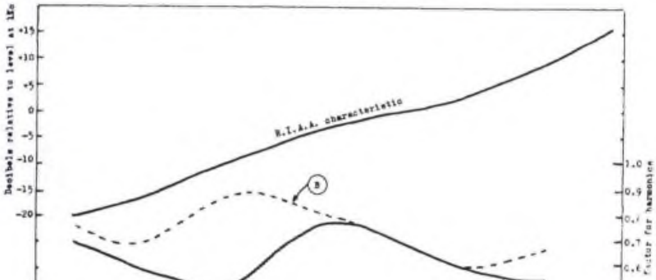


FIG. 2. Harmonic relations with the RIAA replay characteristic. *Curve A*: relation of second harmonic to fundamental. *Curve B*: relation of third harmonic to second harmonic.

The following precautions must, of course, be taken (as in fact with other methods): *a*. Other conceivable variables, such as vertical and lateral tracking angles, temperatures, rumble, and frequency response over harmonic range must be properly controlled; *b*. All harmonics to be measured must fall within the normal frequency range of the pickup; and *c*. A wide range of frequencies and levels must be covered before any conclusions are drawn.

The above conditions were observed in the measurements presented in this paper.

RELATIVE EFFECTS OF STYLUS MASS AND STYLUS RADIUS

In order to know what variables to use in any analytical measurement, the pickup designer must have *some* theory, however inadequate.

In an earlier paper[6] the present author presented some electron micrographic evidence of the very significant extent of the distortion due to stylus inertia, but the expression given for stylus mass distortion, i.e.,

$$z_m\% \propto (m^{2/3} f^{5/3} V^{-1/3}) r^{-1/3} \qquad (5)$$

was only based on maximum instantaneous values of velocity and acceleration.

Shiga's more thorough analysis[1] gives a similar sort of prediction and it is clear that stylus mass distortion will vary more severely with frequency than does tracing distortion. It also seems that smaller stylus radii will slightly increase stylus mass distortion. It would follow that whereas the second harmonic amplitude of tracing distortion at constant groove velocity, corrected for scanning loss, varies very approximately as $1/\lambda$, stylus mass distortion (total harmonic) should vary approximately as $1/\lambda^{5/3}$, where $\lambda =$ wavelength. It was also shown how the second harmonics of the two distortions can be in opposite phase to one another. Let us proceed on this loose assumption.

In the absence of fuller knowledge, an arbitrary starting point was taken where $z_m = 1\%$ for a stylus of unit mass when $f = 1$ kc, and several curves then constructed on this *same* basis according to the above laws.

In Fig. 1a two curves are thus constructed; one for a stylus of 3 units of mass, the other for the tracing distortion of a .0006 in. radius stylus (both after correction for RIAA reproducing characteristic, shown in Fig. 2) and summated to produce a third curve. The shape of this curve approximately coincides with the results presented in the earlier paper for the 3 mg stylus (Fig. 1b). If the factor of stylus mass in Fig. 1 is divided by 3 (i.e., $z\% \times 1/3^{2/3}$), and the curves redrawn, it will be seen (Fig 3a) that the resultant curve shape again approximates the results of experiments for the earlier paper, this time for a 1 mg (Fig. 3b).

While the above procedure was little more than juggling to align the expressions with the facts, it did produce an amazing agreement between "theoretical" and measured results in outline principle. It was therefore decided to see if, keeping the same basic units as above, it would be possible to predict the measured results of using elliptical styli in current pickups with lower tip masses.

Until a more appropriate and more rigorous approach is made, the presenting of these results seems justified.

In the earlier paper it was also shown that the depth of

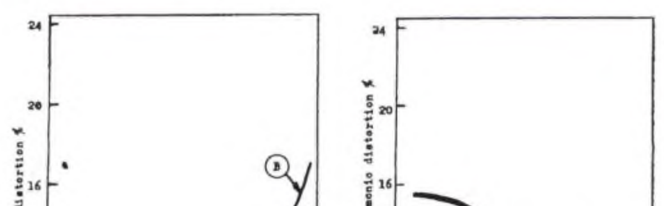


FIG. 3. *a*. "Theoretical" distortion of a stylus with unit mass. *b*. Measured distortion of a 1 mg stylus. *Curve A*: tracing distortion corrected for scanning loss. *Curve B*: stylus mass distortion. *Curve C* = *Curve A* — *Curve B*: resultant total distortion. (*Dashed line* shows deduced curve, *heavy line* the curve that would be plotted from measurements at the denoted frequencies.) All curves corrected for RIAA characteristic. Record diameter 5 in., record velocity 12 cm/sec peak, replay stylus radius .0006 in.

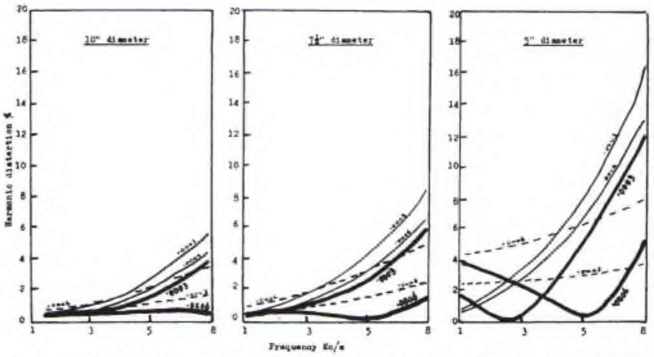


FIG. 4. "Theoretical" distortion with same assumptions as in Fig. 3a but where stylus mass has been halved in all cases, stylus radius has been halved in the .0003 in. cases, recorded velocity has been halved in all cases, and groove speed has been changed for 7½ in. and 10 in. diameters. *Thin lines*: stylus mass distortion; *dashed lines*: tracing distortion; ***heavy lines***: resultant "total."

indent D of a moving spherical stylus of radius r varies as $1/r^{1/3}$ and any resulting distortion also varies accordingly. Thus, if we now consider a stylus with a radius of .0003 in. instead of the .0006 in., the stylus mass distortion will be multiplied by $1/(.0003/.0006)^{1/3} \simeq 1.25$. If we also consider that the pickups to be assessed have stylus masses in the region of ½ mg, then we must also multiply $z\%$ for the 1 mg case by $(\frac{1}{2})^{2/3}$ (from the expression for $z_m\%$ above). Similarly, for the lower velocity, $z\%$ must be multiplied by $(\frac{1}{2})^{-1/3}$. Redrawing this curve with corrections, together with that of the tracing distortion for the .0003 in. stylus, gives the resultant shown in Fig. 4 where this is done for 10, 7½, and 5 in. diameters. It will be seen that a small-

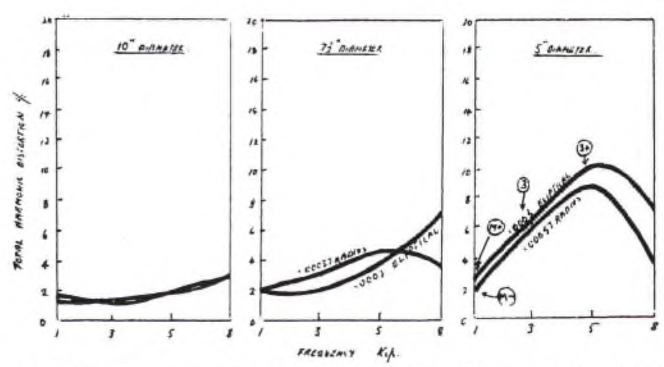


FIG. 5. Measured distortion on a high-quality magnetic pickup having a measured stylus mass of 1½ mg. *M* indicates occurrence of modulation noise, *W* indicates production of wide range of harmonics of high amplitude, *3* indicates production of large proportion of third harmonic. Recording velocity 6 cm/sec peak.

radius stylus could, therefore, result in more distortion.

The measured results in Figs. 5-8 seem to corroborate this view. The results presented are a small representative selection of the measurements made over a wide range of cases and levels; the measurements not shown, however, do not detract from (but rather add to) the present case.

It is seen that the pickup with the higher stylus mass (Figs. 7 and 9c) shows generally greater distortion with the elliptical stylus. This is typical for styli of over 1 mg. (It is interesting to note that the lower levels have suffered most—see Fig. 7[7].) For examples of pickups with the smaller measured stylus mass shown in Figs. 6 and 8 (see Fig. 9 for stylus impedance curves), there is an advantage in using a smaller stylus radius in the one shown in Fig. 8, whereas in the other (Fig. 6) there is again

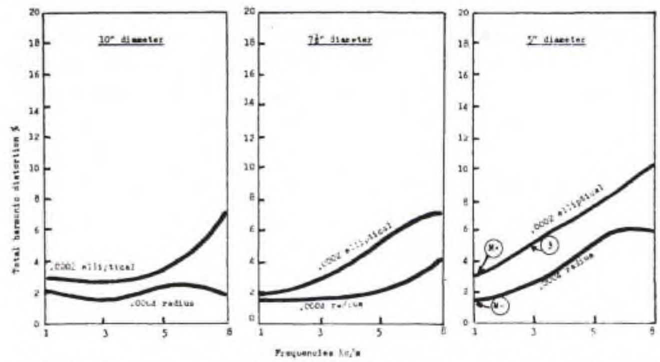


FIG. 6. Measured distortion on a high-quality magnetic pickup with a measured stylus mass of under ½ mg. Note that the already low distortion from a .0004 in. stylus is here increased when using .0002 in. elliptical stylus. (*M, W,* and *3* as for Fig. 5.)

greater distortion when the elliptical stylus is used. The exception provided by this pickup, which has the very lowest stylus mass, might well be explained by the equally very low minor radius of .0002 in. The impedance curves are not repeated for each stylus radius used because whether the stylus was changed by using a complete assembly or by using separate pickups, there was no significant measurable difference in impedance.

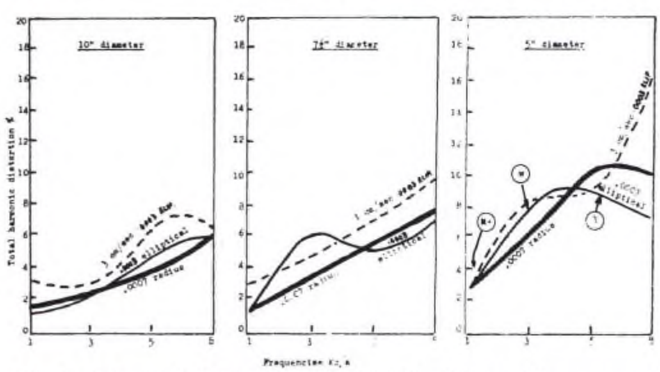


FIG. 7. Measured distortion on a magnetic pickup with a measured stylus mass of 1.2 mg. (*M, W,* and *3* as for Fig. 5.)

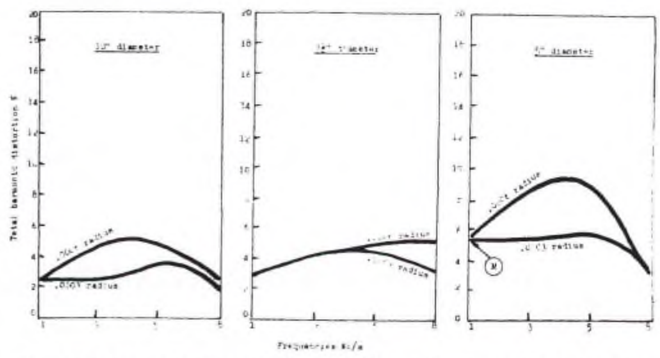


FIG. 8. Measured distortion on a high-quality ceramic pickup having a measured effective stylus mass of 0.6 mg. (In this case, the stylus is a "decoupled" type.) (*M, W,* and *3* as for Fig. 5.)

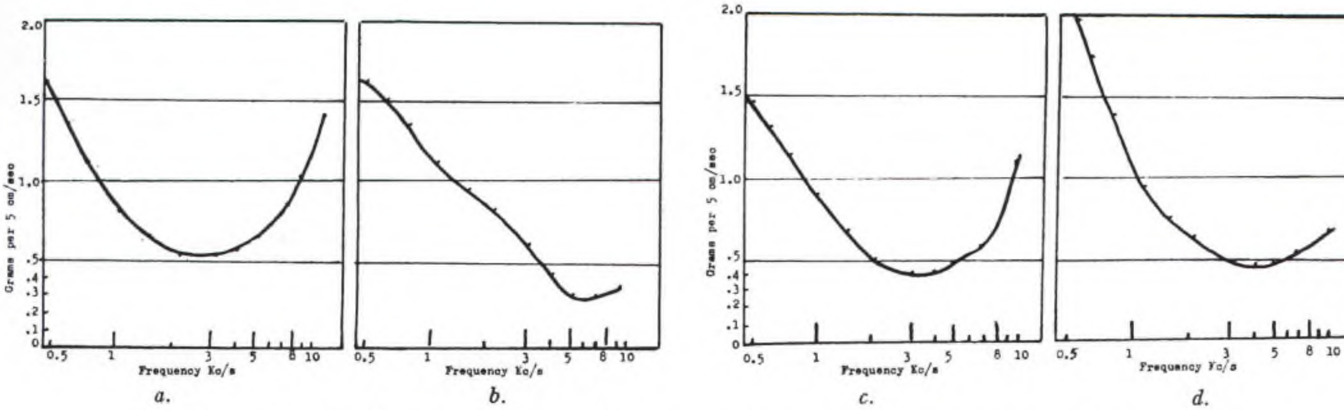


FIG. 9. Mechanical impedance curves at the minimum tracking weight required for the stylus to remain in contact with the grooves. Modulation is plotted against frequency for *a*. Pickup of Fig. 5, *b*. Pickup of Fig. 6, *c*. Pickup of Fig. 7, *d*. Pickup of Fig. 8. Note that due to the low figures and the slope of the curves above 5 kc, the styli of parts *a*, *b*, and *d* are considered to have low stylus mass, whereas that of part *c* with its steeply rising curve between 5 and 10 kc is considered to have higher stylus mass. Note also that the curve of part *d* does not rise steeply even above 10 kc on account of its decoupled rondel, while the other pickups have normal steep rise at the stylus-groove resonance somewhere above 16 kc.

The evidence of the electron micrographs in Figs. 10 and 11 is significant in showing that there is a much greater phase shift between stylus path and groove modulation for this smaller-radius stylus. Since this phase shift is due to indentation of the record groove, it should, according to the above concepts, be accompanied by greater nonlinearity even though this nonlinearity is too small to be detected visually.[7]

Five pickups of similar characteristics had an average stylus mass of 0.9 mg. They showed neither an increase nor a decrease in distortion whether using a .0005 or a .0003 in. radius stylus. All pickups with stylus masses over 1 mg, however, showed some distortion increase with elliptical styli.

Pickups that were measured with stylus radii above .0006 in., however, again showed an increase of distortion with such large radii. It seems, therefore, that there is an optimum relation between stylus radius and stylus mass.

It is seen (Figs. 12 and 13) that even when the modulation is lateral (as is the case for much of the information from stereo records, i.e., midway between the two loudspeakers) and the pinch distortion is then reproduced by the stereo pickup, a small-radius elliptical point can still show more distortion than the .0005 in. radius.

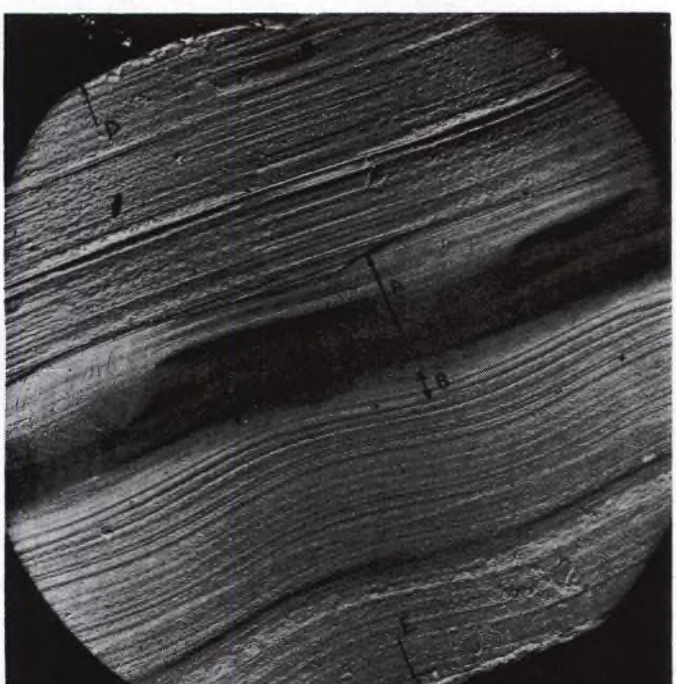


FIG. 10. Electron micrograph (enlarged 2200×) of the tracks caused by a .0004 in.-radius stylus in the pickup of Fig. 6 (5 cm/sec, 10 kc, 11 in. diameter, 3 g tracking weight). *D* is the top edge of the modulated wall; *E* is the top edge of the unmodulated wall; *C* is the groove bottom; *A* is the track of the stylus on the modulated wall; and *B* is the track of the stylus on the unmodulated wall. Note that track B appears to be identical with the cutter path.

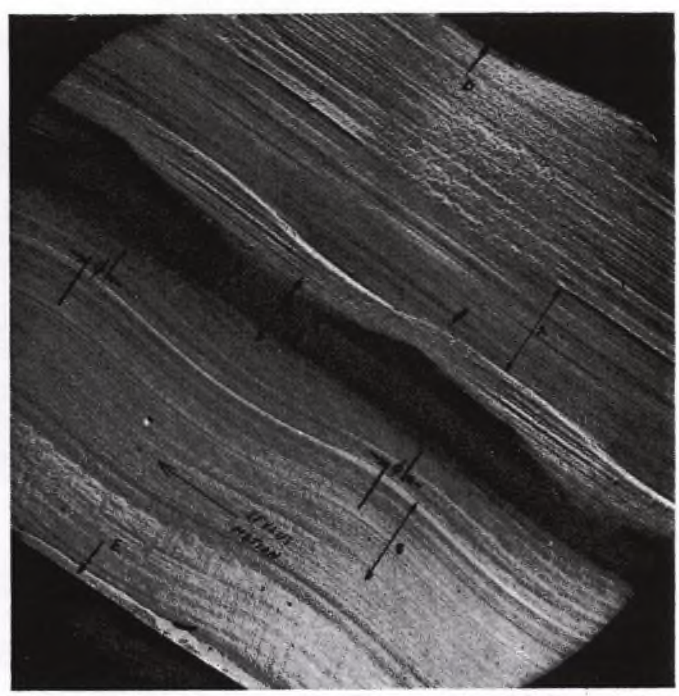


FIG. 11. Electron micrograph of same groove, modulation and pickup as for Fig. 10 but tracked at 2¼ grams with an elliptical point of .0002 in. minor radius. It will be seen there is a phase shift of approximately 18° in this case. While the velocities are too small to detect visibly any distortion, the phase shift should be significant if the stylus is considered as a curved profile indenter. Track B shows the stylus path in relation to the cutter path lines on the same wall.

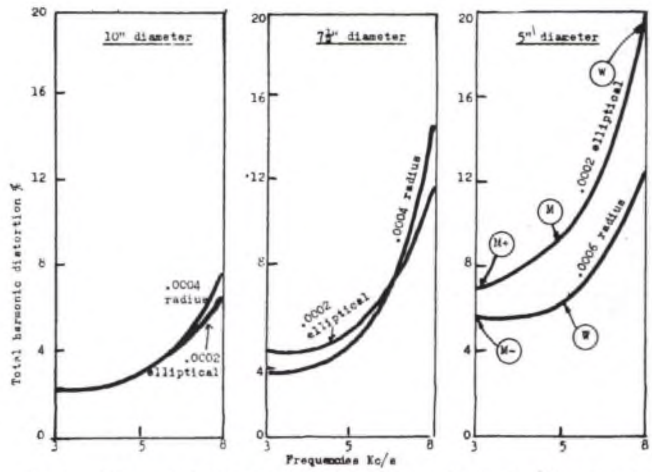


FIG. 12. Measured distortion of the pickup shown in Fig. 6 reproducing 10 cm/sec rms lateral (center of stereo image) recording, while connected as stereo pickup. The measurements are on the left-hand channel.

MODULATION NOISE

It was also noticed that there was an increase in third harmonic with use of the smaller stylus radius in every case, particularly at high level, low frequency, towards the inside of the disc, and at 5 in. diameter. At 6 cm/sec (and over) this increase in third harmonic content developed into a *predominance* of third harmonic and a developing *series* of harmonics. At the lower frequencies (1 to 3 kc) this series of harmonics was extended and became synonymous with an audible effect of "modulation noise."

