$$m_2 = -J_t\ddot{\phi}. \tag{28}$$

Again, in response to steady-state sinusoidal inputs, these equations become

$$V_2 = k_1 (Z - U) - M_t\omega^2 Z \tag{29}$$

$$M_2 = J_t\omega^2\Phi. \tag{30}$$

At the end of the shank (position $x = a$) the following is true:

$$Z = W \tag{31}$$

$$\Phi \simeq \frac{\mathrm{d}W}{\mathrm{d}x} \tag{32}$$

$$V_2 = k_r (W - U) - M_t\omega^2 W \tag{33}$$

$$M_2 = J_t\omega^2 \frac{\mathrm{d}W}{\mathrm{d}x}. \tag{34}$$

Substituting the shear and moment relations in terms of W gives the final two boundary conditions for the problem at $x = a$:

$$EI \frac{\mathrm{d}^3 W}{\mathrm{d}x^3} = k_r (W - U) - M_t\omega^2 W \tag{35}$$

$$EI \frac{\mathrm{d}^2 W}{\mathrm{d}x^2} = J_t\omega^2 \frac{\mathrm{d}W}{\mathrm{d}x}. \tag{36}$$

2. MATRIX REPRESENTATION OF SYSTEM

Eqs. (24), (25), (35), and (36) are the four boundary conditions expressed in terms of W and its derivatives. These can be rewritten in terms of the four unknowns A, B, C, and D using the relations determined for $W(x)$ and its derivatives at each endpoint $x = 0$ and $x = a$ [Eqs. (8)–(11), etc.]. This system of four equations is best arranged in matrix form $[A] \cdot [X] = [C]$, where $[A]$ represents the coefficient matrix which describes the constraints on the vibrating beam, $[X]$ represents the unknown vector to be solved containing variables A, B, C, and D, and $[C]$ is a column vector representing the inputs to the mechanical system. The matrix representation of the system derived from Fig. 1 is shown in Fig. 4.

A computer program was written which inputs all the necessary data describing the system and arranges it into the matrix form. The matrix is solved by using a Gaussian elimination method and uses the results to calculate the beam deflection function $w(x,t)$ and the desired outputs. Initially, the program was written to provide the frequency response and impedance characteristics over the frequency range specified.

3. IMPROVEMENTS TO BASIC SYSTEM DERIVATION

The results of the first system modeled were encouraging but revealed the weakness of not including damping in the model. The frequency response showed several resonances in frequency ranges which seemed reasonable; however, the shape of the curve was unrealistic due to the infinite Q of each resonance.

The use of complex notation $(a + \mathrm{i}\, b)$ permits the original system to be readily improved by the addition of system damping. It is assumed that internal damping in the stylus shank is negligible and that the entire damping which occurs is due to the elastic bearing and record material deformations. It was found that both rotational and translational damping was required to realistically represent the stylus mechanical system. Thus dampers were paralleled with each spring element in the system. This was computationally accomplished by including imaginary terms for each of the spring elements. For example, k_t, shown in Fig. 1, was replaced by $(k_t + \mathrm{i}\omega c_t)$, and k_e became $(k_e + i\ \omega c_e)$, where c_t and c_e are the usual coefficients of viscous damping.

Initially it was assumed that the compliance and damping coefficients remained constant with changes in frequency, and the values were determined empirically to give realistic high-frequency response characteristics. However, fixed values for the bearing stiffness or the damping coefficient cannot be representative in both the high- and low-frequency regions since typical bearing elastomers have material properties that change with frequency. To determine how these parameters might change with frequency, direct-impedance measurements were made on several potential bearing materials.

Fig. 5 represents the measurement technique used in the study. A small cube of each material was loaded into the transducer and measured under approximately the same loading as might be expected in a phono cartridge application. By measuring the material directly over the frequency range of interest, the overall impedance characteristics can be determined. Also, using the measured

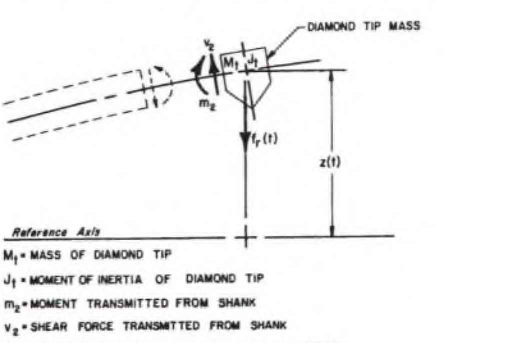


Fig. 3. Right end beam boundary conditions.