Deformation of the groove by stylus inertia could conceivably produce a wide range of harmonics due to the complex nature of its nonlinearity; frequencies above 3 kc are unlikely to produce the same noise phenomenon if the higher harmonics of their deformation are suppressed by scanning losses, etc.

Cronin has shown[8] how discrepancies of vertical tracking angle can result in modulation noise due to frequency modulation effects with combined lateral and vertical components. This hardly seems to be the effect here, however, since not only is just one channel involved, but the noise was least on a pickup with the greatest vertical tracking error (see Fig. 8).

A modulation noise effect has been noted by other workers, some of whom have investigated it from the point of view of a groove-top defect combined with the large major axis of the elliptical stylus. Since it also occurs with small radius *spherical* styli, however (see Fig. 8), it seems unlikely that this consideration is relevant, and one is drawn to the idea that the spurious harmonics are the result of an increased proportion of stylus mass distortion.

CONCLUSIONS

The complex nature of the effects of stylus radius, as demonstrated by the curves in Figs. 5-8, make it necessary to measure distortion over a wide range of frequencies and levels before assessing the merits of either increasing *or decreasing* the effective stylus radius.

The fitting of an elliptical stylus to *most* pickups will result, at present, in more distortion—not less. Only those pickups with a *measured* effective stylus mass in the region well below 1 mg can benefit by a .0003 in. minor radius, and the present radius of .0005 in. seems best suited to the majority of good pickups with their stylus masses in the 1-2 mg region.

One cannot, of course, undertake pickup development work without considering the implications of developments in recording.

Professor Cooper has pointed out[9] that the whole cycle of designing lower-mass styli for smaller stylus radii, and then finding more stylus-mass distortion requiring an even smaller stylus mass, is broken with the adoption of recording-

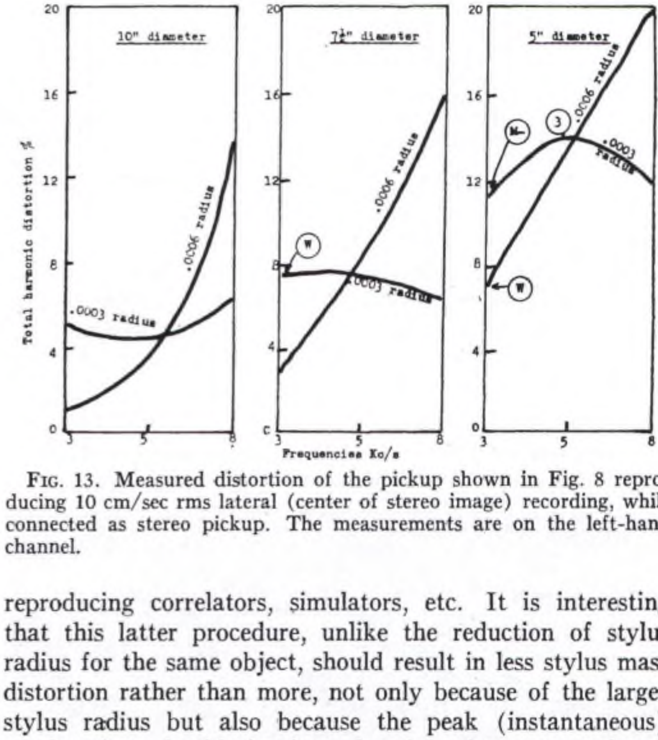


FIG. 13. Measured distortion of the pickup shown in Fig. 8 reproducing 10 cm/sec rms lateral (center of stereo image) recording, while connected as stereo pickup. The measurements are on the left-hand channel.

reproducing correlators, simulators, etc. It is interesting that this latter procedure, unlike the reduction of stylus radius for the same object, should result in less stylus mass distortion rather than more, not only because of the larger stylus radius but also because the peak (instantaneous) accelerations should also be reduced in the process of correcting the waveform. Measurements of resulting distortion carried out in the same manner as described here confirm that less overall distortion does in fact occur.

If the best stylus radius is to be pre-determined by the recording engineer, however, then the effective scanning aperture[10] of the system is set. But since the disc medium will allow of a smaller scanning aperture without deterioration of signal-to-noise ratio,[11] then to set this aperture (i.e., stylus radius) too large is to set a limitation to the exploitation of the disc medium. Such limitation will then need nearly as much effort to overcome, as did the change from 78 rpm to LP microgroove. It is therefore difficult to understand why .0007 in. is being chosen as the standard when the better replay devices with .0005 in. radius will thus not receive the same advantage, especially when it is the replay device that still limits the ultimate development of the disc recording.

With this qualification in mind, however, it is apparent that the two methods of reducing tracing distortion, i.e., in recording and in reproduction, are not of equal value in the present state of the art, and that any reduction of reproducing stylus radius should be related to the stylus mass. To this end the "decoupled rondel"[12] with its cheap mass production should be of considerable significance (see Figs. 8 and 13) since it gives advantages in the reproduction of the "correlated" recording as well as that of the uncorrelated.

ACKNOWLEDGEMENTS

I wish to acknowledge the unique efforts of Dr. Chippindale of Salford C. A. T. in producing the electron micrographs, and to thank Dr. Woodward of RCA for the notice of, and for the copy of, Shiga's paper. Thanks also to Professor Cooper and to Mr. Heath of Regent Polytechnic Institute for their stimulating discussions on the mathematics of distortion.

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THE AUTHOR

John Walton obtained a Higher National Certificate in Electrical Engineering after several years experience in a variety of aspects of industry which ranged from telephone linesman to factory progress chaser to accounting.

After 1941 he worked for several years on electronic measuring instrument testing and later on instrument and amplifier design. He has spent the last nineteen years mainly on design of both disc and tape recording apparatus, and the last eight years specifically on the design of gramophone pickups.

Mr. Walton is now with the Decca Record Company Ltd., London, England, and is engaged in gramophone pickup design and research into its relation to the gramophone record.

FURTHER REMARKS ON STYLUS MASS AND ELLIPTICAL POINTS

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Further to my paper on stylus mass and elliptical points (see p. 153), I would like to discuss one particular aspect of controversy that apparently is based on a misunderstanding both on my part and on that of some of those who would question the validity of my results.

The basis of the misunderstanding may be described, I believe, in the following manner.

The conditions under which my measurements were taken are those within the range of proper tracking, whereas the listening tests are usually concerned with that "end of side" distortion where curvature overload occurs, i.e., so many records have high frequency modulations that present a curvature that is of smaller radius than that of the stylus, with the result of much audible distortion. The elliptical stylus, of course, with its lower effective radius, reduces this considerably. This is a valid reason in practice for the use of an elliptical stylus, because any increase in distortion at more normal levels may not be so noticeable anyway.

I cannot help saying, however, that such a comparison of performance of styli is like the comparison of two ten-watt amplifiers at a 20-watt level when one amplifier has negative feedback and the other has not. It is known that the amplifier with feedback will have lower distortion over its intended working range, but any overload may cause more severe distortion on this amplifier than on the one without feedback.

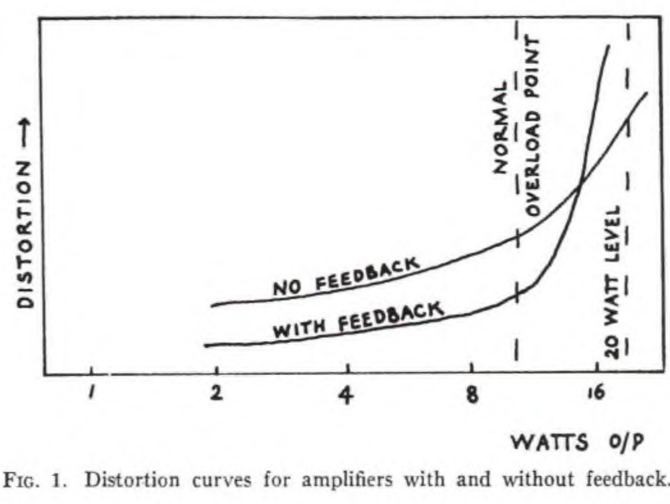

FIG. 1. Distortion curves for amplifiers with and without feedback.

The Diamond as an Industrial Material, With Special Reference to Phono Styli*

E. J. AND M. V. MARCUS

The Tetrad Corporation, Yonkers, New York

Editor's Note:

Writing in the North American Review in 1878 about sound reproduction from tinfoils, Edison asserted that the use of "perfectly smooth stone point—diamond or sapphire—(would) render the record capable of from fifty to one hundred repetitions, enough for all practical purposes." This anticipation makes the work of E. J. and M. V. Marcus no less valuable. An enormous leap was needed to make the dream of a diamond tip in everyman's pickup a practical reality.

During the development of the phonograph industry in the 1920's and 30's, the predominant stylus material was steel. Not that all else had not been tried—there were even "cactus" styli which were supplied together with a little machine for sharpening after each play! (Not long ago we were asked to re-record some rare instantaneous recordings made on *aluminum* blanks. We had to outfit a modern pickup with a Delrin tip to play them without scoring.) But steel was king, and steel "needles", as they were called, came in all sorts of sizes and shapes, labeled "loud," "medium," and "soft," depending upon length and thickness. They were readily replaceable by means of a thumb-screw. A packet of needles bore the invariable warning, "For best results, use only once." No one ever paid attention to this admonition, which was merely a "gimmick" to promote the sales of needles. The sound was equally poor after the first as after the tenth playing, but we loved it. This magic of unlocking the sound in a groove has never disappeared.

In the early phonographs, no one seemed to worry too much about the geometry of the needle tip, which often was allowed to wear down to a chisel shape. To AES President Frederick Hunt goes the credit for demonstrating in 1937 that a smooth round tip *was* needed for proper groove reproduction, and that it ought to be supported by the walls of the groove, not by its bottom. This idea acted as a powerful stimulant to the engineer to seek more durable styli and lighter-weight pickups, thus ushering in the era of "high-fidelity."

In the late 30's there was a growth of popularity of needles tipped with osmium and other long-wearing alloys. These gave way to sapphire, which with Linde's development of synthetic sapphire rod became readily available. For many years, sapphire dominated the pickup stylus scene, while the diamond was used only in pickups intended for broadcasting, such as the famous Western Electric 9A. Even as late as the mid-50's diamond tips were beyond the reach of an average user. E. J. and M. V. Marcus can be credited with developing technology needed to make diamond styli available for the vast majority of the better pickups. Synthetic diamond chips also are playing a major role in the phonograph field. It is fair to say that without the hardness, toughness, and durability of diamond, the elliptical stylus tip so popular today could not have become a reality. Reprinted below is the original Marcus paper relating to the use of diamond styli in phonograph pickups. **B.B.B.**

INTRODUCTION Upon examining the physical properties of groups of raw materials found in nature, we generally find small increments in the values of these properties. Diamond is the outstanding exception to this rule in that it possesses durability and hardness values

* Reprinted from **Audio Engineering,** July, 1950.

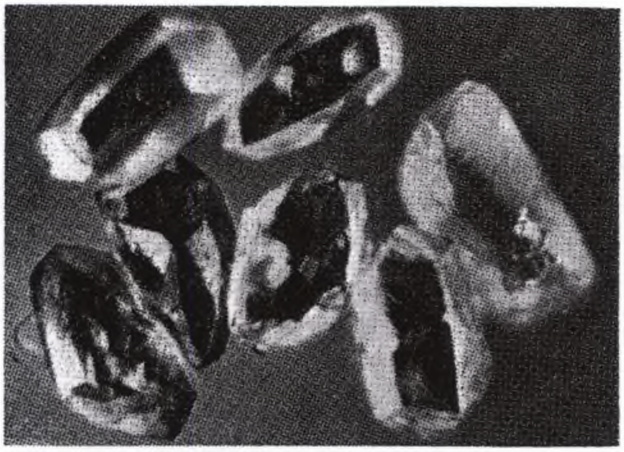

Fig. 1. Photomicrograph of a number of industrial diamonds. Seventy of these tiny stones weigh 0.2 g, or one carat.

which exceed those of the closest comparable material—sapphire—by almost a hundred fold.

Diamond was recognized, even in antiquity, as possessing properties which have made it the most valuable material (except for some radio-active elements) on earth. These properties are durability, beauty, and rarity.

For technical purposes, we are most concerned with the durability and hardness of diamond.

FORMATION AND OCCURRENCE OF DIAMOND

Diamond is pure crystalline carbon occurring sparingly in volcanic rock. Carbon, trapped in molten lava and subjected to tremendous pressure and heat crystallizes slowly to form diamond. Although it is supposed that all diamonds were formed in this manner, only in South Africa do we actually find large volcanic intrusions or "pipes" containing deposits of diamond. Diamonds found in other parts of the world, the Belgian Congo, Brazil, India, Borneo and Australia, are found in alluvial deposits. The original volcanic sources remain undiscovered. Africa produces about 95 per cent of the world's production, which has amounted to as much as 13,000,000 carats yearly (one carat = 0.2 g).

The diamond-bearing volcanic pipes in South Africa were discovered accidentally between 1866-1888. These pipes, called blue ground, contain approximately 1 part of diamond in 15,000,000 parts of rock. The soft blue ground is mined, crushed and washed in troughs containing grease pans. Diamond, having a great affinity for oil or grease, sticks to the grease in the pans, and the stones are then collected and sorted.

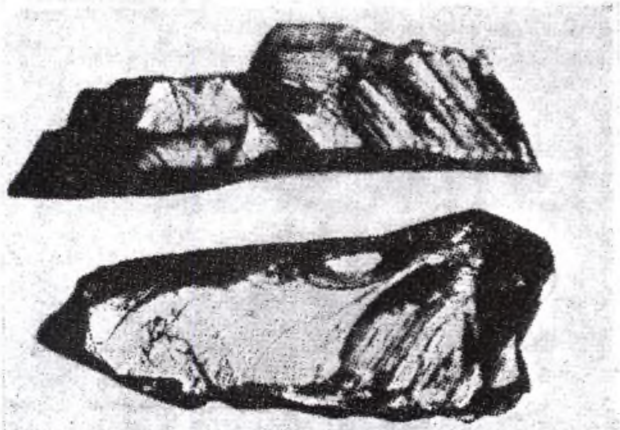

Fig. 2. Diamond cleavages, or chips from larger stones.

CLASSIFICATION OF DIAMOND

There are various types of diamonds, and each has its specific application. The most valuable are the large, clear white, nearly flawless stones which are used for gem purposes. However, there are other types used for industrial purposes which possess the same physical and chemical properties but are unsuited for gems. Diamonds may be transparent, cloudy or opaque, clear white, or colored in various hues. The qualities of those diamonds suited for use as styli for audio equipment will be discussed in this paper. Commercial diamonds are classified in several groups, two of which are:

1. *Fine Industrials.*—Diamonds unsuited for use as gem stones due to small size, color or presence of small inclusions. In all other respects these stones are similar to the diamonds used for gem purposes.

2. *Cleavages or Splints.*—Chips cleaved from larger stones, usually during gemstone production. Lack the structural strength of small fine industrials. Sometimes employed for use as phonograph styli due to the comparative ease of fabrication.

Fig. 3. Each diamond stylus must be inspected carefully on a shadowgraph.

FABRICATION OF DIAMOND STYLI

There are three principal methods of shaping diamond for industrial use, namely bruiting, lapping, and sawing. All other methods are usually variations of the three employing special fixtures.

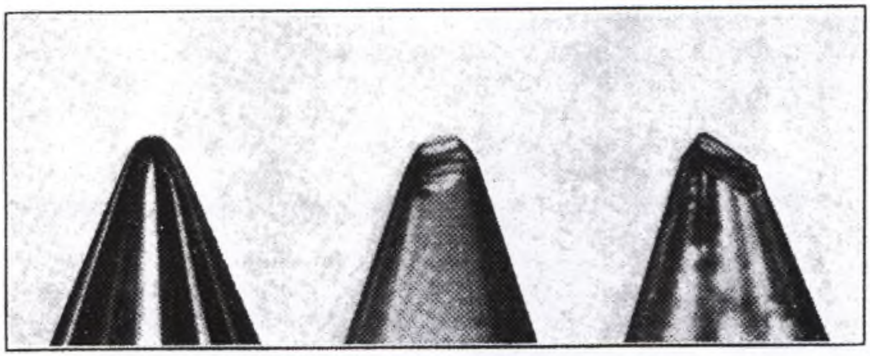

Fig. 4. Different 1.0-mil styli after fifteen plays on 12-in. Viny lite records with an 8 g pickup. *Left*, diamond; *Center*, sapphire; *Right*, osmium.

All methods use diamond in some form to reduce the size or change the shape of another diamond. Diamond is the only material hard enough to cut diamond effectively. Large stones are sometimes cleaved along cleavage planes which can be determined by experts.

Bruiting

The bruiting operation is similar to freehand woodturning in a lathe, using a tool rest. The diamond to be worked is placed in a special holder, called a *dop*, and put in the lathe chuck. A diamond tool placed in a long-handled holder and braced under the arm is worked against the spinning diamond in the lathe. This method is quite effective for rapid removal and rough shaping of diamond cones.

Lapping

This is the term used to describe the polishing of diamond surfaces and usually means the generating of a facet on a stone. Lapping is done on a rapidly spinning, porous cast-iron wheel which has been impregnated with a mixture of diamond powder and olive oil. The stone is held in the dop and placed in a removable arm called a *tang*. It is then placed against the lap or wheel. Diamond crystallizes in the cubic system and has three equal axes intersecting at 90°. Lapping can only be done along planes which have a definite relation of these axes. Under the usual lapping conditions, diamond will not polish or cut readily against the so-called *grain* or out of relation to these planes. Therefore, in the polishing of diamond facets it is necessary to orient the stone properly to obtain the lapping direction. In polishing a cone and radius on a diamond, it is evident that rotating the diamond cone will encounter grain running in several directions and that conventional polishing methods can not be used. It was found necessary to polish at extremely high speeds with very fine diamond powder to develop a high polish on the cone and radius of the stylus. The entire polishing apparatus must be free of vibration and exceptionally accurate.

Sawing

Diamond can be sawed with a thin phosphor-bronze saw charged with diamond powder and olive oil. Diamond-impregnated metal saws are also used. The diamond to be sawed is nicked with a sharp pointed diamond and moved into the saw. There are optimum sawing directions, depending upon the crystal structure of the stone.

CHEMICAL AND PHYSICAL PROPERTIES

Lavoissier discovered that diamond consisted of carbon when he burned diamond at high temperatures to form carbon dioxide, CO_2. Diamond can be burned in oxygen between 700° and 900° C.

Diamond is very resistant to strong acids and alkalis. Concentrated solutions are frequently used to clean the stones. A mixture of sulphuric acid and potassium bichromate can oxidize diamond slowly at 200° C. Diamonds are cleaned of foreign matter from the mines by storing in hydrofluoric acid. They are cleaned after processing by cooking in a concentrated solution of potassium hydroxide, usually followed by a second cooking in sulphuric and nitric acid. Aca regia is sometimes used to clean diamonds.

Remarkable as diamond is in its resistance to chemical action, it is even more outstanding in its amazing physical properties. The compact arrangement of the carbon atoms in the diamond has resulted in an extremely durable and hard material.

Table I lists the physical properties of diamond compared with alternative materials.

Table I. Physical properties of diamond and alternative stylus materials.

	Diamond	Alternative Material
Resistance to abrasive wear (Rosiwal)	90,000	Sapphire 1,000
Wear resistance, path of turning tool (Grodzinski)	1,250	Carbide 12.5–20.5
Ratio of time required to saw given area (Grodzinski)	100–300	Sapphire 1.0
Indentation hardness (Knoop)	6000–6300	Sapphire 1600–2000
Initial bearing friction (Shotter)	0.70	Sapphire 1.13–1.60
Breaking load on a radius (Schuler)	25	Sapphire 5
Compressibility (Williamson)	0.18	Sapphire 0.38
Surface Finish (Kayser)	Better than any other material	
Index of refraction-sodium	2.419	Glass 1.426
Dispersion	2.465	Glass 1.532

Note: Moh's scale (1820) which gives comparative but not quantitative hardness values of gem materials has been omitted. Moh listed gem materials in order of ability to resist scratch marks.