$$\begin{bmatrix} EI\lambda^3 + (M_m\omega^2 - k_r)\rho\lambda & (k_r - M_m\omega^2) & (M_m\omega^2 - k_r)\rho\lambda - EI\lambda^3 & (k_r - M_m\omega^2) \\ (M_m\omega^2 - k_r)\rho^2\lambda - (k_t - J_m\omega^2)\lambda & EI\lambda^2 - (M_m\omega^2 - k_r)\rho & (M_m\omega^2 - k_r)\rho^2\lambda - (k_t - J_m\omega^2)\lambda & -EI\lambda^2 - (M_m\omega^2 - k_r)\rho \\ EI\lambda^3 \cosh \lambda a + (M_t\omega^2 - k_r) \sinh \lambda a & EI\lambda^3 \sinh \lambda a + (M_t\omega^2 - k_r) \cosh \lambda a & -EI\lambda^3 \cos \lambda a + (M_t\omega^2 - k_r) \sin \lambda a & EI\lambda^3 \sin \lambda a + (M_t\omega^2 - k_r) \cos \lambda a \\ EI\lambda^2 \sinh \lambda a - J_t\omega^2\lambda \cosh \lambda a & EI\lambda^2 \cosh \lambda a - J_t\omega^2\lambda \sinh \lambda a & -EI\lambda^2 \sin \lambda a - J_t\omega^2\lambda \cos \lambda a & -EI\lambda^2 \cos \lambda a + J_t\omega^2\lambda \sin \lambda a \end{bmatrix} \bullet \begin{bmatrix} A \\ B \\ C \\ D \end{bmatrix} = \begin{bmatrix} 0 \\ 0 \\ -k_r U \\ 0 \end{bmatrix}$$

Fig. 4. Matrix representation of mechanical system of Fig. 1.

phase relationship between the force and velocity signals it is possible to separate forces which are primarily "springlike" from those that are primarily "dissipative" in nature. These components representing the elastic and loss material modulus are plotted against frequency on log–log paper. Then, fitting a straight line through each set of data, equations can be formulated of the form

$$E(\text{elastic modulus}) = a f^{+\alpha} \tag{37}$$

$$E'(\text{loss modulus}) = b f^{-\beta} \tag{38}$$

where a, b, α, and β are constants to be determined.

These functions were determined for the material characteristics of the elastic bearing supporting the transducer. Using the dimensions of the bearing, a corresponding function was derived for the stiffness and dissipative model elements k_t, k_e, c_t, and c_e. This procedure resulted in fairly good agreement with measured data; however, it was found that slight adjustments were necessary to more closely simulate the bearing reactions.

The final improvement desired was the ability to simulate the composite stylus beam, that is, styli which can be thought of as composed of two or more continuous beam sections (Fig. 6). The analysis for each section is done independently. Each section may have a different cross-sectional mass ρ, Young's modulus E, or cross-sectional moment of inertia I. However, it is assumed that both shank sections undergo small deflections compared to the overall length and each has no internal damping. Then each section can be described by a separate beam equation with boundary conditions determined by the neighboring sections.

If $w(x,t)$ represents the motion of section 1 as already discussed and $q(x,t) = Q(x)e^{i\omega t}$ represents the motion of section 2, then $Q(x)$ is of the form

$$Q(x) = E \sinh \lambda_2 x + F \cosh \lambda_2 x + G \sin \lambda_2 x + H \cos\lambda_2 x \tag{39}$$

where

$$\lambda_2^{\,4} = \frac{\rho_2 \omega^2}{E_2 I_2}\,. \tag{40}$$

This introduces four new unknowns E, F, G, and H. Since it is assumed that the ends of each section are rigidly joined, the displacements, slopes, shear, and moment reactions must be similar for both sections at the junction. If the intersection of the tubes is at the point $x = a\alpha$, where α is a percent of the total length, then the four boundary conditions which describe these relationships at the intersection $x = \alpha a$ are

equal displacements $$W = Q \tag{41}$$

equal slopes $$\frac{dW}{dx} = \frac{dQ}{dx} \tag{42}$$

equal bending moments $$E_1 I_1 \frac{d^2W}{dx^2} = E_2 I_2 \frac{d^2Q}{dx^2} \tag{43}$$

equal shear forces $$E_1 I_1 \frac{d^3W}{dx^3} = E_2 I_2 \frac{d^3Q}{dx^3}\,. \tag{44}$$

With these four additional conditions the problem is again solvable. The matrix of simultaneous equations can be formulated, programmed on the computer, and the coefficients describing the motion of the shank sections 1 and 2 determined. This procedure must be repeated for each additional section included in the system model.

The present model has been derived using four separate shank sections. This allows a wide variety of compound shank geometries to be exactly simulated and provides for a far greater number of geometries to be approximated in form. Examples of stylus shanks that can be modeled using the present system of equations are shown in Fig. 7.

The stylus shank system as it is presently modeled is shown in Fig. 8. This represents the system derived in this report plus the improvements mentioned above. Fig. 8 lists all the variables that can be used as inputs to describe the system. The matrix representation of Fig. 8 is a 16-equation system and is not shown in this paper.

4. RESULTS AND APPLICATIONS

The basic solution that is obtained by the computer program is in terms of $W(x)$, $Q(x)$, etc., which define the geometrical description of the composite stylus shank centerline for a particular shank geometry, boundary conditions, input amplitude, and frequency. From this output all the other results can be derived. To date the computer program has been used to supply information on 1) modal shape of the stylus shank, 2) stylus frequency response curve, 3) stylus impedance or trackability curve, and 4) stylus phase response curve. Examples of the various output formats follow.

4.1 Modal Analysis

Using the equations determined for $W(x,t)$, $Q(x,t)$, etc., the position of the shank centerline at a particular frequency, amplitude, and phase of input is drawn. Fig. 9 is a model of a uniform 0.020 in (0.5-mm) diameter, 1-mil-

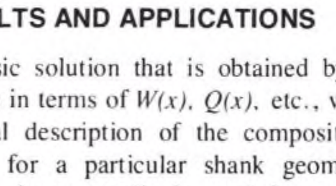


Fig. 5. Impedance measurement.

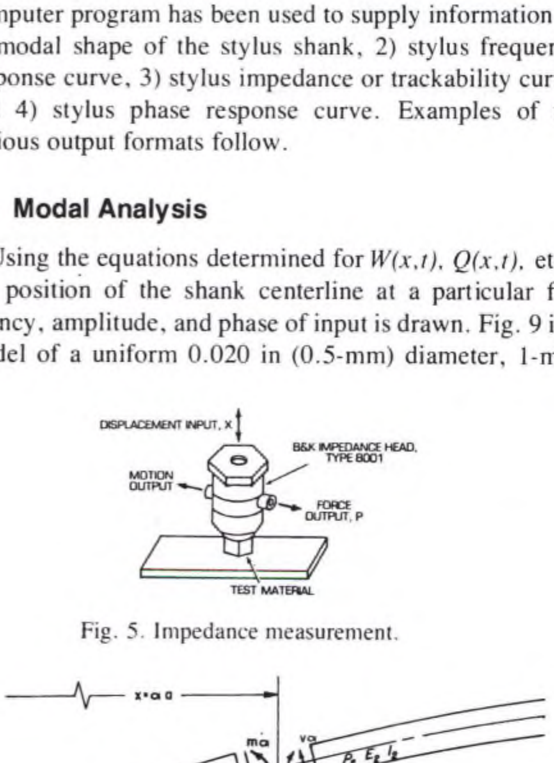


Fig. 6. Diagram of beam junction.

wall aluminum tube having relatively high inertia boundary constraints and undamped elastic supports. It is shown vibrating at its resonance frequency of approximately 19.5 kHz. All displacement dimensions (input amplitudes and shank deflections) have been scaled by more than the horizontal dimensions in order to better observe the modal shape. It is interesting to note the position of the diamond tip at this instant. The diagram shows a significant amount of wear at this frequency and also difficulty in maintaining record contact (poor trackability). Also, note that the center of the transducer element has been displaced from its intended pivot position in the system. Fig. 10 illustrates positions of the same stylus system at sequential instants of time. The series represents one quarter of a full cycle of the resonance frequency.

Fig. 11 shows the same system at a frequency of approximately 35 kHz. At this frequency the simulated response curve showed a null and the diagram confirms the relatively small amount of transducer rotation compared to Fig. 10 throughout the cycle. Note also that at this frequency the transducer rotation is approximately 180° out of phase with the motion of the record and tip.

Fig. 12 illustrates the dynamic interactions between the

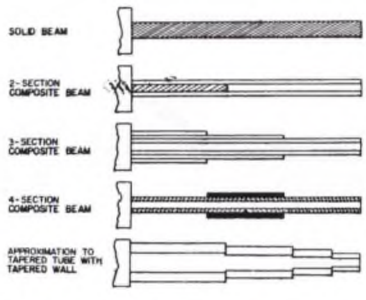


Fig. 7. Possible stylus shank geometries that may be simulated using present model.

shank and the record where a phono stylus is vibrated at the frequency of resonance between shank mass and record compliance. The system is a 2-mil-wall aluminum tube with additional tubular aluminum reinforcement, a transducer element, and realistic damping values for bearing supports. The drawings illustrate that only the front portion of the shank and diamond tip mass resonate with the record compliance since the transducer element mass is partially decoupled from the system.

Fig. 13 illustrates the motion of the same system at 90 kHz. While this frequency is not of practical interest, it does indicate the capabilities of the model. Here two nodes are present at a higher order resonance of the system.

4.2 Frequency Response Curve

Since many magnetic phono cartridges obtain an output based on the rotation of the transducer, it is necessary to calculate the transducer element rotation angle for each frequency. Because the model assumes the transducer element to be rigidly attached to the end of the shank, the slope of the shank at the transducer interface is used to determine the rotation angle. Thus the program calculates the absolute value of dW/dx evaluated at $x = 0$, and scales the results for a velocity-sensitive device to obtain the response data. After all the frequencies have been calculated, the entire graph is outputted on a log–log grid system which is scaled to be easily traced on standard response graph paper. Since the graphical display provides only limited resolution on the expanded log scale, the numerical response values are also listed on a separate page.