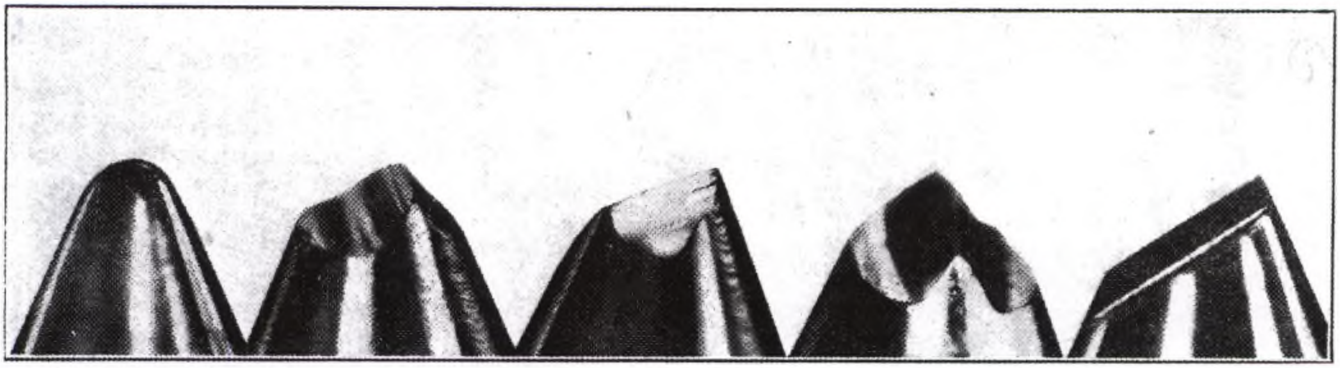

Fig. 5. 2.5 mil styli after 1000 plays on 10 in. Vinylite record with a 1½ oz pickup. *From left to right*: diamond; sapphire, front view; sapphire, side view; osmium, front view; osmium, side view.

The question is frequently asked, "How hard is diamond?" This is a difficult question to answer, since hardness is a composite property embracing many characteristics. When one thinks of something as being hard, one is at the same time thinking of a number of properties such as resistance to wear, ability to resist indentation of a sharp point and non-compressibility. For example, if we say a piece of wood is hard, we mean it cannot be sawed or chopped easily, or that it is difficult to drive a nail into it. If we say a piece of metal is hard, we may mean that it cannot be filed, sawed, or bent readily. Therefore, an answer can only be given in terms of the job being performed by the given materials. Numerical results for comparative hardness and durability of materials will vary from one set of conditions to another. However, a glance at the table of comparative physical properties shows that diamond has a tremendous advantage over its next-best alternative material in resistance to abrasive wear, breaking load on a radius, indentation hardness, and compressibility.

PHONOGRAPH STYLUS WEAR TESTS

Tests have been conducted to determine the comparative durability of various stylus materials.

When a playback stylus touches a record groove, only a small area of the stylus tip actually makes contact with the groove walls. The pressure per square inch may amount to several tons. The dynamic forces acting on the stylus tip are several and severe. Hard record materials containing abrasive have a greater ability to wear styli, but soft record materials often become imbedded with abrasive particles which also can cause rapid stylus wear. Stylus wear is rapid at first, and then, as the contact area becomes larger and the pressure per square inch decreases, wear continues at a slower rate. This effect is noticed with new clean records: however, as records become worn and the grooves become progresively loaded with abrasive particles, the rate of stylus wear may continue at a comparatively rapid rate.

The softer stylus materials have a great tendency to load the record grooves with abrasive particles, as shown by G. A. Briggs [1]. It is a fallacy that styli made of soft materials cause less record wear. Brigg's photomicrographs show steel flakes imbedded in the grove walls after only one playing of a new shellac disc by a steel needle. Another photomicrograph shows a fibre needle completely abraded after one playing. These conditions lead to excessive record wear and poor response.

Our tests were conducted on Vinylite records, both standard groove 78 rpm and microgroove 33⅓ rpm, using three stylus materials: diamond, sapphire, and osmium. The conditions observed for all styli under test are summarized in Table II.

Table II. Test conditions for stylus wear tests.

	Standard Groove	Microgroove
Record material	10 in. Vinylite	12 in. Vinylite
Record player	Garrard changer	Webster 3 speed changer
Cartridge	Astatic L 70	G.E. RPX-041
Pickup weight	1¼ oz	8 g
Radius of stylus	2.5 mil	1.0 mil

Test methods were made to conform to the severe playing conditions usually encountered with home-type equipment, and both tests were repeated twelve times to eliminate the possibility of error.

The results of the standard-groove test showed that after 1000 plays, or approximately 50 hours playing time, wear on both the sapphire and osmium tips were very severe. Considerable wear was noticed as early as 100 plays. There was complete conformity to groove shape after 1000 plays. The diamond stylus did not show any wear after 1000 plays. Shadowgraphic tracings were made after each 100 plays and showed the progression of wear on the sapphire and osmium tips. Photomicrographs were taken of the stylus tips after 1000 plays. Two views were photographed, one parallel to the groove position, the other perpendicular to the groove.

Superimposing the osmium shadowgraph tracing over the sapphire tracing, we found the same amount of wear for sapphire at 1000 plays as for osmium at 400, or a 10:4 ratio. Since the amount of material revoved from the diamond was so slight as to escape notice at 500 times magnification, the ratio of wear could be written Diamond ∞, Sapphire 10, Osmium 4. However, eventually even the diamond will begin to show wear.

The results of the microgroove test were rather surprising. We found that after only 15 plays or 5¾ hours playing time, both the sapphire and osmium tips were very badly worn, and the test was stopped at the time. Again the osmium wore out faster than the sapphire.

Photomicrographs and tracings were made at the completion of the test. The diamond showed no sign of wear after 15 plays. The microgroove diamond was continued in use for a total of 100 plays or 37 hours. A slight flat was noticed on the diamond at the end of 37 hours.

Apparently, stylus wear is far more rapid on microgroove than on standard groove records. A rough estimate is about three times faster. The rate of wear of the 1.0 mil osmium and sapphire styli caused by microgroove playing is so rapid as to make these materials unsuited for continuous use on microgroove records.

To return now to the hardness values given in the table, we can see that it is necessary to conduct tests under the conditions of use before hardness values can be assigned to any materials.

If we could accurately weigh the stylus tips before and after the tests, it is quite certain we would find a difference between the diamond and sapphire in excess of the 90:1 ratio given by Rosiwal.

According to Ridgway and Eppler, the differences in relative mechanical corrosion hardness become smaller as the hardness of the material used for the corroding or abrading increases. They used a sand-blast technique in determining the resistance of various materials to corrosion. As the hardness of the powder used for the sand-blast was increased in going from quartz to sapphire to silicon carbide, the differences in the hardness values for the test materials became smaller. We observed corresponding results in our stylus wear tests. As we have already shown, the wear ratio between diamond and sapphire when played on Vinylite records is far above the Rosiwal 90:1 ratio, while this difference is not quite as great when these materials are used on shellac records.

Other results of our stylus and record wear tests showed that wherever there was excessive stylus wear it was always accompanied by excessive record wear. The badly worn styli are unable to track properly and have a tendency to chop off the high-frequency crests in the grooves. It was observed that stylus pressure, pick-up arms, and cartridges have a great effect on stylus and record wear. Considerable variation in rate of stylus wear was noticed in making wear tests on different types of equipment. For example, one professional arm caused 1.0 mil styli to wear out about five times faster than did other arms designed for amateur use.

Quality of response deteriorates gradually as stylus wear increases. This deterioration is noticed more readily on good quality audio equipment. On hometype equipment, with its narrower frequency range, stylus wear is usually not noticed audibly until it has become very bad. The gradual wear over a long period of time occurs slowly, and therefore the listener does not readily notice the change. He cannot remember how his equipment sounded six months before when the stylus was new. This has the unfortunate effect of causing excessive record wear by continuing the worn stylus in use.

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ON DIAMOND STYLI

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The earliest reference cited by Marcus and Marcus[1] was 1917, but the authors mention in the first paragraph a remark made by Edison in 1878. I have two Edison phonographs, one purchased new in 1923 (disk) and one believed to be a product of about 1905 (cylinder). Both have diamond styli which appear to be in excellent condition. These are "black" diamonds, supposedly even harder than the white ones.

[1] E. J. Marcus and M. V. Marcus, "The Diamond as an Industrial Material, With Special Reference to Phono Styli," *J. Audio Eng. Soc.*, vol. 18, pp. 51-55 (Feb. 1970).

Biradial and Spherical Stylus Performance in a Broadcast Disc Reproducer*

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A number of observations are reported on an improved disc reproducer consisting of a tone arm and cartridge assembled as a matched system. Interchangeable stylus assemblies with spherical or biradial tips were used with conventional test records. Since only the stylus assemblies were changed, the effects caused by changes in tip radii are readily seen. The data touch on several problems involving biradial and spherical stylus relationships for which a theoretical model is incomplete, especially such effects of biradial styli as extremely high stress imposed on the record groove and the possibility of shorter record life. Record wear data are shown for the biradial stylus which indicate that record life is about the same as with a spherical tip, when both are used in low-inertia assemblies.

INTRODUCTION The biradial stylus has become widely accepted for braodcast, professional audio, and high-fidelity applications. In 1968 more than half of the pickups sold in these markets were so equipped. Unfortunately, it is very difficult for a user to make a choice of pickup and stylus, as very little scientifically measured data are available on pickups currently on the market.

The scanning electron microscope photographs published by Woodward [1] show apparent record damage caused by a biradial stylus, but he mentions that high-frequency distortion results favor the biradial tip. Since the distortion data is not presented, the evidence seems to weigh against the biradial, and in favor of the spherical tip stylus.

RCA has introduced a complete tone-arm–pickup–stylus system known as the BDR-1 Broadcast Disc Reproducer. The BDR-1 has quick-change stylus assemblies with several tips having different radii, which makes it possible for the user to select the stylus best suited to his application or personal preference. The .0002 × .0007 in biradial stylus and the .0007 in spherical stylus assemblies have identical mechanical systems except for the stylus tip. This makes it possible to study the effects of changing only the stylus tip.

EQUIPMENT FOR STYLUS TIP TESTS

The Disc Reproducer System

The BDR-1 is shown in Fig. 1. The inertia of the arm is much lower than that of most arms currently in use, as may be deduced from the photograph. The low inertia is achieved by incorporating a lightweight moving-magnet cartridge as an integral part of the arm; thus the arm does not need to be built to a Gibraltar-like scale to accommodate a wide range of cartridge masses. The arm tube is close to the record to reduce excitation of torsional resonance modes. The straight-line design eliminates a bend in the tube and achieves lateral balance without "outrigger" adjustable weights. Stylus force is obtained by a small offset in vertical balance. The system

* Presented May 5, 1970 at the 38th Convention of the Audio Engineering Society, Los Angeles.

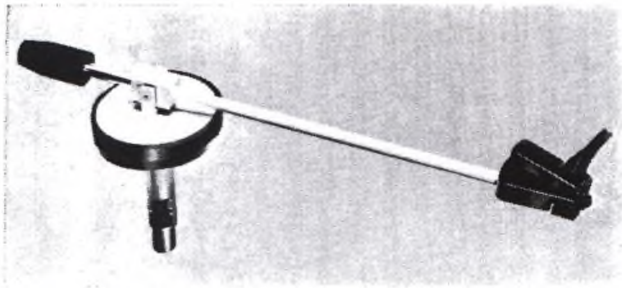

Fig. 1. BDR-1 broadcast disc reproducer.

is factory-adjusted and individual stylus assemblies are weighted as required for optimum force. Anti-skating force is obtained by a linear spring and lever arm arrangement under the decorative cover plate, and is fixed. Precision instrument bearings used on both axes have starting torque very much less than the torque generated by operating forces.

The data were measured using an RCA BA-36A equalizer/preamplifier modified for test purposes, plus various filters and processing networks, and automatically recorded. The curves are unretouched and show the performance that would actually be obtained by a user with a complete RCA disc reproducing system.

Pickup X is a laboratory reference moving-coil pickup with .0007 in spherical tip that was manufactured in Denmark about 5 years ago, which has the lowest tip mass that was available previous to the BDR-1. It was mounted in a high-quality 12 in arm of British manufacture which has several adjustments to optimize pickup performance.

Stylus Tips

The tips were best seen visually through an 80× stereo microscope, and were sketched approximately to this scale. Figure 2 shows the various BDR-1 styli and the stylus of Pickup X. The flats on the biradial tips are a normal result of fabrication and do not contact the groove. It is apparent that the BDR-1 .0002 in × .0007 in biradial, .0007 in spherical, and Pickup X styli have comparable masses and are good units for comparison.

EXPERIMENTAL RESULTS

Frequency Response and Tip Mass

Figure 3 shows the performance of the .0002 in × .0007 in biradial stylus on the STR-100 record, with the measuring system flat for constant velocity. An RC pickup load flattens the response to 20,000 Hz. The CBS records appear to have consistently less output on the left channel at high frequencies. No significant resonances are apparent.

Figure 4 shows the same stylus on the STR-120 record. At low frequencies the measuring system was flat for constant amplitude, while at high frequencies the measuring system was flat for constant velocity. The capacitance load was omitted in order to reduce roll-off above 20,000 Hz. Thus a sag and peak characteristic are observed between 3,000 and 20,000 Hz. The low-

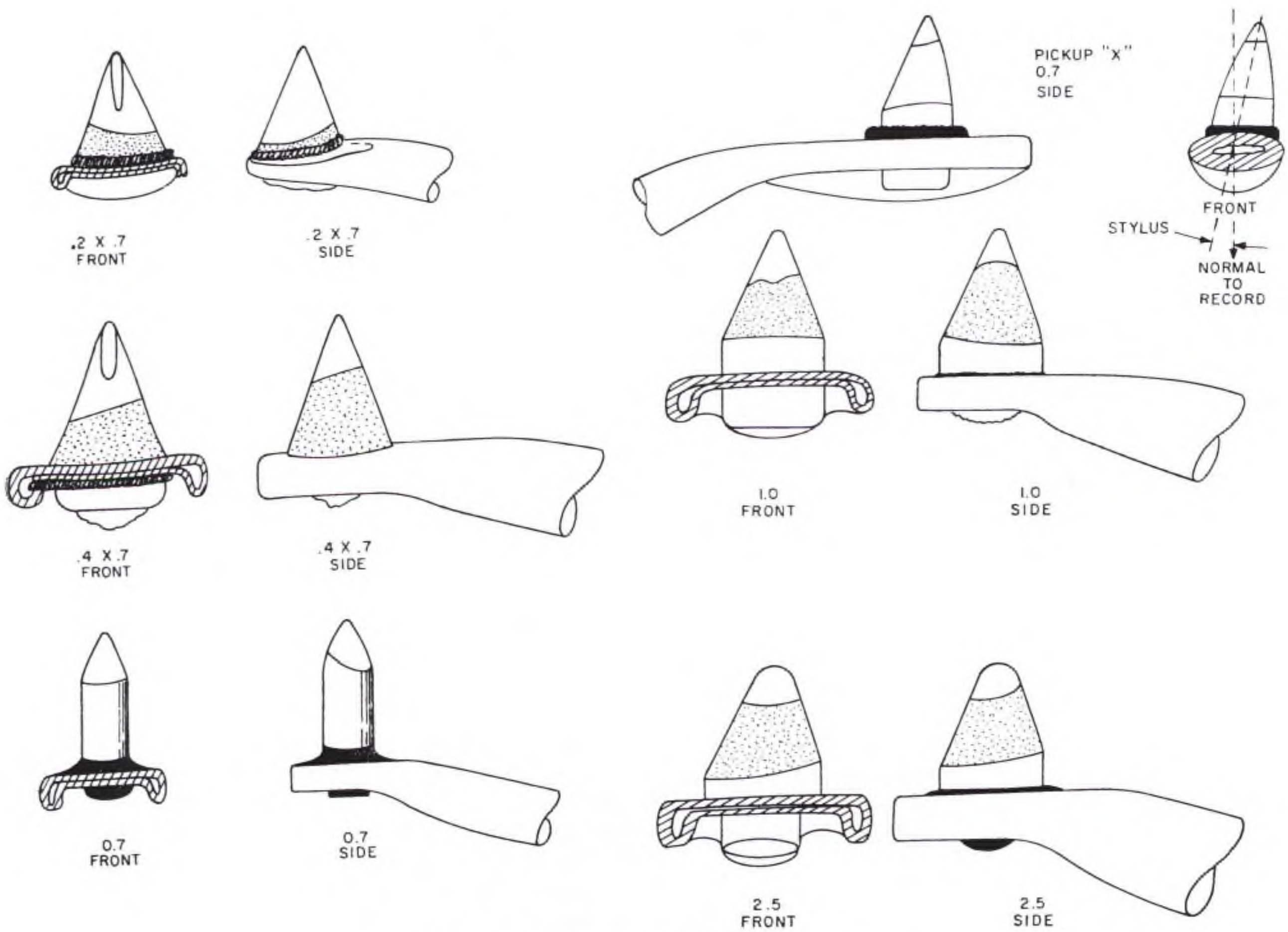

Fig. 2. Stylus tips of BDR-1 and Pickup X.

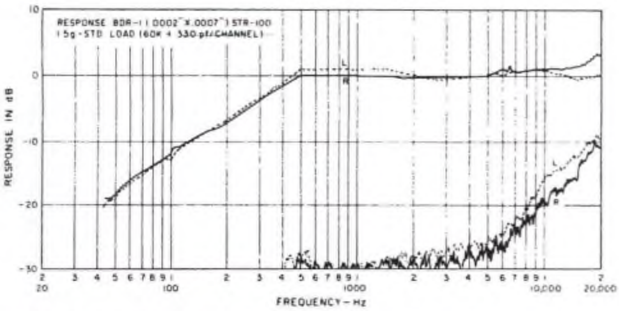

Fig. 3. Response of the .0002 in × .0007 in stylus, STR-100 record.

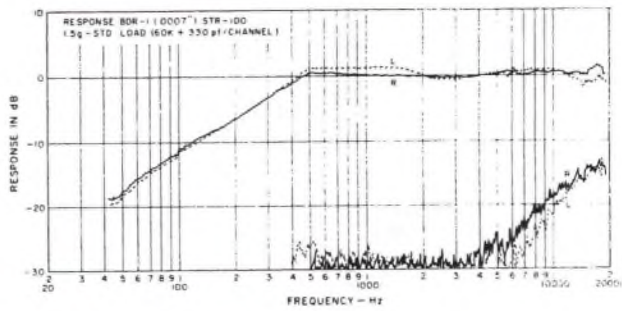

Fig. 5. Response of the .0007 in stylus, STR-100 record.

frequency resonance is high enough for good immunity from record warps and mechanical shock excitation, but low enough to permit good performance in the audible range. The apparent vinylite resonance is 20,000 to 22,-000 Hz.

Figures 5 and 6 show the same tests repeated with the .0007 in stylus. Results are surprisingly similar to those for the biradial stylus. This is not in agreement with theory, which reasons that the vinylite resonance should increase with a larger stylus radius: The larger radius applies less indenting pressure on the groove wall, so that the compliance of the vinylite then appears to decrease, and if tip mass is unchanged, the resonance frequency increases. Previous pickup observations have followed this theory. The best explanation of this phenomena was offered by B. Jacobs [2], who stated that the peak which is displayed on the curves results from a resonance of the mass of the stylus tip and the compliance of the stylus lever arm, in these particular stylus assemblies. The vinylite resonance is about one octave higher but is not displayed due to the electrical system roll-off, as cartridge impedance is very high around 40,000 Hz.

Figure 7 shows the performance of Pickup X on the STR-120 test record. It can be seen that the apparent resonance is about 30,000 Hz. These data were recorded without capacitance load to maximize high frequency output. Figure 8 shows the performance of Pickup X on the STR-100 test record. An RC load was used to flatten the response. Under these test conditions, an apparent resonance of 15,000 Hz is observed.

The evidence tends to confirm Jacob's explanation. The 15,000 Hz and 30,000 Hz resonances of Pickup X would correspond to 22,000 Hz and approximately 40,-000 Hz for the BDR-1. The effective tip mass of the BDR-1 styli would then be lower than that of Pickup X, making the BDR-1 a very good pickup indeed! The differences in response above the audio range are of course inconsequential to the user. However, the decreased mechanical impedance in the audio range is important and could make the BDR-1 biradial stylus as gentle to the grove as the spherical stylus of Pickup X.

Diameter Loss

An easily demonstrated advantage of biradial tips, even before the low-tip-mass .0002 in × .0007 in BDR-1 was introduced, was the reduced diameter loss compared to spherical tips. It is obvious that a loss of this kind, analogous to the scanning loss in an optical film reproducer, can be reduced by making the scanning slit smaller. In this case, this corresponds to the stylus side radius.

Figure 9 shows the diameter loss of Pickup X on the STR-120 record. The 20,000 Hz loss is about 12 dB. This is thought to be typical of high-quality .0007 in spherical-tip pickups.

Figure 10 shows the loss of the BDR-1 biradial stylus. It is much less severe and obviously due to the .0002 in contact radius.

Figure 11 shows the loss of the spherical stylus. This accords with theory in that the spherical stylus has more loss than the biradial one; however, it has much less loss than Pickup X, and therefore agreement with theory is very poor in this respect.