Fig. 14(a) shows an actual response curve generated by the program. In Fig. 14(b) a line has been traced as a smooth curve through the data points on the grid system of Fig. 14(a). Fig. 14(b) represents the predicted response of a

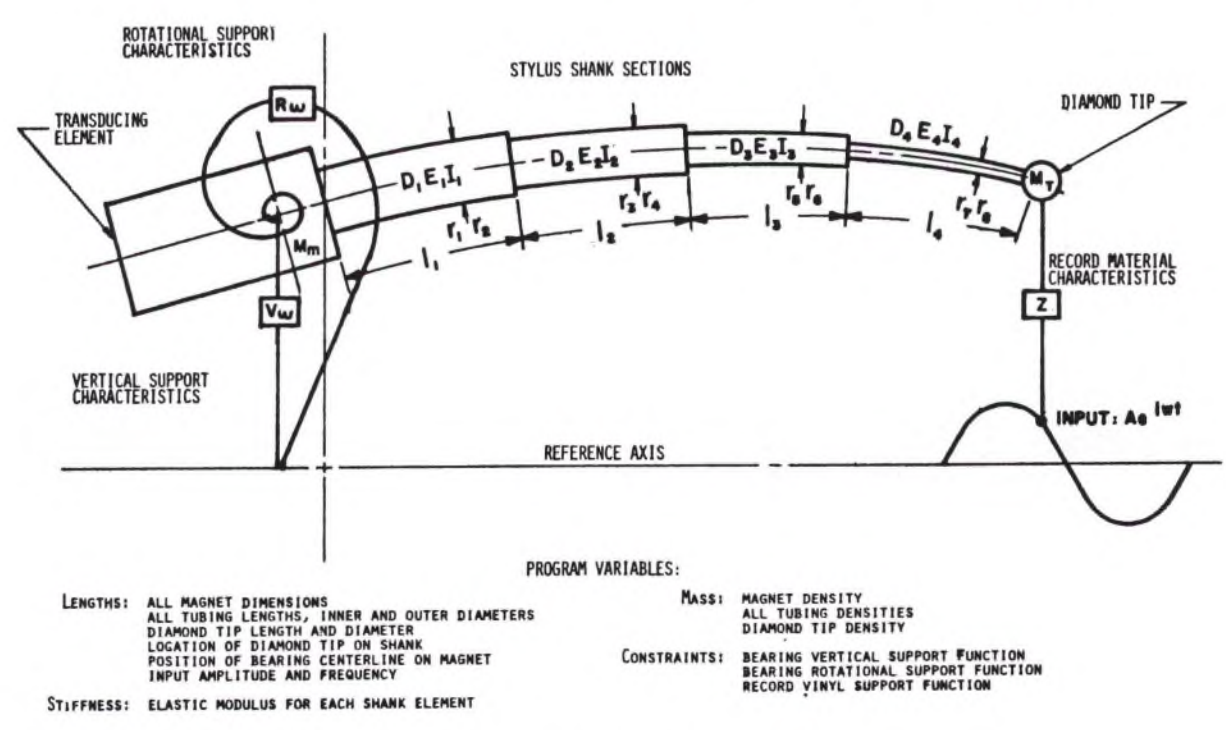


Fig. 8. Phono stylus model used in computer simulation.

stylus shank composed of an 0.018-in (0.46-mm) diameter, 1-mil-wall aluminum tube having an inner reinforcement rod of beryllium, approximately 40% of its length. The transducer element, bearing system, and diamond tip parameters represent typical values found in present high-quality phonograph stylus systems. Fig. 14(b) compares the actual measured mechanical response curve up to 50 kHz to the predicted curve from Fig. 14(a). Except for small variations in the height of the resonance peak, the predicted curve closely resembles the measured values.

As well as indicating the positions and heights of various resonances the response data can be used to determine if significant bending occurs in the mid-frequency region. If the shank deforms due to the impedance of the bearing, a mid-frequency sag in the response curve will result. Using the same bearing, transducer system, and diamond tip parameters as above, Fig. 15 shows two response curves of a similar composite shank

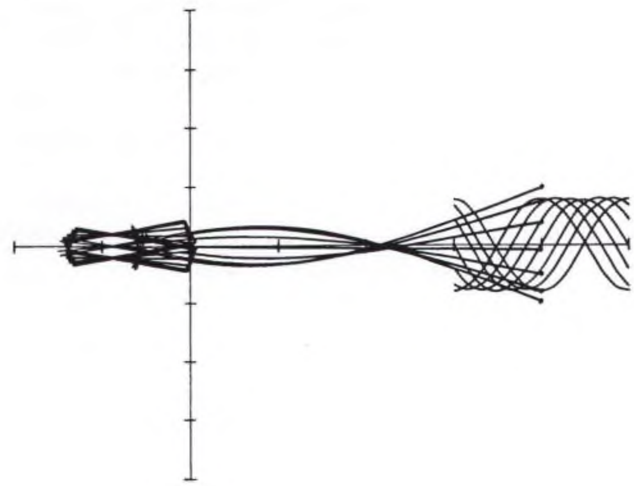


Fig. 11. Modal analysis, example 3. Frequency 35 kHz.