No suitable theoretical model has been found that would explain these data. Sufficient time was not available to investigate the diameter equalization presently

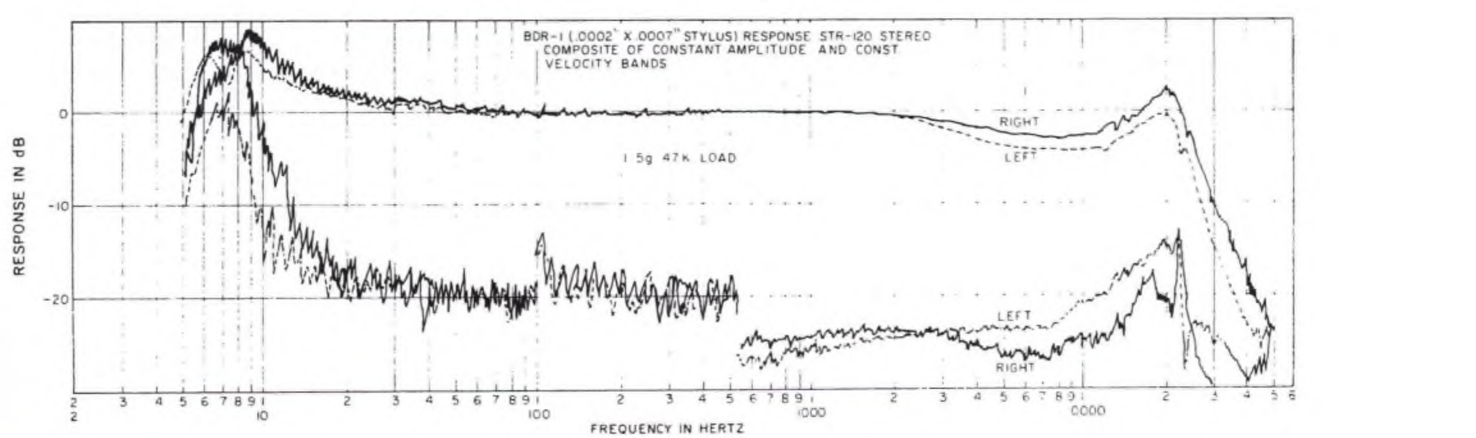

Fig. 4. Response of the .0002 in × .0007 in stylus, STR-120 record.

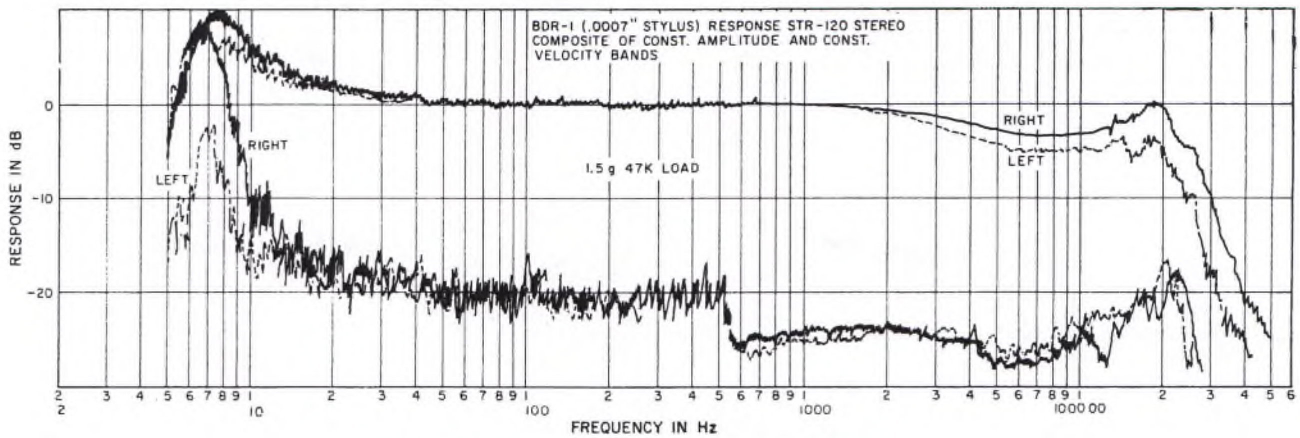

Fig. 6. Response of the .0007 in stylus, STR-120 record.

used by record companies. It would be difficult to choose an optimum equalization from these data.

Record Life

Record life is important for high fidelity applications and some professional users, but perhaps less important to broadcasters. In my experience, a record with the best of care picks up enough dirt in 50 plays to cause "ticks" that would be objectionable in professional use.

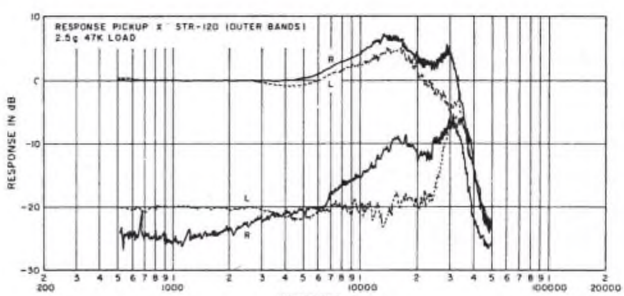

Fig. 7. Response of Pickup X, STR-120 record.

The simplest and most readily performed test was to make 50 plays of a frequency-response record with each pickup in question. The middle bands of the STR-120 test record were chosen to present average wavelengths to the stylus. The 50,000 Hz range would insure that some wear would be recorded on the very best pickup.

The .0002 in $\times$ .0007 in biradial was an obvious choice. Pickup X was chosen as the comparison spherical-tip pickup because of the interesting differences observed previously between it and the BDR-1 biradial. A second reason is that the .0007 in spherical tip was not yet available for the BDR-1 when the tests were made.

Figure 12 shows normalized data for Pickup X, and Fig. 13 shows similar data for the BDR-1 biradial pickup.

It is evident that either pickup produces negligible wear in the audible range. They are both particularly good compared to the previously developed definition of "negligible wear", that is, less than 2 dB change up to 15,000 Hz in 20 plays.

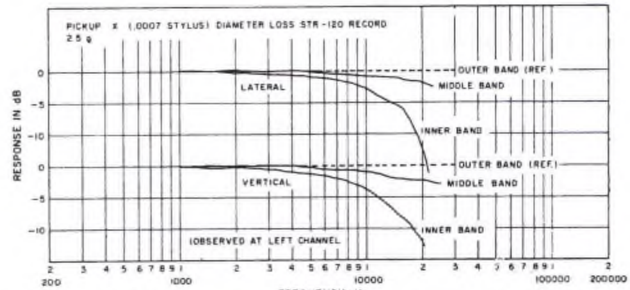

Fig. 9. Diameter loss with Pickup X.

Listening Quality

The BDR-1 .0002 in $\times$ .0007 in and .0007 in were compared to Pickup X, using individual BA-36A preamplifiers with optimum RC pickup loads for the tests.

A few listeners prefered the biradial, but most could not tell the difference between the three pickups. This is a good result, since Pickup X has for years sounded distinctly better than other pickups. The preferences for the biradial were obtained using RCA Dynagroove records, for which the biradial theoretically has no advant-

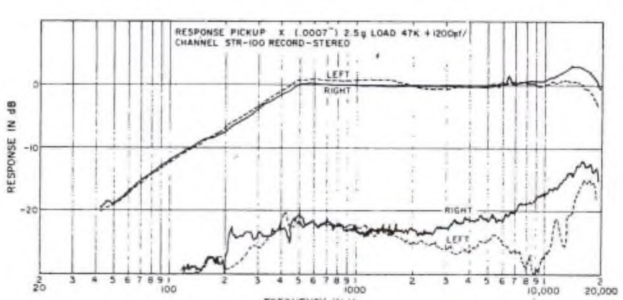

Fig. 8. Response of Pickup X, STR-100 record.

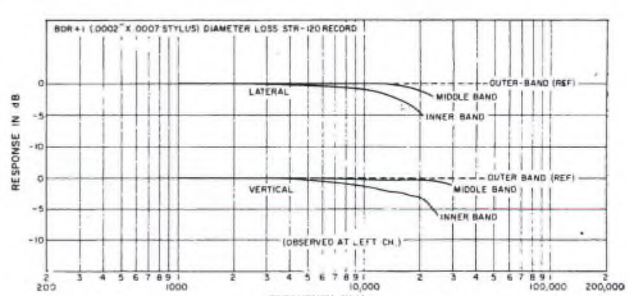

Fig. 10. Diameter loss with BDR-1, .0002 in $\times$ .0007 in stylus.

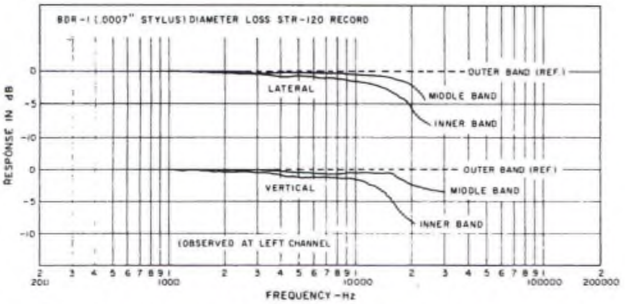

Fig. 11. Diameter loss with BDR-1, .0007 in stylus.

age. A non-RCA record was found for which the distortion on trumpet passages was equal on all three pickups.

The large differences observed for diameter loss were not heard, and no reason for this is known.

CONCLUSIONS

The .0002 in $\times$ .0007 in biradial stylus as used in the BDR-1 broadcast disc reproducer has no performance disadvantages compared to the .0007 in spherical-tip stylus. There are some small advantages to the biradial evident in these data, and others have found some more

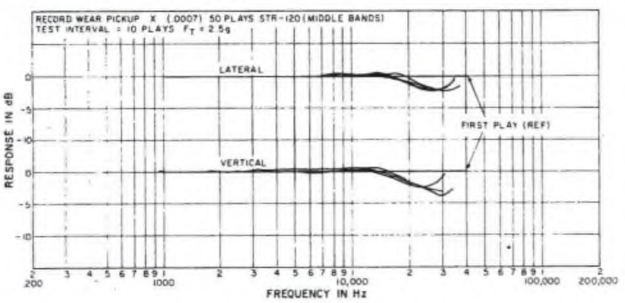

Fig. 12. Record wear with Pickup X, 50 plays.

significant advantages. The biradial stylus appears best for the critical user, but a choice of tip radii allows for record variations and personal preferences. These conclusions are of course not valid for biradial styli of greater effective mass than the type studied.

If the resonance theory is correct, the BDR-1 comes very close to the criteria proposed by Hunt [3] for 1.5 gram tracking. Further work is needed to positively identify the high-frequency resonances, and to explain the mysteries of diameter loss. A standard test for record

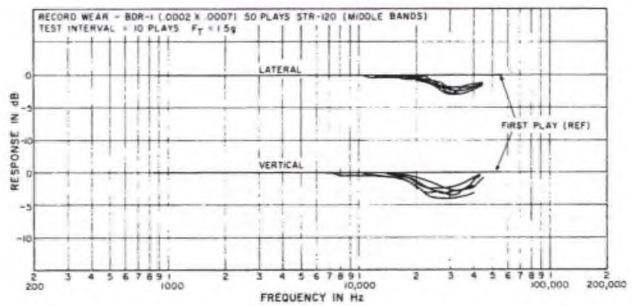

Fig. 13. Record wear with BDR-1, biradial stylus, 50 plays.

life needs to be developed. Beyond this it appears that records exist that have distortion independent of the playback transducer, and the cause and cure remain to be determined.

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2. B. Jacobs, private communication.
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THE AUTHOR

J. R. Sank received the B.S. degree in Electrical Engineering from Drexel Institute of Technology in 1957. After graduation, he joined RCA in Camden, N. J.

As an electroacoustics engineer he has designed and developed various types of microphones, loudspeakers, and phonograph pickups for broadcast and professional audio applications. He has one U.S. patent filed in the microphone field.

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LETTERS TO THE EDITOR

COMMENTS ON "BIRADIAL AND SPHERICAL STYLUS PERFORMANCE IN A BROADCAST DISC REPRODUCER"

D. A. Barlow

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The above paper by Sank[1] calls for a number of comments. The peaks in the response observed on pickup X at 15 kHz and on the BDR-1 pickup at 20 kHz are almost certainly not the resonance of the mass at the stylus tip with the compliance of the record groove. Hunt [1] gives the equation for the stylus–groove resonance as

$$f_0 = 0.6382\, E_v^{1/3}\, r^{1/6}\, M_b^{1/6}\, M_s^{-1/2}$$

where

- E_v plane strain elastic modulus of record material, $= E/(1-\nu^2)$, where E is Young's modulus and ν is Poisson's ratio
- r stylus radius
- M_b playing weight
- M_s tip mass.

Hunt quotes the value of 3.76×10^{10} dyn/cm^2 for E_v, measured on a sample of record material. This agrees with measurements on record material by the writer, and with values quoted by the material manufacturers.

If the writer is correct in guessing the identity of pickup X, it has a tip mass of 1 mg. With a 0.0007-inch radius stylus and a playing weight of 2.5 g, the stylus–groove resonance is calculated as 29 kHz, agreeing with the upper peak at 30 kHz, shown in Sank's Fig. 7.

The accuracy of Hunt's formula is borne out by experience with a mono moving coil pickup. The armature was mounted in torsional rubber bearings, and was extremely stubby, so that there were no bending resonances. Over the years, many thousands of these pickups were made and the response of each one was measured. There was one top resonance only, averaging 19 kHz. The tip mass was 2.5 mg, the playing weight 3.5 g, and the stylus radius 0.001 inch, giving a calculated stylus–groove resonance of 19 kHz. This pickup was also supplied with a 0.0025-inch radius stylus for use on 78 r/min shellac records. On coarse-groove vinyl records, the resonance was about 22 kHz, again agreeing with Hunt's formula.

This formula is derived from the Hertz equation for the elastic range, whereas considerable plastic deformation takes place under any practical pickup. The groove deforms elastically and plastically until it is able to support the load without further plastic deformation; it then behaves elastically and the Hertz equations can then be applied (although not of course for predicting plastic deformation). Correction would need to be made for the lateral curvature of the permanent track made in the groove. The correction would be small, so that Hunt's formula is a close approximation. The value of modulus used is the static one; under dynamic conditions, the modulus will not be less than this, so that the formula will not overestimate the frequency of resonance.

The peak in response of pickup X at 15 kHz is probably due to a bending resonance of the rather slender cantilever. The peak at 20 kHz and subsequent fall in response of pickup BDR-1 are typical in shape of moving-magnet and induced-magnet designs. The peak appears to be the first overtone of the cantilever, simply supported at the stylus and being free or substantially so at the magnet end, as at these frequencies, the rubber mounting will appear very compliant. The fundamental resonant frequency of a simply supported free cantilever is zero. The first overtone would be supersonic in most commercial pickups if it were not for the mass of the magnetic material. The mass of the magnetic material is enough to bring the overtone down into the audio range. This is the chief limitation on the weight of magnetic material, rather than the effective mass of this material at the stylus tip; the effective mass of the magnet is usually much less than that of the cantilever and diamond.

Above this peak, the torsional resonance causes a dip in response, and this is sufficient to obscure the stylus–groove resonance and roll off the higher frequencies. Immediately before the torsional dip, in designs where there is little damping, there is a peak, quite different from the cantilever overtone. There is also the possibility of an electromechanical resonance of the electrical inductance of the coils with the mechanical mass in bending and torsion. To further complicate matters, many commercial pickups have sufficient shunt capacity in contacts and leads to resonate with the coil inductance in the audio range. This capacity prevents the develop-

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[1] J. R. Sank, *J. Audio Eng. Soc.* **18**, 402 (Aug. 1970).

ment of heads with higher output voltage and/or lighter moving parts.

The sag in the response curve of the BDR-1 pickup above 1 kHz is typical of the majority of pickups. It may be offset by a lightly damped resonance higher up, or may be increased due to heavy damping. The sag is due to translation loss, not scanning loss. Translation loss, as defined by Hunt, might better be called differential curvature loss. On a convex surface, i.e., the crests of the recorded modulation, the stylus will penetrate more than on a flat surface for a given load. On a concave surface, i.e., the hollows of the waveform, the stylus will penetrate less. This gives a net loss of amplitude. The effect of the inertia forces due to the tip mass is to accentuate this. In practice, there is considerable plastic deformation and Hunt's formula for translation loss is not valid, although the general trend of results may be similar.

Scanning loss can only occur with curvature overload, i.e., when the radius of curvature of the modulation is less than that of the stylus. If there were no deformation, there would be no loss of peak–peak amplitude until curvature overload (although there would of course be considerable distortion of the waveform). Before this point is reached, the crests will have become of a similar order of size to the stylus, and considerable ploughing and loss of signal will result with any present-day pickup. Scanning loss is therefore of academic interest only at present. On the STR 120 record used by Sank, curvature overload only occurs at the highest frequencies on the inner bands.

The writer hopes to publish more detailed observations at a later date.

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1. F. V. Hunt, "The Rational Design of Phonograph Pickups," *J. Audio Eng. Soc.* **10,** 274 (Oct. 1962).

J. R. Sank

Haddonfield, N. J.

I would like to thank Mr. Barlow for his contribution that appears for the most part to support the data and conclusions in my paper. His conclusions regarding stylus resonance seem to be in agreement with the paper. I hope to have the opportunity in the future to examine and positively identify the suspected stylus–groove resonance of the BDR-1 above the audio range.

Mr. Barlow's detailed explanation of the various types of stylus–groove loss appears to reveal the reasons for the observed responses. It is difficult to tell if he intends to disagree with my remarks in this regard. My remarks seem to be sufficiently general in nature so that there appears to be little room for controversy, save for minor points of interpretation, which is what he may be picking up.

An Experimental Study of Groove Deformation in Phonograph Records

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Groove deformation has been analyzed in the literature primarily in terms of classical elasticity theory, which is based on assumptions that are not appropriate for stylus–groove contact. To determine the actual deformation–force relations, complex groove impedances have been measured as a function of tracking force, groove speed, etc. The results obtained are contrasted with classical predictions.

INTRODUCTION The phonograph has reached its present state of development in spite of the fact that it is still not known how to calculate the forces that act on a stylus while it slides in a record groove. If these forces were known, one could predict the motion of the stylus and thereby gain useful information about the generation of playback distortion. One could also reduce playback distortion without lowering the signal-to-noise ratio by predistorting the recorded modulation so as to complement the nonlinearities of playback. Thus, a prime subject for phonograph research is the force law that describes the force acting on the stylus as a function of the groove deformation. This paper describes an experiment that was performed to determine this force law in the special case of a vibrating stylus sliding in an unmodulated groove.

VERTICAL DEFORMATION OF UNMODULATED GROOVES

Before tackling the general deformation problem for a modulated groove, one should first have a reasonable knowledge of the interaction that occurs when a vertically vibrating stylus slides in an unmodulated groove. This simpler limiting case has much to teach us even though it may appear to be of little practical importance. There are at least two reasons why the unmodulated groove deserves careful study.

First of all, an unmodulated groove provides the simplest possible testing ground for new experimental techniques. Note the phrase *new* techniques: with one exception there do not appear to have been any experiments performed that were intended to expose directly the force relations of phonograph deformation. The one exception was an experiment at Harvard completed by Robert Walkling [1] in 1963 for his doctoral thesis. He measured the mechanical impedance presented to a 1 mil (25 μm) spherical stylus by the surface of an ungrooved 7 in vinyl record rotating only 1 rpm. Therefore, his results do not necessarily apply to the playing of commercial records.

A second reason for studying the deformation of unmodulated grooves becomes clear once it is realized that all existing calculations of playback distortion that take into account groove deformation [2-7], do so by resorting to a force law of questionable applicability that was originally derived by H. Hertz [8-10] for other purposes. Hertz used classical elasticity theory to calculate the stress and strain that are developed when two unstressed, frictionless, paraboloidally shaped, elastic solids are placed in light static contact with each other. The qualifications that the contact be static and that the solids be ideally elastic, paraboloidal, frictionless, and initially unstressed, show that the theory is based on assumptions that are not appropriate to the phonograph, which produces instead a contact that is dynamic, has friction, produces plastic flow, and at high frequencies involves groove geometries that are not paraboloidal. Data on the actual force law of an unmodulated groove are important, therefore, because they provide us with a check on the accuracy of Hertz's force law in an important limiting case.

* Based on a paper presented (by title only in support of the Vietnam Moratorium) Oct. 15, 1969 at the 37th Convention of the Audio Engineering Society, New York.