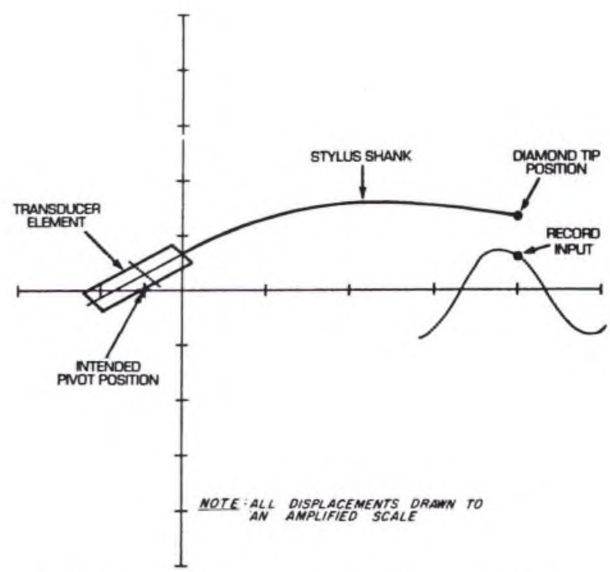


Fig. 9. Modal analysis, example 1. Frequency 19.5 kHz.

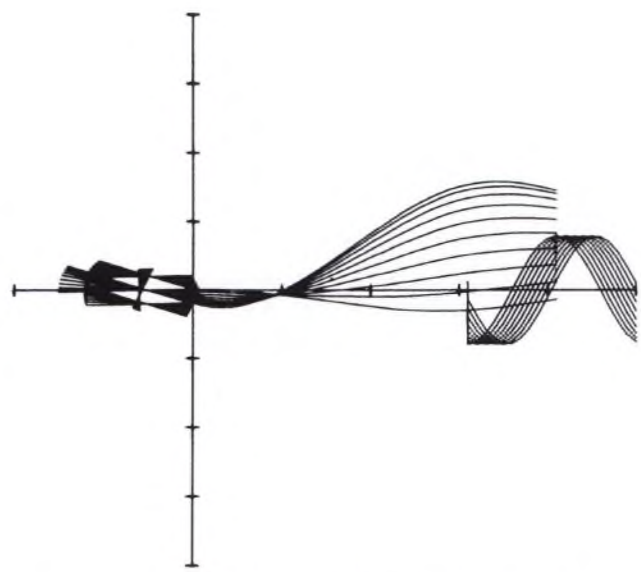


Fig. 12. Modal analysis, example 4. Frequency 59 kHz.

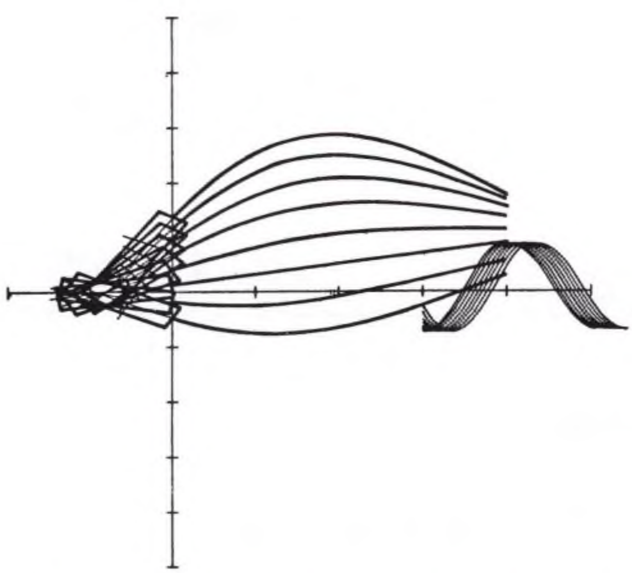


Fig. 10. Modal analysis, example 2. Frequency 19.5 kHz.

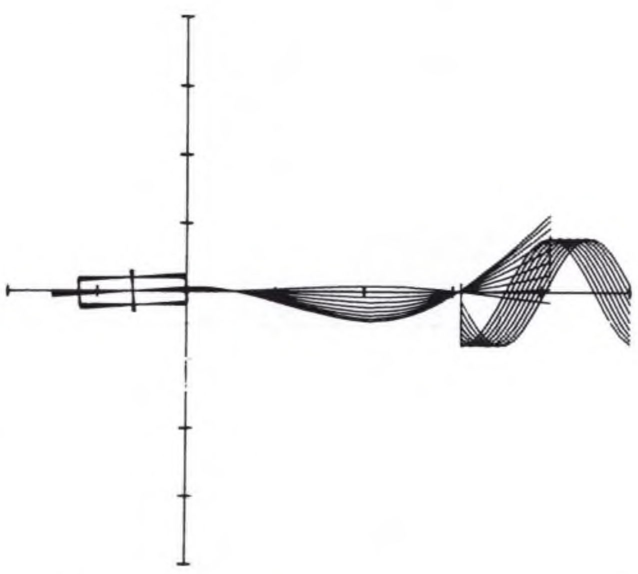
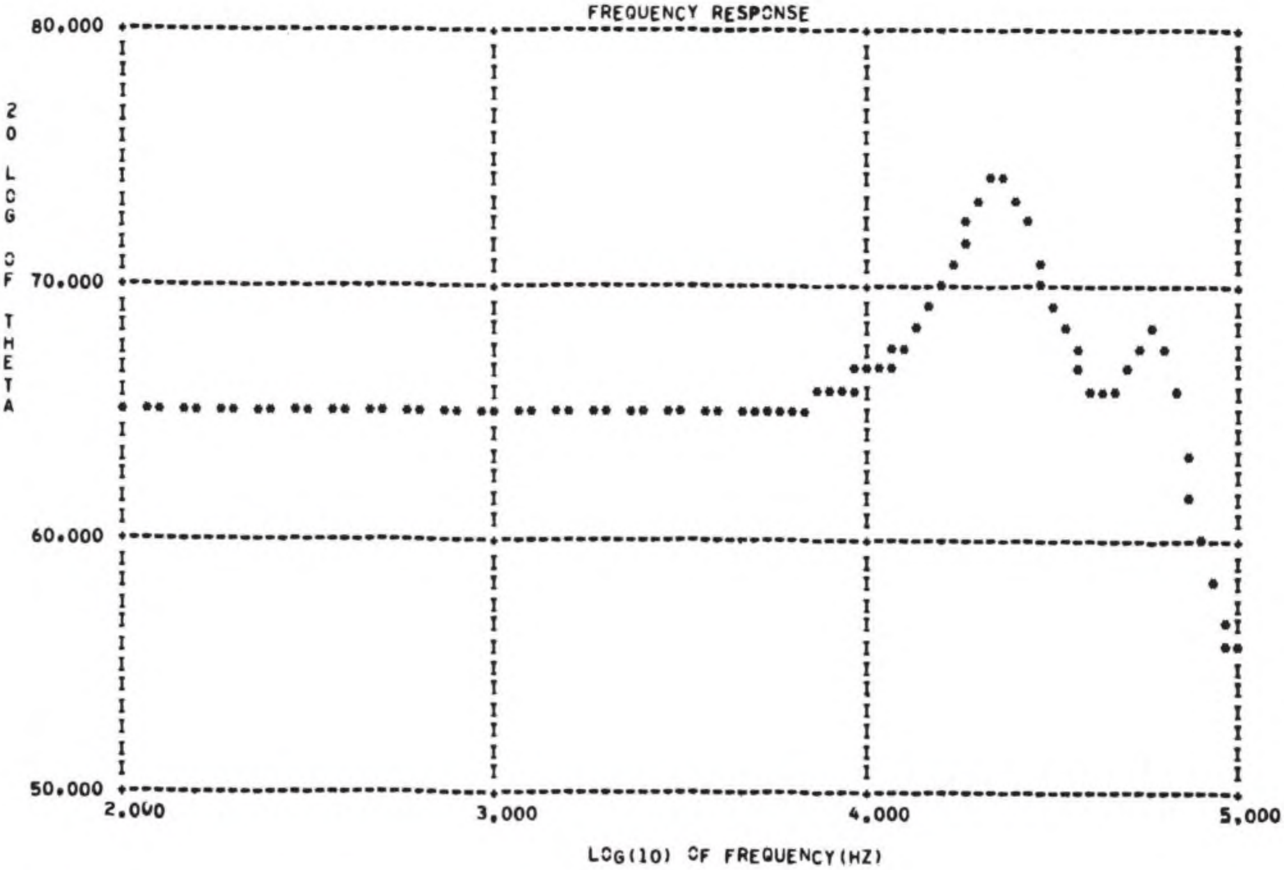
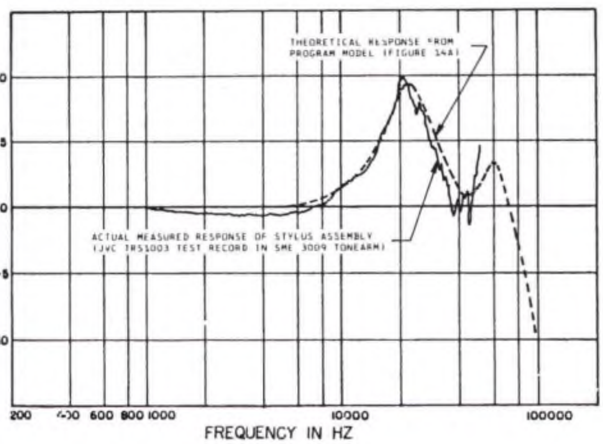


Fig. 13. Modal analysis, example 5. Frequency 90 kHz.



14(a)



14(b)

Fig. 14. Actual and predicted frequency response curves.

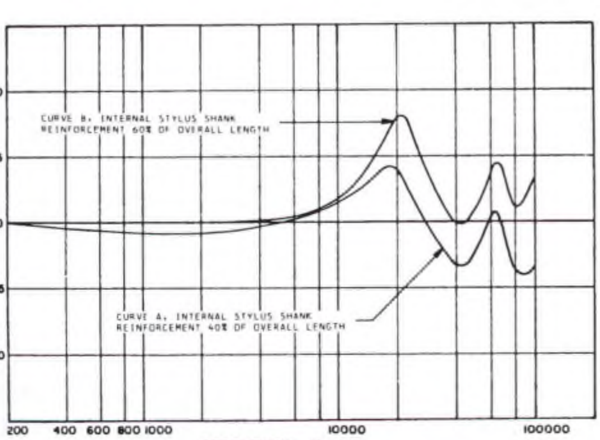


Fig. 15. Response curves for different reinforcing methods.

structure, except that the tubing diameter and wall thickness have been reduced. In curve A the beryllium reinforcing rod measures 40% of the total shank length, and in curve B the reinforcing rod is 60% of the total length. Thus for this geometry the 60% reinforcement must be used to minimize the mid-frequency response losses due to bending.