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An experiment has been conducted to determine the force law for a 0.7 mil (18 μm) spherical stylus that was vibrating vertically while sliding in the unmodulated grooves of special vinyl pressings. To introduce the experimental method, first consider the equation which gives the force law obtained from Hertz's theory:

$$F(t) = Cx^{3/2}(t) \equiv H_H[x(t)], \tag{1}$$

where $F(t)$ represents the vertical deformation force caused by the vertical stylus displacement $x(t)$, and C is a constant that depends on the stylus geometry and the elastic constants of the vinyl. This force law shows no dependence on the velocity of the stylus vibrations $x'(t)$ and can be represented as a function H_H, which we shall call the Hertzian hardness function.

This force law will now be used to show what is meant by the impedance of an unmodulated groove. Suppose the stylus is attached to a transducer that forces the stylus to perform infinitesimal vertical vibrations at a fixed frequency. The stylus displacement can then be represented as the sum of a constant x_0, which corresponds to the average groove indentation, and a complex sinusoidal term with infinitesimal amplitude ϵ: $x(t) = x_0 + \epsilon \exp(j\omega t)$. Using this expression for $x(t)$ in Eq. (1) and expanding in powers of ϵ gives the following series for the force $F(t)$:

$$H_H[x_0] + H'_H[x_0]\epsilon e^{j\omega t} + H''_H[x_0]\epsilon^2 e^{j2\omega t}/2 + \ldots \tag{2}$$

The vertical groove impedance $Z_g(f)$ is defined in terms of the coefficient of the linear term by $j\omega Z_g(f) \equiv$ coefficient of $\epsilon \exp(j\omega t)$. Thus, the impedance in this special Hertzian example turns out to depend simply on the derivative of the Hertzian hardness function: $Z_g(f) = H'_H[x_0]/(j\omega)$.

If the impedance is written in terms of a resistance r_g and a stiffness k_g, $Z_g(f) = r_g + k_g/(j\omega)$, then the resistance is zero, showing that the Hertzian force law does not account for dissipation, and the stiffness is

$$k_g = H'_H[x_0]. \tag{3}$$

The above shows how the force law determines the impedance. It is then appropriate to ask whether one can reverse this process and use the impedance to calculate the force law, that is, whether one can determine the hardness function H_H from the stiffness k_g. The answer is that this is possible, but that one must perform an integration because Eq. (3) is a differential equation.

It is now possible to consider the form of the actual force law. For interpreting the experiment, the reasonable assumption was made that the deformation force is an unknown function η which depends on both the displacement $x(t)$ and the velocity $x'(t)$: $F(t) = \eta[x(t), x'(t)]$. It was arbitrarily assumed that no significant error was being committed by ignoring higher-order derivatives such as the stylus acceleration. This assumption is justified when the ignored derivatives are small enough. Next this expression was simplified by expanding η in powers of $x'(t)$ and retaining only the first two terms. The first term is called the hardness function H, while the second term contains the resistance function R:

$$F(t) = H[x(t)] + R[x(t)]x'(t). \tag{4}$$

The retention of only these first two terms seems justified because the experimental results show that dissipation is usually low in the phonograph interaction. The groove impedance of this force law turns out to give the following resistance and stiffness functions:

$$r_g = R[x_0] \tag{5}$$

$$k_g = H'[x_0]. \tag{6}$$

Note that the groove resistance r_g gives the resistance function R directly, whereas the groove stiffness k_g must be integrated to calculate the hardness function H.

In summary, it has been shown that the assumed nonlinear force law can be deduced from measurements of the vertical groove impedance. This impedance method may seem round-about, but to my knowledge it is the only method that can yield accurate results.

The procedure used for measuring the groove impedance is not limited to phonograph measurements, but can be used to measure the impedance presented by any surface to a vibrating indenter. The procedure appears to be new and has three useful properties: *1*. The real and imaginary parts of the impedance can be continuously plotted with on-line recorders; *2*. The method does not depend on a detailed knowledge of the transducer that vibrates the stylus; *3*. Calibration is a reasonably painless operation.

EXPERIMENTAL STUDY

If the tip of the stylus functions as the mechanical port of a transducer, then the groove impedance can be deduced in principle by measuring the transducer's electrical input admittance. Normally this is not a feasible approach because the input admittance depends on the groove impedance in a complicated way. However, this complicated dependence always reduces to a simple linear dependence when the load impedance is small enough, and the method then becomes attractive.

What constitutes a small enough load (groove impedance)? This question is answered by reference to Fig. 1. Here an arbitrary, linear, time-invariant, physically realizable transducer is represented as a two-port network attached to a complex voltage source E and loaded by mechanical impedance Z_L. The electrical input admittance Y is defined by

$$I \equiv YE, \tag{7}$$

and can always be written in the form

$$Y = A + \{BZ_L/[1 + (Z_L/Z_M)]\}, \tag{8}$$

where A and B are complex functions of frequency that are different for different transducers, but never depend on the load Z_L, and Z_M is the mechanical output impedance of the transducer referred to the tip of the stylus. This equation shows that if the load is much smaller than

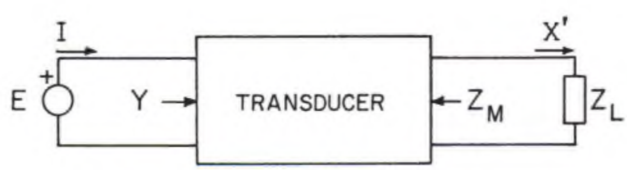

Fig. 1. Transducer represented as a two-port network.

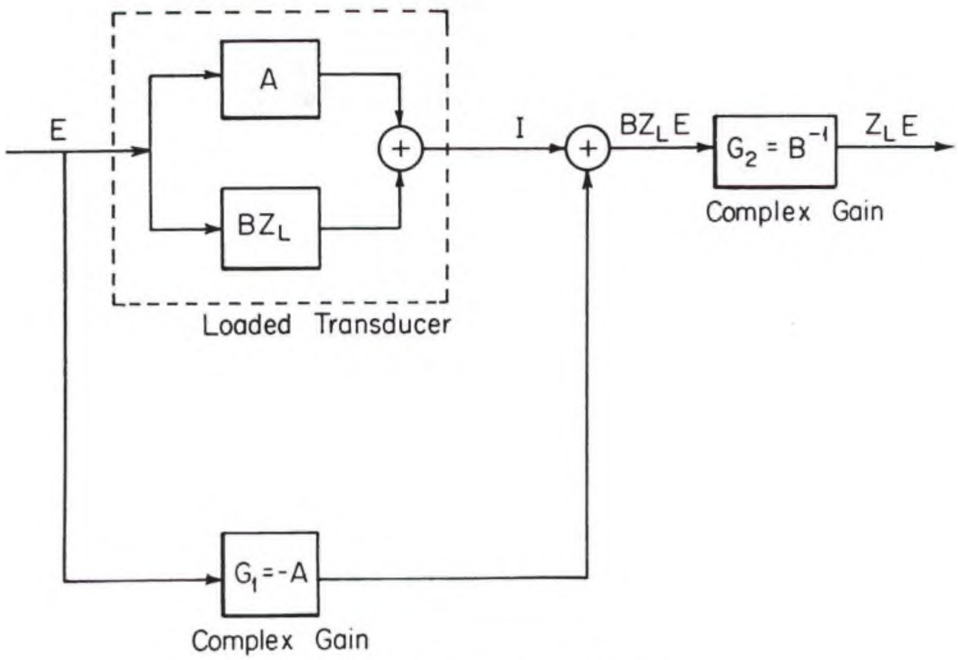

Fig. 2. Isolation of the impedance term.

the output impedance of the transducer, then the input admittance reduces to a simple linear form,

$$Y \cong A + BZ_L, \quad |Z_L| << |Z_M| , \tag{9}$$

where the percent error is of order Z_L/Z_M. Therefore, a groove impedance one hundredth the magnitude of the output impedance will be accounted for with an error of about one percent. A simple means of verifying that this small-load requirement has been satisfied in practice, when $Z_L \equiv Z_g$, will be described later.

Once the input admittance has been forced to depend linearly on the load impedance through the small-load approximation, a simple linear network can be used to determine Z_L from admittance measurements. The first half of a scheme for doing this is shown in Fig. 2.

In this block diagram the gain from input to output depends on four elements; these are the input admittance of the transducer, which is represented by the dotted box, two complex gains, and a summer. As will be shown later, this network can be implemented with operational amplifiers. Note that its output depends in a simple way on the unknown load Z_L. Thus, if the drive voltage is a known sinewave, the impedance Z_L is determined at that frequency.

The complex gains can be set to their proper values without having to know the values of parameters A and B explicitly. First, Z_L is caused to vanish by lifting the stylus out of the groove. The remaining contribution to the output from parameter A is then cancelled by adjusting gain G_1 until the output is observed to vanish. Lastly, gain G_2 is adjusted by first putting a known load on the stylus, such as a small-mass "particle." Such a load constitutes a purely imaginary Z_L. Gain G_2 is then adjusted until the output is also purely imaginary and of convenient amplitude for direct recording. (To simplify matters, the amplitude scaling is not shown here.) The network has now been adjusted to isolate the impedance term even though A and B were not explicitly known.

The proper setting of gain G_1 is rather touchy because this gain cancels out the A factor, which is typically much larger than the impedance term. For instance, during measurement of groove impedances it was found that the A factor of the transducer was as much as 300 times larger than the impedance term. Thus, the setting of gain G_1 was frequently checked during the experiment by simply lifting the stylus out of the groove and observing the output for possible drift in the A factor. In contrast, the B factor of the transducer was surprisingly stable, so that the calibration of gain G_2 would often hold true from one day to the next without readjustment.

Figure 3 shows how the real and imaginary parts of Z_L can be resolved with two synchronous detectors by using the drive voltage as a phase reference. One implementation of a synchronous detector consists of a multiplier followed by a low-pass filter. The input is multiplied by the phase reference signal. The output of the multiplier then consists of a dc component, corresponding to that part of the input that is in phase with the reference, plus extraneous high-frequency components that are blocked by the low-pass filter. Thus, the dc outputs of the in-phase and quadrature detectors correspond to

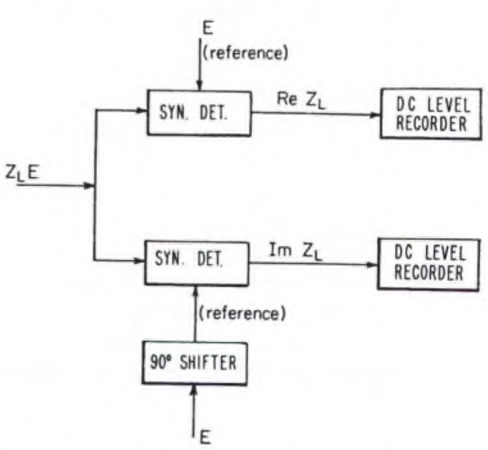

Fig. 3. Synchronous detection of the real and imaginary parts of Z_L.

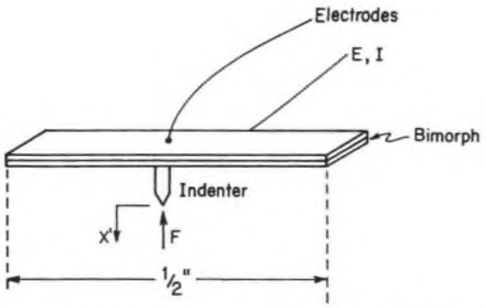

Fig. 4. Piezoelectric transducer for phonograph records.

the real and imaginary parts of Z_L, and can be plotted continuously in real time by dc recorders. Synchronous detection is highly immune to additive noise in the input signal because the effective bandwidth is equal to the frequency passband of the dc recorder—typically 10 Hz.

Since the small-load requirement is basic to this method, it is important to verify that it is satisfied in practice. To see how this can be done, note that if the stylus is clamped so that $|Z_L|>>|Z_M|$, then Eq. (8) implies that

$$Y \cong A + BZ_M, \quad |Z_L|>>|Z_M| . \tag{10}$$

Comparing this equation with Eq. (9), one can see that the output impedance Z_M appears now in the place that is normally occupied by Z_L during impedance measurements. Therefore, the small-load requirement is satisfied if the outputs of the detectors are small when the groove impedance is measured, as compared to the outputs obtained when the stylus is clamped.

Equation (10) also provides a means of choosing a suitable operating frequency. One simply clamps the stylus, measures the magnitude of the input admittance, and varies the frequency until a suitable resonance is observed. For a transducer of sufficiently simple design, this resonance must of necessity correspond to the desired relative maximum in Z_M.

The Transducer

Figure 4 shows the type of transducer that was used. A spherical diamond phonograph stylus with a 0.7 mil (18 μm) tip radius is attached to the center of a piezoelectric bimorph so that it vibrates vertically when the bimorph operates in a free-free bending mode. The stylus of this transducer can be clamped by pushing it against a piece of plate glass. At the top of Fig. 5 the measured admittance of the clamped driver is shown, plotted to logarithmic scales as a function of frequency. The fundamental resonance peak corresponds to a well-defined local maximum in the mechanical impedance Z_M, and therefore determines the operating frequency at 6.32 kHz. When the stylus is lifted off the glass, the admittance changes to that shown in the lower graph. The operating point is far removed from spurious modes and remains so even during groove measurements because the groove impedance, which satisfies the small-load requirement, produces only a small perturbation on the admittance of the free driver.

The transducer was attached to a metal mounting block, which in turn was mounted at one end of a tone arm made of an 8 in soda straw. The other end of the soda straw was attached to a gimbaled support with jeweled bearings that permitted the tone-arm to pivot freely about a fixed point. The gimbal was always positioned for zero average lateral tracking error when data were taken. The average vertical deformation force F_0 was controlled by first adjusting a counterweight for static balance and then unbalancing the structure by placing calibrated weights in a weight pan at the free end of the tone-arm. The mounting block was massive enough to act as an inertial reference for the transducer at the operating frequency of 6.32 kHz, but light enough so that the low-frequency effective mass of the entire system, excluding the tracking-force weights in the weight pan, was about 1¾ g referred to the tip of the stylus. As a result of this low mass, the tone-arm assembly could easily negotiate the slight warpage of the test records to maintain stylus–groove contact even though the tracking force was reduced to less than 5 mg. The transducer, nevertheless, was sturdy enough to permit a maximum tracking force of about 5 g. This range of tracking forces is believed to extend the interaction from virtually perfect elastic deformation to gross plastic flow.

Since the groove impedance is defined in terms of the linearized force law, the amplitude of vertical stylus vibration must be small compared to the average groove indentation to guarantee linear response. To check that this small-amplitude requirement was satisfied in the experiment, the amplitude of stylus vibration was measured with a capacitive proximity probe. The measurements show that the peak vertical amplitude was about 100 Å during the experiment. Since this amplitude is four orders of magnitude smaller than the smallest indentations that were deduced from the data, it appears that the small-amplitude requirement was satisfied at all times.

Calibration

The complex gain G_2 was always adjusted before each set of data was taken. Its calibration usually held true

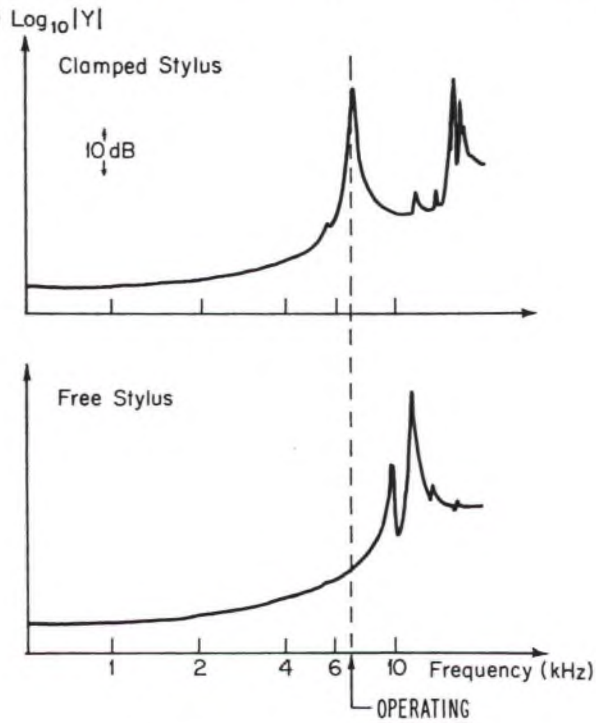

Fig. 5. Measured electrical admittances.

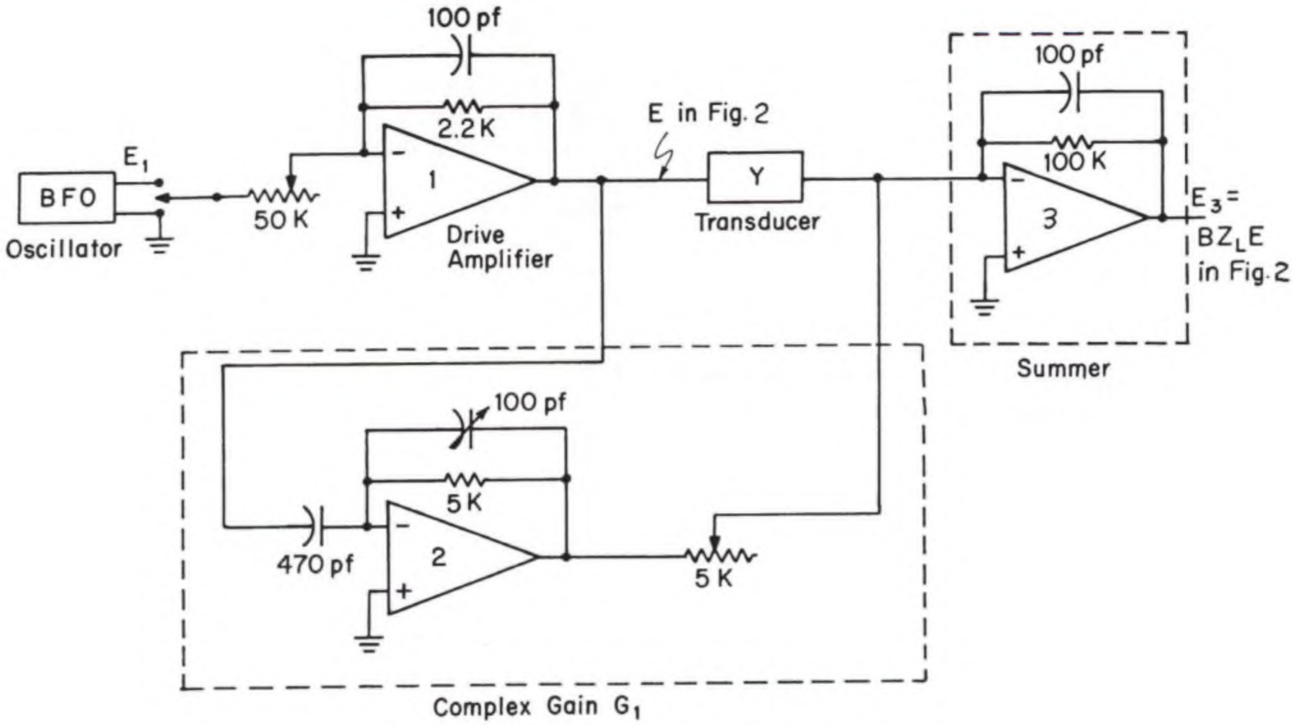

Fig. 6. Realization of the first part of the circuit of Fig. 2.

from one day to the next. As previously described, the proper adjustment of G_2 requires that the stylus be loaded with a known impedance. For this purpose, "particles" of known mass in the form of small cylindrical permanent magnets were used[1] that just fit around the shank of the stylus and firmly fixed themselves by magnetic force to the thin strip of spring steel to which the stylus was attached. Although these magnets did not load the tip of the stylus directly, they did present a purely inertial load that was more or less symmetrically located about the stylus shank and that would require the same motion as the stylus tip if the stylus vibrated without rotation. The symmetry of the transducer was chosen to discourage rotational motion because the vertical groove impedance was to be measured. Furthermore, microscopic examination of the stylus tip with a stroboscopic light source did not reveal spurious rotational motion. One can therefore hope that the use of cylindrical magnets was a justified approximation to a particle of equal mass located at the stylus tip.

1. This method of producing a known load was suggested by Professor F. V. Hunt.

The needed magnets were ground from slices (35 mil thick) of a ceramic magnet taken from a loudspeaker. Typical cylindrical magnets made in this way weighed about 6 mg. Calibration data obtained with these magnets were repeatable; the rms scatter in a batch of twenty measurements was typically less than 5% of the mean.

Circuit Realization

The block diagram in Fig. 2 was implemented by using operational amplifiers. The first half of the system, up to and including the summer, was realized with three operational amplifiers, as shown in Fig. 6. The element values were determined through trial and error. The complex gain $G_1 = -A$ in Fig. 2 was realized by adjusting the feedback capacitor of Amplifier 2 and the series resistor connected to the summing point of Amplifier 3 until the

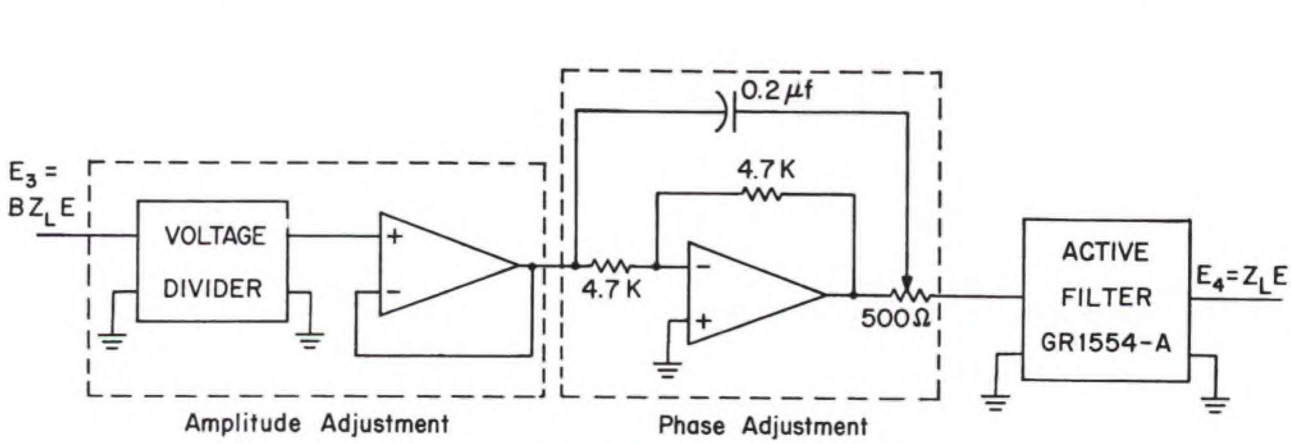

Fig. 7. Realization of the complex gain $G_2 = B^{-1}$ in Fig. 2.

output voltage E_3 vanished when the stylus was unloaded.

The remaining part of Fig. 2, the complex gain G_2, was implemented in three stages, as shown in Fig. 7. The first stage is a voltage divider for controlling the amplitude without introducing phase shift. The operational amplifier prevents the voltage divider from being influenced by the phase circuit. The second stage introduces phase shift without affecting the amplitude, and the third stage is a General Radio active filter (⅓-octave bandwidth) that provides amplification and noise rejection.

The last part of the electronics, corresponding to Fig. 3, included an Ad-Yu vector resolver for synchronous detection and two General Radio 1521A graphic level recorders. Two phase-reference voltages for the synchronous detectors, E_1 and E_2, were generated as shown in Fig. 8. The 500 Ω resistor was adjusted until the phase difference between E_1 and E_2 was 90°, as indicated by using E_1 as input and E_2 as phase reference to one of the synchronous detectors. The accuracy of this method for adjusting phase was confirmed by repeating the procedure with the roles of E_1 and E_2 reversed. (Both the inherent phase error of the detector and the phase error of the reference signals can be checked by this method of reversing the roles of E_1 and E_2.) The data of this experiment showed that the inherent phase errors of the Ad-Yu resolver were below the manufacturer's rating of 2% of full scale.

Test Records

The test records were specially made for this study.[2] The vinyl pressings were made from a standard antistatic mix ten months before the date of the experiment. Each record had four bands of unmodulated grooves. Only the outermost and innermost bands were used. The average radii of these bands were 13 cm and 8.5 cm. The groove geometry of a typical record was measured by putting some RTV-102 silicone rubber on a grooved area of the record and then carefully peeling it off after it had set, to produce a positive replica of the record topography. A thin slice of this replica, when viewed under a calibrated microscope, indicated that the groove cross section had the dimensions shown in Fig. 9. Simple calculations based on this geometry show that a spherical stylus with a tip radius of 0.7 mil (18 μm) will touch the bottom of the groove if the normal groovewall indentation exceeds 3.7 μm. Since the experimental results show that this indentation is more than 3.5 times the maximum indentation produced during the experiment, it appears that the data are free of groove-bottoming effects.

2. The records were kindly donated by RCA. J. G. Woodward cut the master disc, and the pressings were produced by R. C. Moyer.

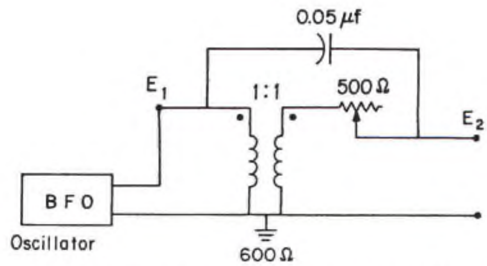

Fig. 8. Generation of two phase-reference voltages for the synchronous detectors.

Experimental Results

Figure 10 shows a typical run of data produced by the dc level recorders when the tracking force was 0.1 g. The top curve is the groove stiffness as a function of time, while the bottom curve is the corresponding groove resistance. These curves show a pulse-type waveform because after every few revolutions of the record the stylus was lifted out of the groove to check the zero level of the set-up (adjustment of complex gain G_1). The irregularity in these curves is typical of the data obtained, although sometimes the variability was either much greater or much less. Apparently, contaminating dirt on the stylus can affect the results. Each time the tracking force was changed, the stylus was cleaned by rubbing with a dab of RTV silicone rubber on the end of a wooden stick. This resilient material catches and holds dirt effectively. The temperature was held between 24° and 26° C.

The resistance data is more random than the stiffness data. This extra noise appears because a larger gain was used. Most of it is due to surface noise on the record and does not represent actual resistance fluctuations. These, as well as all other data, were obtained with the stylus performing 6.32 kHz vertical oscillations with a peak amplitude of about 100 Å. If the transducer had been driven harder to produce larger oscillations, then contamination due to surface noise would have been reduced. It was not possible to take advantage of this fact, however, because with larger amplitudes the transducer showed nonlinear behavior that could have invalidated the

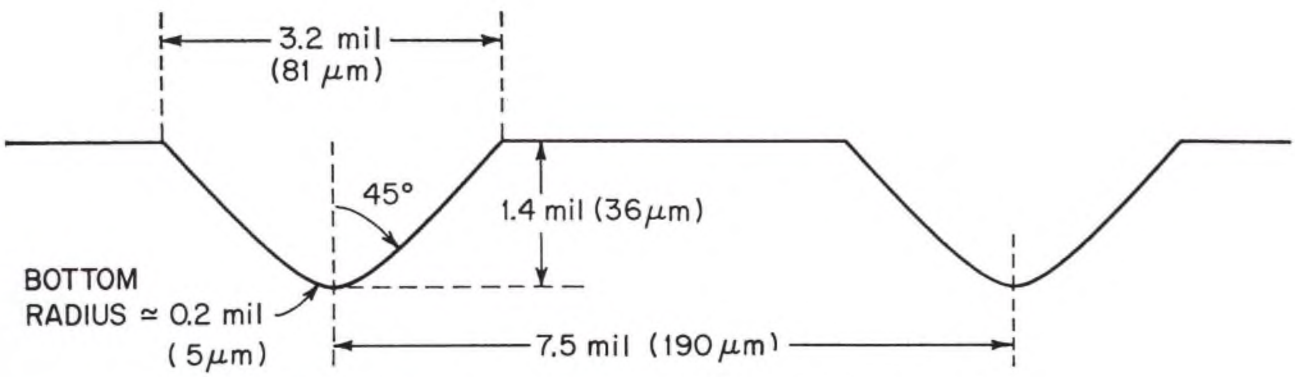

Fig. 9. Observed groove cross section of a test record (not to scale).

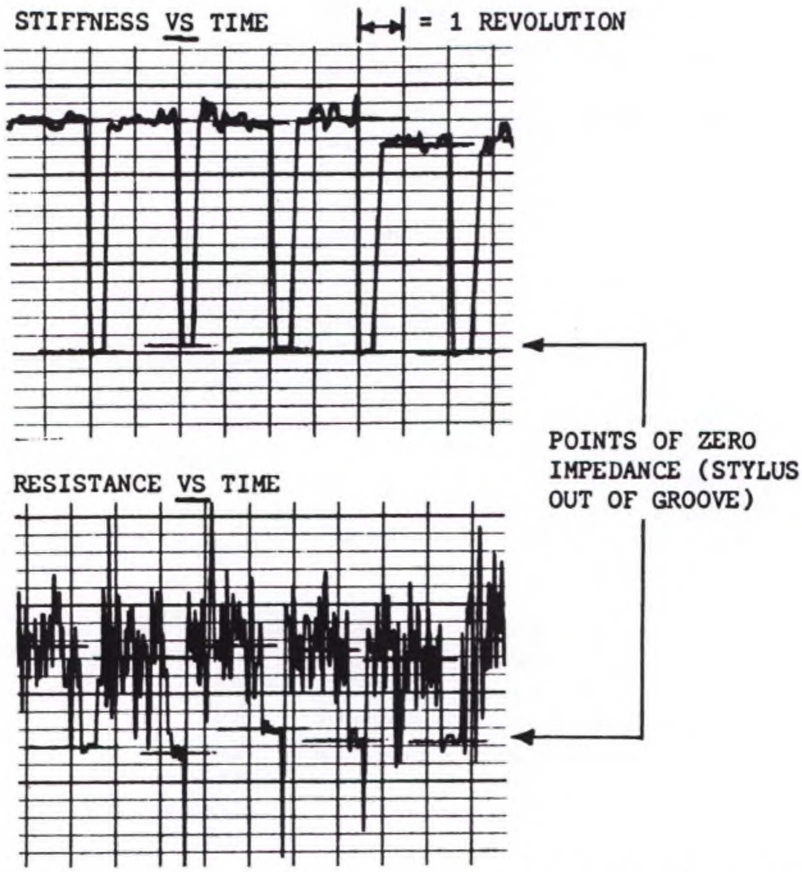

Fig. 10. Typical data from dc level recorders. Horizontal bars denote average values. Tracking force = 0.1 g, groove velocity = 0.30 m/s.

experimental method. The surface noise is not a serious problem in this case, however, because the data must be averaged to obtain the average force law for the record.

Figure 11 shows average stiffness values obtained from a single record and plotted to logarithmic scales. The straight lines are the averages of straight-line fits to all the stiffness data obtained. For comparison with classical elasticity theory, the Hertzian stiffness has also been plotted by choosing arbitrary but reasonable values for the elastic constants of the vinyl (Young's modulus = 3.3×10^9 N/m^2, Poisson's ratio = 0.35). Each data point is the average of ten or more "pulses" of the type shown in Fig. 10. These data include tracking forces that range from 4.6 mg to 4.6 g. This range is believed to extend the interaction from virtually perfect elastic deformation to gross plastic flow.

The most startling feature of these data is that at very low tracking forces, where significant plastic flow is unlikely, the stiffness fails to decrease with the slope of ⅓ that elasticity theory predicts. Perhaps this unexpected behavior arises because the real area of contact between stylus and groove is not a simply-connected area as elasticity theory assumes, but instead is made up of minute separated regions due to the microscopic roughness of the two surfaces. Another possibility is that the vinyl has voids in it, as steel wool does, that make the groove walls unexpectedly soft for small forces.

These data were obtained by starting at the lowest tracking force and working up to the highest. Thus, each data point gives the average stiffness of grooves that had been played previously at forces that were equal to or lower than the current value, but not higher. These might therefore be called the stiffnesses of relatively virgin grooves. An obvious question now arises: what stiffness curve is obtained if the record is replayed, once again starting at 4.6 mg and working up? Again the answer is a surprise: the measured stiffnesses are the same as for relatively virgin grooves, to within experimental variation. Apparently the grooves do not "work harden" even when played at 4.6 g. Measurements taken on both inside and outside grooves of the test records showed that *the groove stiffness does not depend on the groove velocity, whereas the resistance* r_g *appears to be inversely proportional to this velocity.*

Figure 12 shows average resistance values for the inside grooves of a single record (groove velocity = 0.30 m/s). The straight line is the average of the straight-line fits for all the resistance data obtained.

Figure 12 makes it clear that losses in the phonograph interaction are small. Note that at 1 g force the average resistance is 0.1 mks Ω, which is only 0.1 of the value of the corresponding reactance due to the groove stiffness. Since the frequency was 6.32 kHz, these results show that the reactance will not drop to equal the resistance until the frequency is raised to 60 kHz, provided that the stiffness and resistance are independent of frequency. The phonograph interaction, therefore, appears to be one of low loss at audio frequencies.

Finally, Fig. 13 shows the experimental hardness function H obtained by integrating the three straight lines

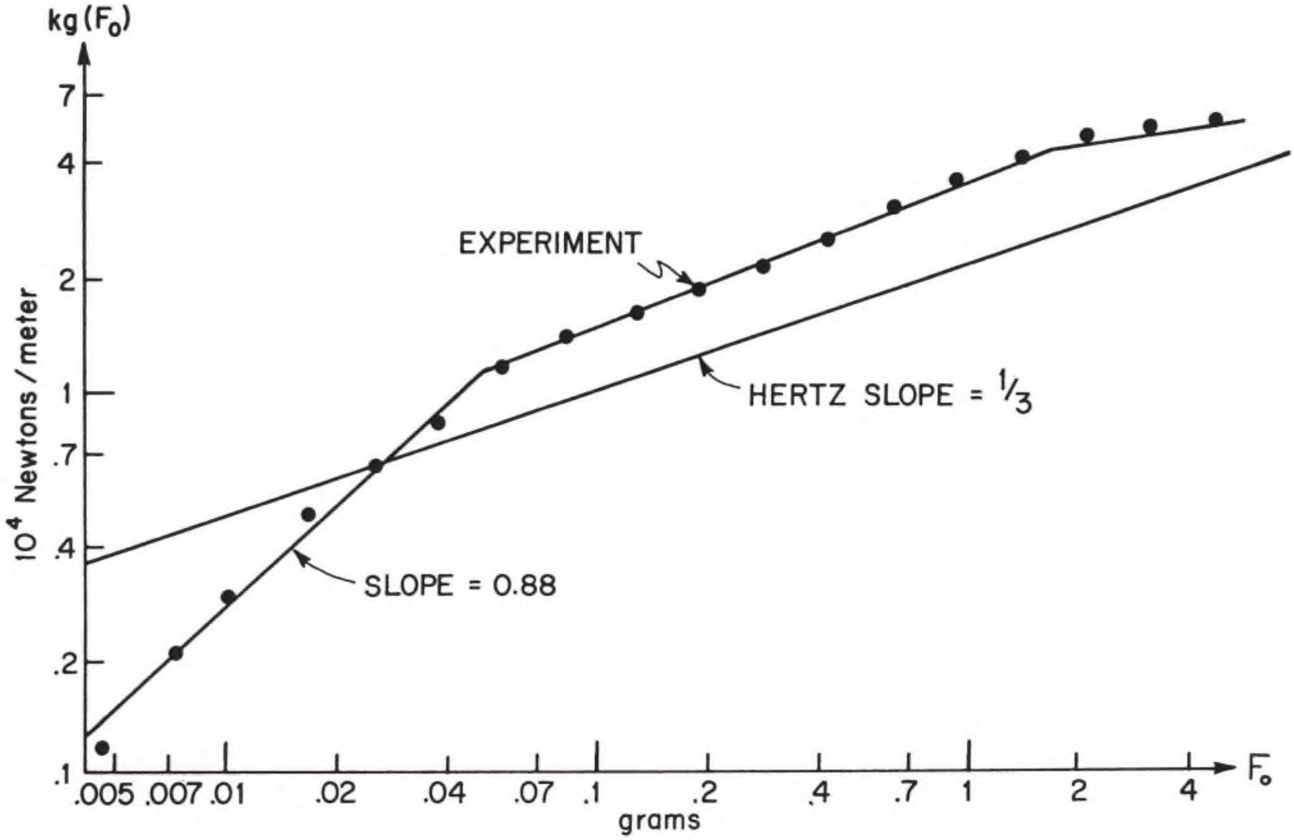

Fig. 11. Stiffness *vs* tracking force.

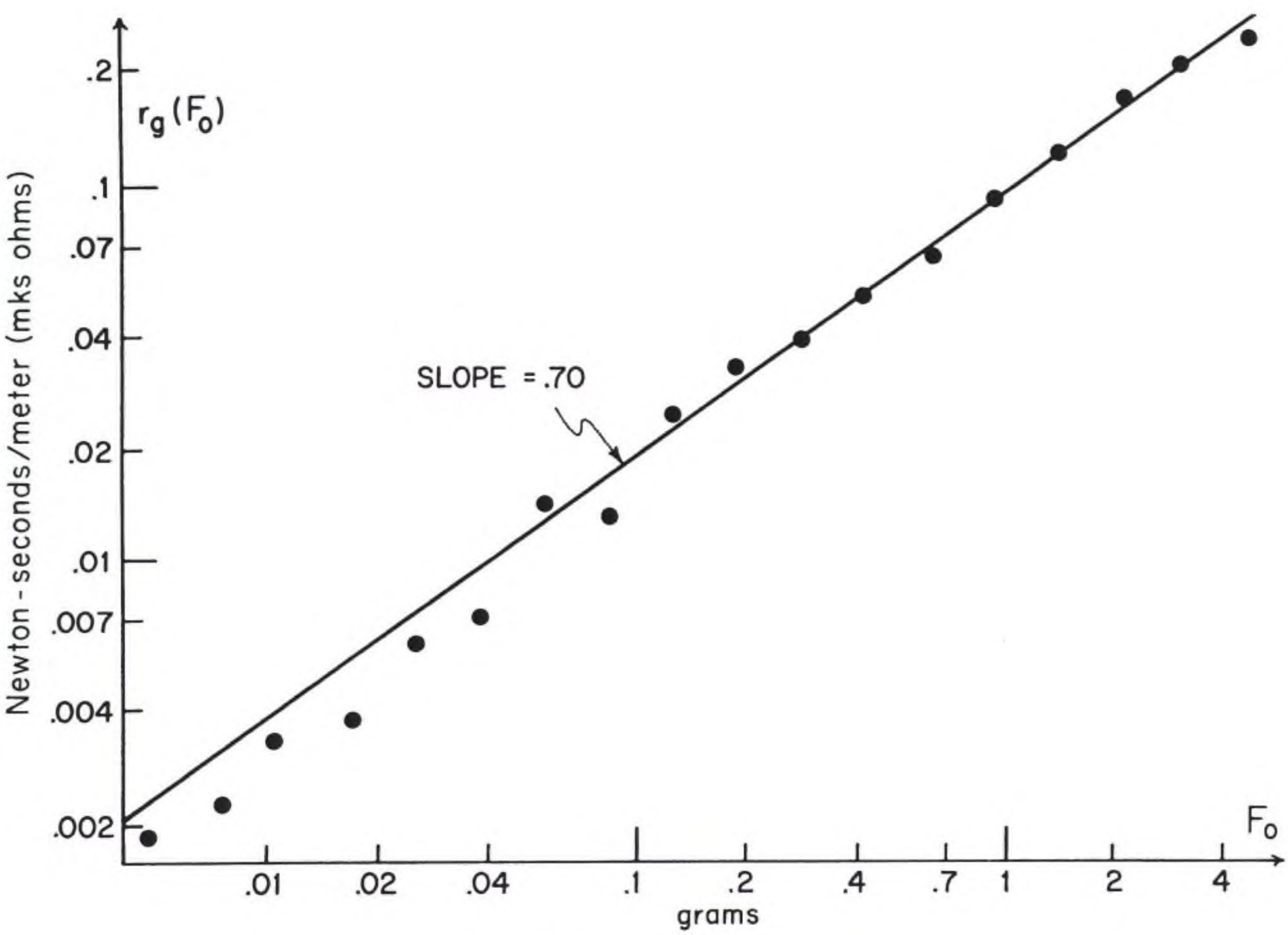

Fig. 12. Resistance *vs* tracking force.

that fit the stiffness data. This experimental hardness equals the hardness predicted by Hertz's theory at a force of about 1.5 g. Below this force the actual hardness becomes significantly smaller than the Hertzian prediction, while above it the actual hardness is greater. The slope of the Hertzian curve is fixed and is independent of the assumed values of the elastic constants of the vinyl. Its vertical location, however, does depend on the arbitrary values assumed for these constants, which in the present case were Young's modulus $= 0.33$ g/μm^2 and Poisson's ratio $= 0.35$. The dotted arrow shows the boundaries within which the Hertzian curve can be expected to lie, assuming that the Young's modulus is greater than 0.15 but less than 0.6 g/μm^2. These results show that *the Hertzian force law underestimates the amount of groove deformation that occurs at low forces.*

There is more than one way to calculate the hardness function from the stiffness data. For instance, one can integrate the stiffness curve upwards from zero tracking force, or one can integrate the curve downward from some large force for which the stylus displacement is known from another experiment. Both methods were used, and resulting hardness functions were found to be identical.