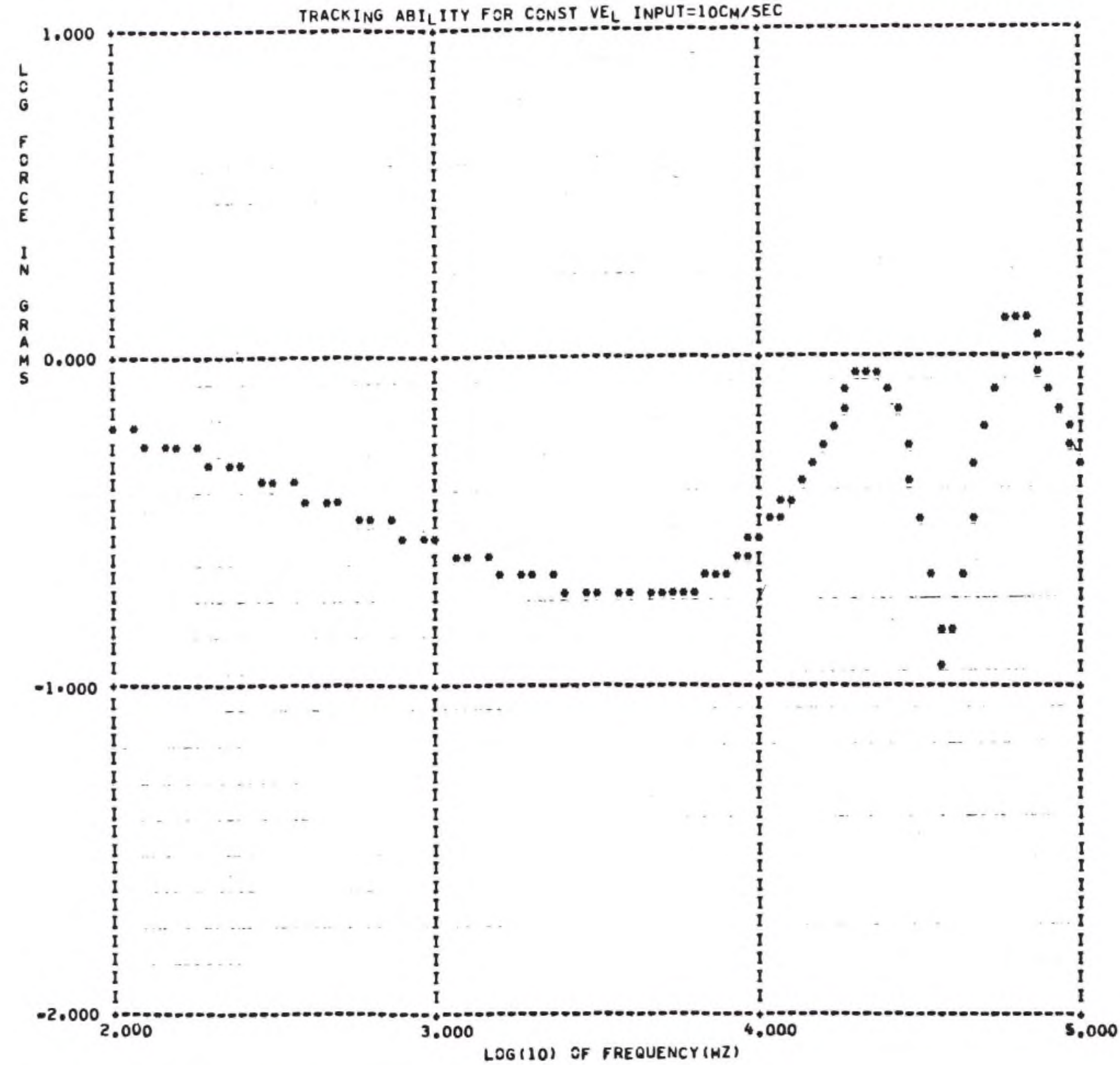
4.3 Impedance (Trackability) Curve

The impedance values are determined within the program by calculating the magnitude of the instantaneous force f_r on the diamond tip from the record input [Eq. (26)]. This value is obtained by multiplying the displacement of the tip from the record input (a measurement of the record vinyl deformation) times the record vinyl compliance term. When damping is included at the tip, this must also be included in the impedance measurement. The impedance plot generated by the program is the inverse of the normal trackability graph in that it represents the force required to track a constant-velocity input (the standard adopted was 10-cm/s peak velocity). As with the frequency response curve, the impedance plot generated by the program is scaled to be easily traced on standard trackability graph paper, and the values are listed on a separate page for reference. Mistracking in the actual sense of allowing the diamond tip to lose contact with the input signal is not possible in this model. However, a stylus with poor trackability will be indicated clearly as one which results in high forces in response to the input signal.