To integrate upwards from zero tracking force it was arbitrarily assumed that the data, which extended only down to 4.6 mg, could be extrapolated down to zero force by simply extending the straight-line average that was obtained on the log-log stiffness curve. That is, it was assumed that the stiffness was proportional to the 0.88 power of the tracking force in the lowest range of forces.

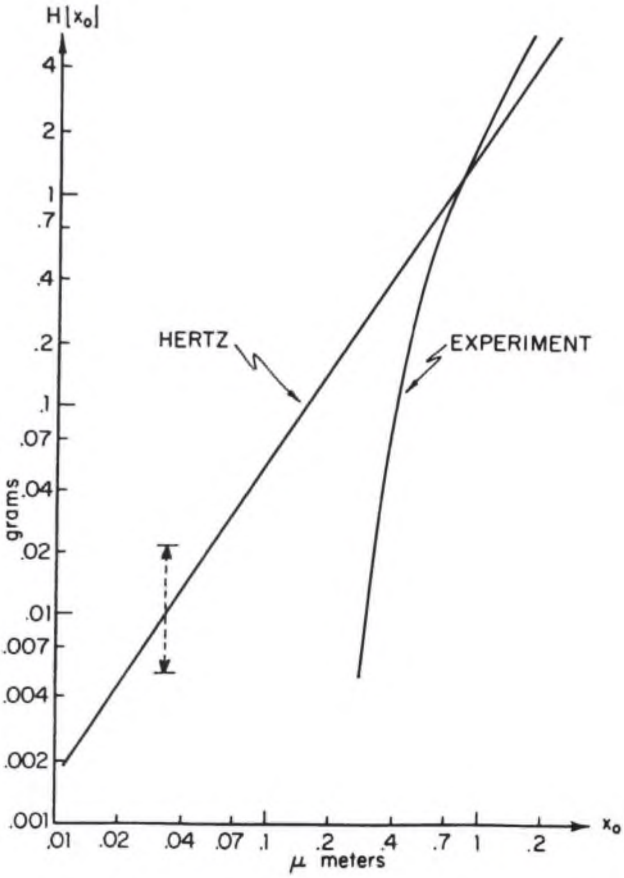

Fig. 13. Theoretical (Hertzian) and experimental hardness functions.

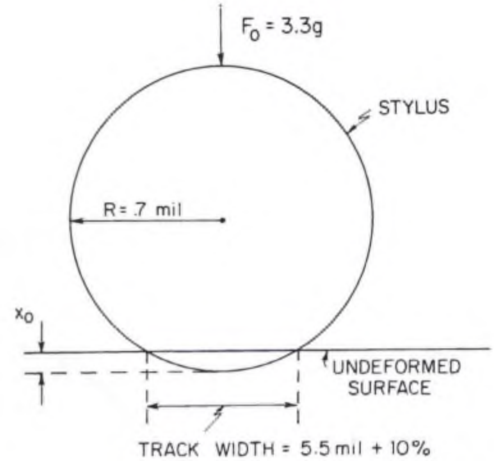

Fig. 14. Contact geometry for a stylus on a flat surface.

The other method of determining the hardness function requires a knowledge of the stylus displacement at a known tracking force. This displacement was estimated by measuring the width of the indentation track that was left by the stylus when it was allowed to rest on the periphery of the ungrooved center of a test record rotating at 33.3 rpm. With a tracking force of 3.3 g the track was clearly defined so that its width could be measured to within 5% by using a calibrated microscope. The average width produced by the 0.7 mil spherical stylus was 0.55 mil (the track width varied from 0.5 to 0.6 mil around the record). It was assumed that elastic recovery had reduced the width by 10%. It was also assumed that the stylus position relative to the undeformed surface could be estimated from the diagram in Fig. 14. A simple calculation based on this geometry shows that $x_0 = 0.08$ mil. This result must be modified for the geometry of a phonograph groove by noting that 3.3 g on a horizontal surface corresponds to $F_0 = 4.8$ g with two supporting surfaces, each 45° with respect to the vertical. Similarly, a 0.08 mil groovewall deformation corresponds to a vertical stylus displacement of $x_0 = 0.11$ mil. These values were used to integrate the averaged stiffness data downward from 4.8 g and gave results identical to the upward integration, which was based on extrapolating stiffness data down to zero force. In view of this agreement between two independent methods, *it is probably justified to conclude that groove deformation is incorrectly described by classical elasticity theory even at the lowest tracking forces for which no data were taken.*

Some formulas that describe the experimental straight-line curves shown in Figs. 11 and 12 are listed below. All quantities are expressed here in mks units except for the deformation force F_0, which is given in units of gram force (1 g $\simeq 10^{-2}$N).

$$k_g = 1.6 \times 10^5 F_0^{0.88} \quad 4.6\,\text{mg} \leq F_0 \leq 50\,\text{mg} \tag{11}$$

$$k_g = 3.6 \times 10^4 F_0^{0.38} \quad 0.05\,\text{g} \leq F_0 \leq 1.6\,\text{g} \tag{12}$$

$$k_g = 4.0 \times 10^4 F_0^{0.16} \quad 1.6\,\text{g} \leq F_0 \leq 4.6\,\text{g} \tag{13}$$

$$r_g = \frac{0.33}{v} F_0^{0.70} \quad \begin{array}{l} 4.6\,\text{mg} \leq F_0 \leq 4.6\,\text{g} \\ 0.30\,\text{m/s} \leq v \leq 0.45\,\text{m/s}. \end{array} \tag{14}$$

where v = groove velocity. The experimental curve for the hardness function H in Fig. 13 is described by the equation

$$H(x_0) = (1.3 \times 10^5 x_0 - 3.8 \times 10^{-2})^{1.6} = F_0. \quad (15)$$

This expression is exact only for $0.05 \text{ g} \leq F_0 \leq 1.6 \text{ g}$, but it can be used for the larger range of 0.01 g to 4.6 g because the resulting maximum error is less than 7%. The inverse hardness function corresponding to Eq. (15) is

$$x_0 = H^{-1}(F_0) = (7.6F_0^{0.62} + 0.29) \times 10^{-6}. \quad (16)$$

SUMMARY

1. It is still not known how to calculate the forces that act on a stylus while it slides in a *modulated* groove.

2. To determine the force law that determines these forces one should first study the case of an unmodulated groove because such grooves not only provide the simplest possible testing ground for new experimental techniques but also provide an important limiting case for checking the accuracy of proposed deformation theories.

3. The actual force law for the vertical component of force in an unmodulated groove can be deduced from measurements of the complex vertical groove impedance by making the reasonable assumption that the deformation force in an unmodulated groove depends nonlinearly on stylus displacement and linearly on stylus velocity.

4. The needed measurements of groove impedance can be obtained by using the stylus as the mechanical port of a transducer and choosing the operating frequency so that the mechanical driving-point impedance referred to the stylus tip is much larger than the groove impedance being measured.

5. Experimental results obtained by this method at 6.32 kHz with a 0.7 mil stylus showed that losses were small at audio frequencies and that the resistance function was inversely proportional to the groove velocity in the range from 0.30 to 0.45 m/s. The hardness function, on the other hand, was independent of the groove velocity.

6. Experimental results also showed that Hertz's theory incorrectly describes the groove deformation even when the tracking force is small enough to preclude macroscopic plastic deformation. The grooves were softer at small tracking forces, and harder at high tracking forces, than Hertz's theory predicts.

ACKNOWLEDGMENTS

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James V. White was born in Hammond, Indiana in 1941. He received a BS in electrical engineering from Northwestern University in 1964, an SM in engineering in 1965 from Harvard University and a Ph.D. in engineering in 1970, also from Harvard, for which his major field was acoustics and signal processing.

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PROJECT NOTES/ENGINEERING BRIEFS

LACQUER WARP, ADVANCE BALL, AND DISC CUTTER DYNAMICS*

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The disc cutter head, with associated springs, dashpot, advance ball, and lacquer disc, is modeled by an equivalent circuit. Numerical values for this model are deduced from experimental data for a Westrex 3DII cutter mounted on a Scully lathe. The model is then used to calculate the audio frequency response and the rejection of unwanted groove modulation caused by lacquer warp. The results obtained with and without the advance ball are compared with each other.

INTRODUCTION: A cutting head mounted on a disc-mastering lathe is a fundamental manufacturing instrument for the music industry. Disc record music begins here—the master tape program is placed in its final format, a grooved disc. This process is to take place in the most efficacious manner, having minimal alterations to the audio program.

All modern disc-mastering systems mechanically suspend the cutting head above a rotating record blank (lacquer disc), permitting the stylus to groove the disc. This mechanical assembly influences the groove modulation. If the lacquer disc has surface irregularities (warp), the pressed vinyl record will contain these irregularities as unwanted modulation. Some disc cutter systems utilize an advance ball device that rides on the lacquer disc and mechanically transmits surface irregularities to the cutter head. This reduces the unwanted groove modulation because the cutter tracks the warp. As a result, quiet grooves can be cut with almost constant depth even if the lacquer surface is not flat.

The purpose of this paper is to portray the effects on the recorded low-frequency vertical modulation caused both by the disc cutter head assembly and mounts, and the movement of the lacquer surface caused by warp or rumble. We also show the effects of utilizing an advance ball assembly.

In order to study the head assembly system, a model is developed for the vertical dynamics of the cutting head and advance ball resting on the lacquer surface. The model corresponds to an electrical analog circuit, from which the equations of motion are derived (valid for low frequencies). The value of each circuit parameter is deduced experimentally for a Westrex disc cutter, type 3DII, mounted on a Scully lathe carriage assembly. Next, numerical solutions to the circuit equations are displayed in familiar terms of decibels versus frequency; these show the influence of the mounting system on the recorded frequency response, the sensitivity to lacquer warp, and the changes caused by the advance ball for the specific system under study.

Three assumptions are made in the paper. First, we treat the disc cutter transducer as ideal, that is, it has infinite feedback for all (low-frequency) signals. Second, the lathe carriage and main frame are both rigid and stationary. Third, the lacquer warp or rumble appearing at the stylus tip has the same waveform as that appearing at the advance ball.

CUTTER ASSEMBLY MODEL

Fig 1 is an outline drawing of the Westrex cutter head and mounting assembly. The cutter head consists of a sizable magnet, associated transducer elements, and a frame. Its motion is constrained to move vertically by the pivot so that it may be lowered and raised for lacquer cutting. The cutter hangs on the spring after lowering, and the dashpot dampens the vibration of the spring and head mass. The advance ball assembly rests on the lacquer surface and is adjusted to supply an upward force on the cutter so that in equilibrium the desired groove width is cut.

Examination of this assembly leads to the dynamic model shown in Fig. 2. The lathe frame is chosen as the inertial reference. The main spring has the stiffness K_m [N/m], and the dashpot has resistance B [N·s/m]; both connect directly to the cutter frame mass M [kg]. Also connected to mass M is the stiffness of the advance ball lever K_a [N/m] in series with the lacquer stiffness K_l [N/m] presented to the advance ball. The cutting stylus works into a lacquer stiffness K_g [N/m].

The vertical audio signal controls the vertical velocity of the stylus with respect to the mass M; this velocity is represented by the velocity source V_s. (The cutter is assumed to be ideal, i.e., it has enough feedback so that V_s is proportional to the audio signal.) Lacquer warp, or surface irregularity, is represented by the velocity source V_w, referenced to the lathe frame. (Due to the close proximity of the stylus and advance ball, it is reasonable to assume that both receive the same warp waveform.)

The velocity v_g with which the groove is vertically modulated corresponds to the velocity with which spring K_g is compressed. Raising the advance ball is modeled by disconnecting springs K_a and K_l from each other.

CUTTER ANALOGY

Fig. 3 shows a force-current analog circuit [1]–[3] of the disc cutter model of Fig. 2. Each mechanical component is labeled by its network admittance; the mechanical nodes

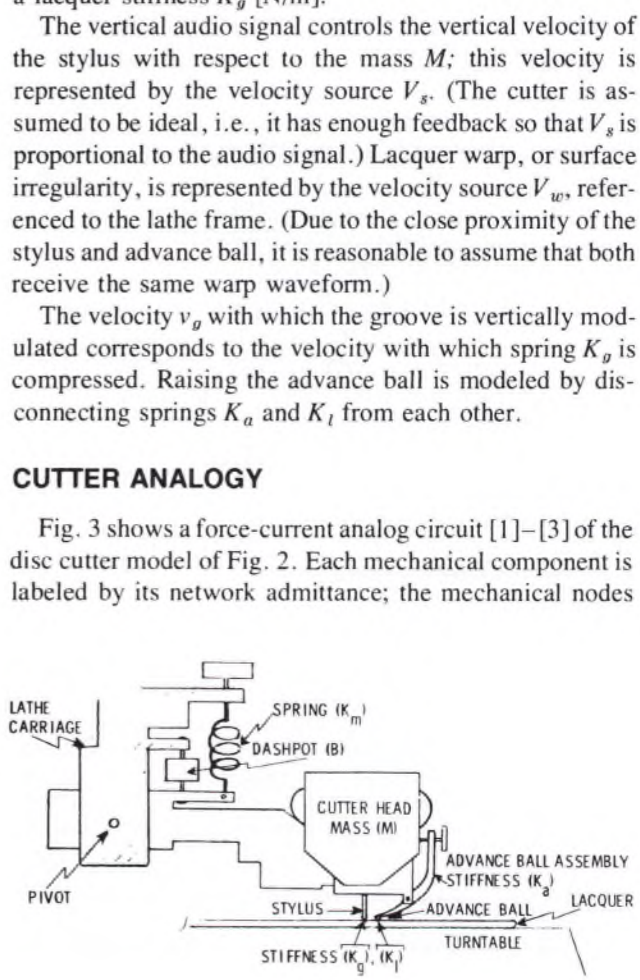

Fig. 1. Outline of Westrex disc cutter head and mounting system.

Presented October 31, 1975, at the 52nd Convention of the Audio Engineering Society, New York.

and loops remain as electrical nodes and loops; velocities are defined as voltages, and forces as currents. Note that network admittance is defined as current/voltage ≡ through quantity/across quantity ≡ force/velocity. Thus the network admittances are also mechanical impedances.

We wish to calculate groove modulation v_g as a function of stylus velocity V_s and warp velocity V_w. This analysis may be simplified by combining the admittances of the three branches:

$$Y_a = \frac{K_m}{s} + sM + B \tag{1}$$

$$Y_b = \frac{K_a \cdot K_l}{K_a + K_l}\,\frac{1}{s} \equiv \frac{K_b}{s} \tag{2}$$

$$Y_c = \frac{K_g}{s}\,. \tag{3}$$

The simplified model is shown in Fig. 4.

If the advance ball is *not* used, then the groove modulation velocity may be written by inspection of Fig. 4 as

$$v_g = (V_s + V_w)\,\frac{Y_a}{Y_a + Y_c} \equiv (V_s + V_w)\,A\,(s). \tag{4}$$

Similarly, if the advance ball *is* used, then

$$v_g = V_s\,\frac{Y_a + Y_b}{Y_a + Y_b + Y_c} + V_w\,\frac{Y_a}{Y_a + Y_b + Y_c}$$

$$\equiv V_s B\,(s) + V_w C\,(s). \tag{5}$$

Eq. (4) and (5) describe how the groove modulation depends on the cutter head mechanical assembly, lacquer warp, and stylus velocity. We need only to substitute the admittance values for a practical cutting system in order to plot the frequency responses in the groove for the audio input V_s and the warp input V_w.

FREQUENCY RESPONSE PLOT TECHNIQUE

To obtain the frequency responses we express each of the transfer functions $A\,(s)$, $B\,(s)$, and $C\,(s)$ in the standard form:

$$\frac{s^2 + 2\zeta_1\omega_1 s + \omega_1^2}{s^2 + 2\zeta_2\omega_2 s + \omega_2^2}.$$

This standard form corresponds to a frequency response curve that has 0 dB gain at high frequencies, 20 log $[\omega_1/\omega_2]^2$ dB at low frequencies, and an asymptote of 12 dB per octave between ω_1 and ω_2. The gain deviates from the asymptote at the corner frequencies ω_1 and ω_2 by amounts of 20 log $(2\zeta_1)$ and -20 log $(2\zeta_2)$.

FREQUENCY RESPONSE PLOTS FOR WESTREX 3DII CUTTER MOUNTED ON SCULLY LATHE

For our specific cutting system the following frequency response curves were obtained. The mechanical parameter values and the method used to measure them are given in the Appendix.

The solid-line curve of Fig. 5 shows the calculated frequency response (20 log$|A(j\omega)|$) of the ideal cutter without the advance ball. The curve was constructed from the following data, which were taken from the Appendix:

$$\omega_1 = \left(\frac{K_m}{M}\right)^{\frac{1}{2}} = 2\pi\ (5.9)\quad \mathrm{s}^{-1}$$

$$\omega_2 = \left(\frac{K_m + K_g}{M}\right)^{\frac{1}{2}} = 2\pi\ (16.5)\quad \mathrm{s}^{-1}$$

$$\text{low frequency gain} = 20\log\frac{K_m}{K_m + K_g} = -17.9\ \mathrm{dB}$$

$$\text{gain deviation at } f_1 \equiv 20\log(2\zeta_1) > 8.3\ \mathrm{dB}$$

$$\text{gain deviation at } f_2 \equiv -20\log(2\zeta_2) < 0.7\ \mathrm{dB}.$$

We observe that the frequency response is flat down to 16.5 Hz and has an asymptotic loss of -18 dB at very low frequencies. This characteristic is due to the cutter head mass, the main spring and damper, and the lacquer stiffness

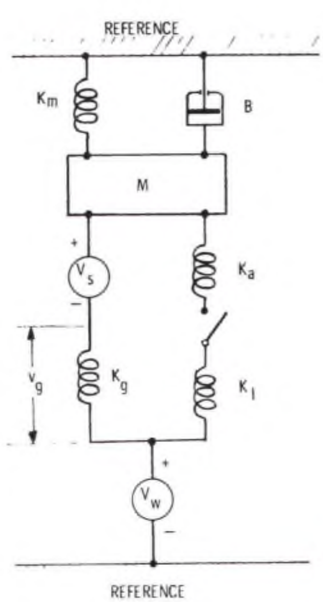

Fig. 2. Model of disc cutter assembly of Fig. 1.

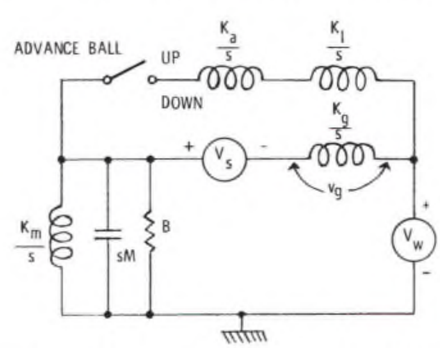

Fig. 3. Analog circuit for disc cutter model of Fig. 2.

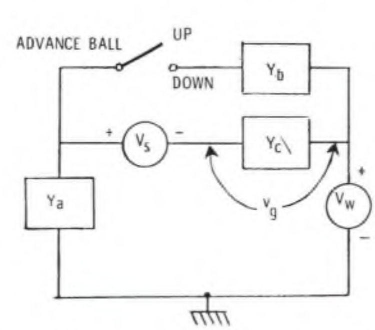

Fig. 4. Simplified version of analog circuit of Fig. 3.

presented to the stylus. Such performance is certainly adequate for audio recording.

Next we examine how the frequency response changes when the advance ball is properly resting on the lacquer surface. The dashed curve of Fig. 5 shows the calculated frequency response ($20 \log|B(j\omega)|$) with the advance ball in operation. The curve was constructed with the following date from the Appendix:

$$\omega_1 = \left(\frac{K_m + K_b}{M}\right)^{\frac{1}{2}} = 2\pi \ (44) \quad s^{-1}$$

$$\omega_2 = \left(\frac{K_m + K_g + K_b}{M}\right)^{\frac{1}{2}} = 2\pi \ (46) \quad s^{-1}$$

$$\text{low frequency gain} = 20 \log \frac{K_m + K_b}{K_m + K_b + K_g} = -1 \text{ dB}.$$

Utilizing the advance ball mechanism has greatly extended the low frequency of the cutter (which is not needed for audio recording). It has also placed a 1-dB drop in the spectrum below 44 Hz. (This drop may be equalized out so that only a ripple exists in the frequency response curve.) The drop occurs because the total advance ball stiffness K_b is not infinitely stiff. If the advance ball mechanism were more compliant, then the frequency response curve would be in between the two curves in Fig. 5, yielding significant audio-frequency losses below 40 Hz.

The groove modulation susceptibility to lacquer warp is also given by $20 \log|A(j\omega)|$ when the advance ball is not employed. This is shown in Fig. 6 by the solid curve. Warp velocities that have very low frequencies below 5 Hz are attenuated almost 18 dB, whereas those above 16.5 Hz are recorded into the groove as modulation with no attenuation.

The dashed curve of Fig. 6 shows the groove modulation susceptibility to lacquer warp when the advance ball is used. This is a plot of $20 \log|C(j\omega)|$ with the following parameter values:

$$\omega_1 = \left(\frac{K_m}{M}\right)^{\frac{1}{2}} = 2\pi \ (5.9) \quad s^{-1}$$

$$\omega_2 = \left(\frac{K_m + K_b + K_g}{M}\right)^{\frac{1}{2}} = 2\pi \ (46) \quad s^{-1}$$

$$\text{low frequency gain} = 20 \log \frac{K_m}{K_m + K_b + K_g} = -38 \text{ dB}$$

$$\text{gain deviation at } f_1 \equiv 20 \log (2\zeta_1) > 8.3 \text{ dB}$$

$$\text{gain deviation at } f_2 \equiv -20 \log (2\zeta_2) < 9.6 \text{ dB}.$$

This curve shows the effectiveness of the advance ball in attenuating lacquer warp. For the very low frequencies the rejection is 38 dB. As the frequency of the lacquer warp increases, the rejection diminishes to 0 dB above 46 Hz. Comparing the solid and dashed curves, we observe a great improvement in warp rejection when the advance ball is used (except for a narrow band of frequencies near 50 Hz).

CONCLUDING REMARKS

The vertical dynamics of a cutter head assembly have been modeled and represented by an analog circuit (valid for low frequencies). Numerical values for the circuit parameters were determined for a Westrex 3DII cutter and Scully lathe by a series of four experiments.

The circuit contains two voltage sources; one represents the audio input and the other (called the warp input) represents surface irregularities of the lacquer disc or turntable rumble. The response of the groove modulation to each of these two inputs was calculated as a function of frequency; the responses with and without the advance ball were compared to each other. These calculations showed that *without* the advance ball there is 18-dB rejection of warp inputs at frequencies far below 6 Hz. In contrast, *with* the advance ball properly adjusted the rejection increases to 38 dB and remains better than 18 dB up to 15 Hz.

The audio frequency response was adequate for normal recording both with and without the advance ball. However, calculations did show that an advance ball with too little stiffness (because of improper use or design) can degrade the frequency response below 40 Hz.

In closing, we would like to point out that the analog circuit presented in this paper can be used to answer additional questions about cutter performance that we have not discussed. For example, what are the forces exerted by the advance ball on the lacquer due to both the audio and warp

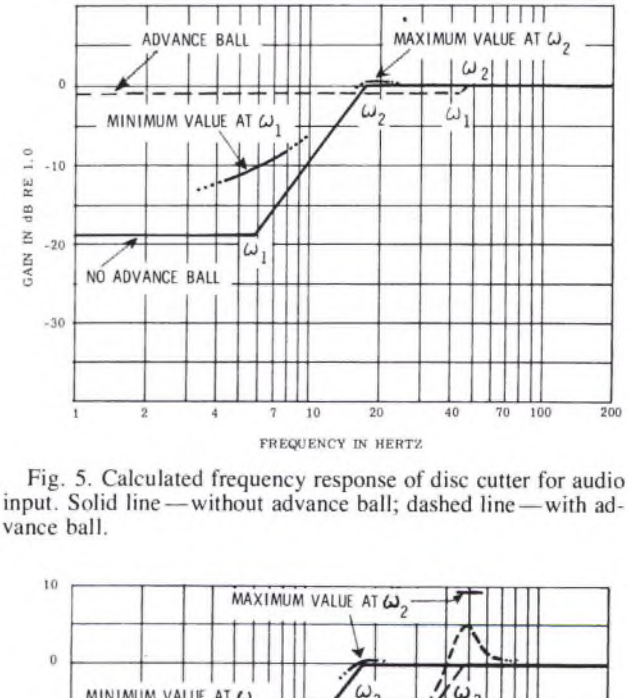

Fig. 5. Calculated frequency response of disc cutter for audio input. Solid line—without advance ball; dashed line—with advance ball.

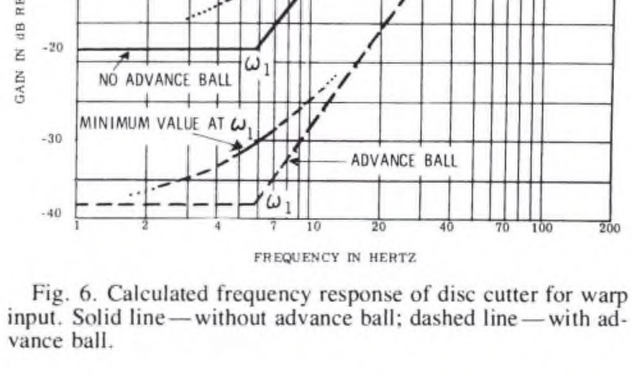

Fig. 6. Calculated frequency response of disc cutter for warp input. Solid line—without advance ball; dashed line—with advance ball.

inputs? What is the mechanical impedance that the cutting stylus works into? In addition, the model can be used to study how well a cutting system is isolated from vibrations of the cutting-room floor. To do this, however, requires that the model be expanded to include the lathe assembly and floor dynamics.

APPENDIX

This appendix describes four experiments we used to determine values of mass, stiffness, and damping for our model. The experiments were performed on a Westrex 3DII disc cutter head mounted on a Scully recording lathe [4].

First, the values of K_m and M were found by measuring the frequency of the lightly damped resonance of the freely suspended cutter head. (The dashpot was disconnected and the stylus and advance ball were both off the lacquer disc.) The stylus of a rigidly held phonograph cartridge was placed in contact with the cutter head frame. The cartridge output was viewed on a storage oscilloscope while the head was shock excited. ω was then calculated from the observed vibration period. Next, a known mass m (0.388 kg) was added to the cutter head frame, and a new resonance frequency ω_m was observed. K_m and M were then calculated by simultaneous solution of the following equations:

$$\omega^2 = \frac{K_m}{M} \quad \text{and} \quad \omega_m{}^2 \quad \frac{K_m}{M+m}$$

$$M = 1.31 \quad \text{kg}$$

$$K_m = 1.79 \times 10^3 \quad \text{N/m}.$$

Second, to estimate the numerical value of the dashpot damping B, we examined the vibration waveform of the suspended cutter head with the dashpot reconnected. (In this experiment the added mass m remained secured to the cutter head.) No overshoot in the transient response was observed. The system was therefore more than critically damped, giving us the inequality $\zeta > 1$, where $\zeta = \omega_m (M + m)/(2 B)$. It follows that $B > 2\omega_m (M + m)$. Substituting the previously measured values for ω_m and M then yields

$$B > 126 \quad \frac{\text{N·s}}{\text{m}}.$$

This inequality allows us to place bounds on the frequency-response deviations from the asymptotes at their corner frequencies.

Third, the stiffness of the advance ball resting on the lacquer was also calculated by measuring the natural vibration of the cutter head. The advance ball was properly placed on the lacquer, the system was shock excited, and the vibration frequency ω was measured. The governing equation, $\omega^2 = (K_m + K_b)/M$, was then solved for K_b:

$$K_b = 9.74 \times 10^4 \quad \text{N/m}.$$

And last, the groove stiffness presented to the cutting stylus, K_g, was deduced as follows. We measured the frequency response ($20 \log|B(j\omega)|$) of a lacquer disc that was recorded while using the advance ball. A 1-dB loss occurred abruptly for all low frequencies in the manner shown in Fig. 5. We chose the numerical value of K_g so that our model would correctly account for the magnitude of this loss. Therefore, we set

$$B\ (0) = \frac{K_m+K_b}{K_m+K_b+K_g} = 0.89.$$

Previously calculated values for K_m and K_b then yielded

$$K_g = 1.23 \times 10^4 \quad \text{N/m}.$$

The frequency at which this 1-dB loss occurs is predicted from our model to be 45 Hz, as shown in Fig. 5.

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Daniel W. Gravereaux, Manager, Recording Research, CBS Technology Center, joined CBS in 1964. He graduated from the University of Connecticut in 1963, earning a BSEE degree, and has pursued postgraduate studies at that institution.

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Dynamic Modeling and Analysis of a Phonograph Stylus*

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There are many possible choices as to the modeling technique used to represent a mechanical system. The traditional method within phonocartridge development has been the electrical/mechanical analog circuit. The approach taken in this study was to mathematically describe the stylus system in terms of a series of continuous, flexible beams undergoing transverse vibrations. It is assumed that the structure undergoes sinusoidal motion and thus the wave equation for the beam has a known general solution. The boundary conditions are progressively improved to provide a realistic representation of the system. New approaches to stylus shank design are presented.

INTRODUCTION: There are many possible choices for the modeling technique used to represent a mechanical system. One method often used in the development of phono cartridges has been the electrical-circuit mechanical analog. Here the stylus shank is simulated by dividing it into a number of reasonable mass divisions separated by compliance terms representing the stiffnesses of the corresponding elements. An electrical circuit can then be constructed whose performance is analogous to the lumped mechanical system. The primary advantage of this method is that with relatively few elements and low complexity, the general system characteristics can be observed. Answers to basic questions such as the effect of increasing stylus mass or bearing stiffness can be evaluated on such a model. However, detailed analysis using this model is limited since it is difficult to determine the exact values for the stiffnesses and masses and their interrelationships in the analog.

The approach taken in this study was to treat the stylus shank as a uniform continuous beam or series of such beams undergoing transverse vibrations. Euler's equation describing the vibration of continuous beams is used in the derivations which follow. It is assumed that the structure undergoes time harmonic motion and, therefore, the governing equation for the uniform beam has a known general solution. To determine the specific solution the supports of the beam, referred to as the boundary conditions, must be defined and incorporated into the analysis.

The boundary conditions for this analysis were progressively improved to obtain a realistic generalized model. Initially the tip end of the shank was represented by a mass for the diamond stylus and a spring used to represent the interaction with the record material. A transducer element with elastic bearing represented the other end. The present model includes damping in conjunction with each boundary condition and allows for changes in compliance and damping at each frequency. The model can also evaluate shanks of nonuniform cross section.

The advantages of this type of analysis are that specific dimensional variations can be simulated. Thus a wide range of structures can be accurately simulated and evaluated with respect to one another. This direct method is well suited to answer questions like: What is the effect on response and trackability of increasing the wall thickness from 0.001 in (0.025 mm) to 0.0015 in (0.038 mm)

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over a portion of the shank? On the other hand, it is often necessary to make many simulations to study the influence of a single parameter. The direct approach may require more time and effort than simply "turning a dial" on an equivalent analog model. Although the continuous-system approach provides for more detail than the conventional analog method, it should be used only when appropriate.

Keeping these advantages and limitations in mind, the development of the mathematical model will be presented in two stages: a derivation for the uniform beam with mass and stiffness end conditions and the development of the improved model. The results will then be discussed and some of the future improvements which are being developed will be described.

1. DERIVATION OF UNIFORM BEAM

Fig. 1 shows the system as it was first formulated. It consists of three parts: the uniform beam representing the phono stylus shank, the transducer element and its spring supports, and the diamond tip mass and attached spring element. The parameters necessary to describe the beam are its mass per unit length ρ, Young's modulus E, cross-sectional moments of inertia I, and its overall length a. The rotational and translational spring attached to the tip mass represents the record vinyl compliance. All stiffness values have been assumed to be linear with respect to amplitude and initially constant with respect to frequency. Although further study is needed in this area, preliminary evidence through response and trackability measurements indicates that the linearity assumption is valid for the relatively small amplitudes present in record grooves.

The model describes only the translational motion of each segment of mass within the system. Torsional or compressional modes of the shank are not considered in this analysis. The displacement of each of the three elements (shank, transducer element, and diamond tip) are denoted by $w(x,t)$, $y(t)$, and $z(t)$, respectively. Each is measured from a common line called the reference line. The function $w(x,t)$ defines the displacement of the stylus shank as a function of the position x along its length for any time t. Dimension y refers to the position of the axis along the bearing centerline. The z displacement function represents the position of the diamond tip from the reference line.

The input to the system is shown as u, and it is assumed to be of the form[1]

$$u = U e^{i\omega t}. \tag{1}$$

Since steady-state sinusoidal inputs have been assumed, it follows that the other displacement functions can also be expressed as

$$w(x,t) = W(x) e^{i\omega t} \tag{2}$$

$$y(t) = Y e^{i\omega t} \tag{3}$$

$$z(t) = Z e^{i\omega t}. \tag{4}$$

The mechanical system shown in Fig. 1 can be analyzed by determining the general form for the beam deflection function $W(x)$ and then applying the boundary constraints due to transducer element and tip dynamics to arrive at a specific solution for $W(x)$. Once $W(x)$ has been determined, other output relations can be readily derived. For example, the Y and Z functions are obtained after evaluation of $W(x)$ at the respective endpoints $x = 0$ or $x = a$.

The Euler beam equation is used to obtain the general form for the instantaneous deflection characteristics of the beam. For a uniform beam, the equation can be written as

$$EI \frac{d^2 w}{dx^4} = -\rho \frac{d^4 w}{dt^2} \tag{5}$$

where ρ represents the mass per unit length and the product EI is the bending stiffness.

The equation can be examined by noting that the left-hand term represents the loading per unit length in a static beam and the right-hand term represents the translational inertia of the beam per unit length. Thus dynamically the equation states that the load on the beam at any point is equal to the inertia force of the beam at that point.

The Euler equation assumes that the deflections of the beam and its diameter are small compared to its overall length. Thus the effects of rotational inertia of the beam cross section are negligible compared to translation and are not included in the formula. In the application of a phono stylus these assumptions are valid since the beam length is typically ten times greater than the diameter and approximately a hundred times greater than the input amplitudes. Also, implicit within the following derivations is the assumption that internal losses within the shank are negligible, that is, no damping exists within the shank material, which is an accurate description for most metals and some plastics.

When the beam undergoes periodic motion,

$$w(x,t) = W(x) e^{i\omega t}.$$

Substituting $w(x,t)$ into Eq. (5) yields the standard ordi-

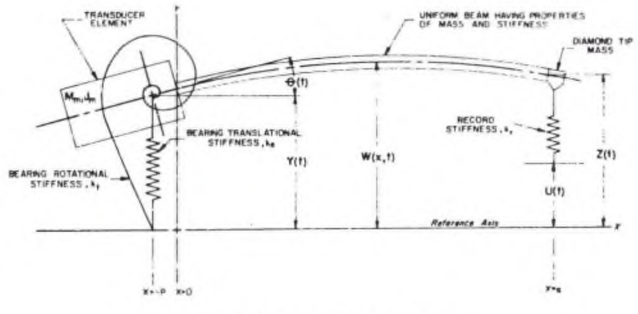

Fig. 1. Mechanical system.

[1] Throughout this paper the convention is adopted that lowercase displacement symbols represent the complete functions of time while uppercase symbols define the magnitudes of the variables independent of time. In the analysis the uppercase letters $W(x)$, Y, and Z represent the displacement magnitudes independent of time, although each may contain real and imaginary terms, that is, that may have a different phase relationship with respect to the input.

nary differential equation for the uniform beam:

$$\frac{d^4W}{dx^4} - \lambda^4 W = 0, \qquad \text{where } \lambda^4 = \frac{\rho\omega^2}{EI}\cdot \tag{6}$$

In this case the general solution for W, the deflection mode of the beam, is of the form

$$W(x) = A \sinh \lambda x + B \cosh \lambda x + C \sin \lambda x + D \cos \lambda x. \tag{7}$$

This equation has four unknowns (A, B, C, and D) for any given frequency coefficient λ. If these constants can be determined, the motion of the stylus assembly can be completely described for any given frequency and amplitude input.

To solve for these four unknowns it is necessary to derive four boundary constraint equations. At the ends of the beam, $W(x)$ and each of the derivatives of $W(x)$ can be derived and evaluated in terms of the unknown constants A, B, C, and D. For example, at $x = 0$,

$$W = B + D \tag{8}$$

$$\frac{dW}{dx} = (A + C)\lambda \tag{9}$$

$$\frac{d^2W}{dx^2} = (B - D)\lambda^2 \tag{10}$$

$$\frac{d^3W}{dx^3} = (A - C)\lambda^3. \tag{11}$$

These equations plus the four boundary conditions to be derived will be used to obtain a specific solution for the deflection shape $W(x)$.

1.1 Boundary Conditions for Reactions at the Transducer End of the Shank

The diagram in Fig. 2 shows the forces and torques assumed to be acting on the transducer with v_1 and m_1 equal to the shear and moment reactions at the shank–transducer interface, and $k_e y$ and $k_t\theta$ representing the instantaneous bearing constraints. Summing these forces and moments and equating with the inertial forces, the equations of motion can be derived. Assuming θ to be small, that is, $\cos\theta \simeq 1$, the equations can be written as

$$-k_e y - v_1 = M_m \ddot{y} \tag{12}$$

$$m_1 - v_1 p - k_t\theta = J_m \ddot{\theta}. \tag{13}$$

Solving for v_1 and m_1 yields

$$v_1 = -M_m \ddot{y} - k_e y \tag{14}$$

$$m_1 = v_1 p + k_t \theta + J_m \ddot{\theta}. \tag{15}$$

These equations represent the time-dependent equations of motion of the transducer-bearing system. Again employing the assumptions of steady-state sinusoidal motion it follows that Eqs. (14) and (15) can be rewritten in terms of the magnitudes of the force variables as follows:

$$V_1 = (M_m\omega^2 - k_e)Y \tag{16}$$

$$M_1 = V_1 p + k_t\theta - J_m\omega^2\theta = (M_m\omega^2 - k_e)Yp + (k_t - J_m\omega^2)\theta. \tag{17}$$

At the boundary defined by $x = 0$, the following is true:

$$Y = W - p \sin\theta \tag{18}$$

$$\frac{dW}{dx} = \tan\theta. \tag{19}$$

Using the simplifications $\sin\theta \simeq \theta$ and $\tan\theta \simeq \theta$, substitution into Eqs. (16) and (17) yields

$$V_1 = (M_m\omega^2 - k_e)(W - p\frac{dW}{dx}) \tag{20}$$

$$M_1 = (M_m\omega^2 - k_e)p(W - p\frac{dW}{dx}) + (k_t - J_m\omega^2)\frac{dW}{dx}\cdot \tag{21}$$

Also, it is known that for a continuous beam subject to a load, the shear and moment reactions can be described in terms of the beam displacement as

$$V_1 = EI\,\frac{d^3W}{dx^3} \tag{22}$$

$$M_1 = EI\,\frac{d^2W}{dx^2}\cdot \tag{23}$$

Substituting gives the two equations of motion for the magnet end of the beam at $x = 0$:

$$EI\,\frac{d^3W}{dx^3} = (M_m\omega^2 - k_e)(W - p\frac{dW}{dx}) \tag{24}$$

$$EI\,\frac{d^2W}{dx^2} = (M_m\omega^2 - k_e)p(W - p\frac{dW}{dx}) + (k_t - J_m\omega^2)\,\frac{dW}{dx}\cdot \tag{25}$$

1.2 Boundary Conditions for Reactions at the Diamond Tip End of the Stylus Shank

Fig. 3 shows the reaction forces acting on the diamond tip due to the stylus shank and record input. The force of the tip f_r is due to the deflection of the record compliance and can be written as

$$f_r = k_r(Z - U). \tag{26}$$

As in the previous case, balancing the instantaneous vertical forces and moments yields the equations describing the diamond tip motion. Solving these for y_2 and m_2 yields

$$v_2 = M_t\ddot{z} + k_r(z - u) \tag{27}$$

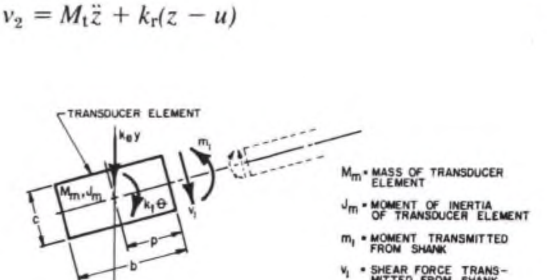

Fig. 2. Left end beam boundary conditions.