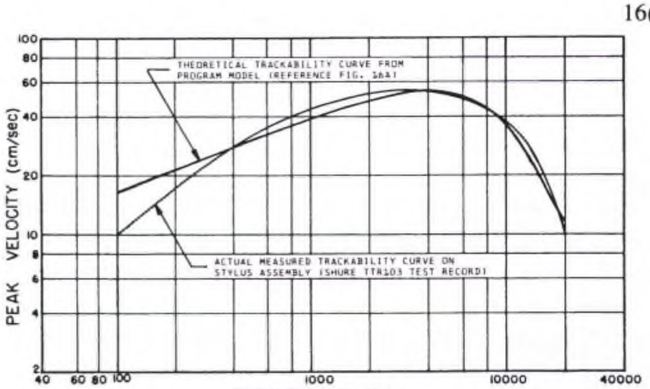
Fig. 16(a) shows the actual program impedance plot for the stylus system shown in Fig. 14. A smooth curve is drawn through the data points and presented in the standard trackability format in Fig. 16(b). The actual measured trackability plot is shown in the same graph and indicates a satisfactory agreement between the predicted and measured values.

5. SUMMARY AND CONCLUSIONS

The initial work during the study provided a basic mathematical model which describes the dynamics of the vibrating system. A stylus shank model was formulated using the equation for a continuous beam, and boundary conditions were developed to model the effects of transducer element, bearing, diamond stylus, and record materials. After the basic system of Fig. 1 was derived, programmed on the computer, and found to give good results, improvements were included which yielded a realistic generalized system representation. The results of the present formulation have been shown to closely match the measured performance on actual stylus assemblies. In addition to the usual frequency response and trackability-type evaluations, the program can provide insights into areas that were unavailable in past computer models. Information involving the deflection behavior of bearing, shank, or record can now be easily obtained. Also, detailed comparisons between styli having very similar



16(a)



16(b)

Fig. 16. Actual and predicted trackability curves.

mechanical structures can now be made by merely including the specific dimensions and material properties of the structures.

Although the present model is capable of simulating a wide variety of stylus assemblies, additional improvements are being considered. One such addition includes damping within the beam flexure derivation. This would greatly increase the range of materials which could be used in stylus shank simulation. Also, additional stiffness elements at the boundaries of the beam may be useful to examine the effects of nonrigid connections between the beam and its inertial end loading.

Although the mathematical manipulations are considered for any one derivation and solution, the simulation of additional geometries within the capabilities of the model can be easily accomplished. In fact, the complexity of the present model was due to the efforts to make the model generally applicable such that future theoretical work will not be necessary. Using the present model most practical stylus geometries can be simulated exactly or approximated in form.

In general, the theoretical results of both the basic system and the improved model have shown good agreement with past measured data. The program has proved useful in the development of the advanced stylus structure of the Shure V15 type IV referred to as the telescopic shank. In this study the requirements of low mechanical impedance and flat frequency response were carefully optimized. It seems likely that future studies using the program will confirm the practical and predictive value of the model.

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record pressing

Record Quality and Its Relation to Manufacturing*

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THE GOAL of record manufacturing is a record which reproduces in microscopic detail the surface geometry of the disk cut by the recording engineer. Anything which the manufacturing operations add or subtract can result only in deterioration of the information on the record, e.g., the addition of noise as ticks and "grit" or as distortion. We do not at this moment foresee the day when manufacturing will be asked to improve on the recording engineer's handiwork. We have our hands full achieving present goals.

The past decade has seen advances in the engineering aspects of recording and playback systems. These improvements have served not only to put more information on the record, in the form of wider frequency range, but to secure this wider frequency range on records having narrower and more closely spaced grooves. Improvements in manufacturing processes have been necessary in order to make this added information available on records delivered to the ultimate consumer. It is our purpose to review some of the changes in the manufacture of RCA Victor records. The changes reviewed will be those which have been major factors in narrowing the gap between the recording process and the final record.

Superficially, the manufacturing process has remained the same. The recorded disk is metalized to make the surface conductive, after which a negative master is made by electroforming. To obtain protection and a number of metal parts, a positive is made from this master by electroforming to obtain a mold or "mother." This mold serves to generate pressing masters or "stampers" which are used in plastic molding presses that were employed many years ago. Figure 1 illustrates the essential elements of the process. Consequently, it was necessary to investigate the matter in some detail to get at the improvements, just as we have to go beyond the four wheels of the automobile and water-cooled four-cycle internal-combustion engine to establish the differences between the latest and previous automobiles.

It should be apparent that the key to the problem is the stamper. If a stamper surface reproduces the surface of the recording in microscopic detail, a molded record surface reproducing this detail is possible. The first step, therefore, was a review of the matrix operations with the object of eliminating any operation which resulted in deterioration of the information on the recorded surface.

In the program which was initiated more than ten years ago it was recognized that mechanical work on the matrix surface, whether called polishing or abrasive cleaning, could result only in deterioration. Although tolerable on 78-rpm filled records, tested as they were on then current commercial equipment, it was recognized that such operations could not be tolerated on either fine-groove records or equipment of wide frequency range.

However, it was soon found that polishing could not be eliminated by fiat. The cure was worse than the disease. The first logical step was to substitute electrochemical and chemical methods of cleaning for the polishing and hand scrubbing which was then common. It was found that the resultant parts, molds or records, were so full of ticks that the sound was certainly not improved on. Many changes were still needed. Of these changes, three were chiefly responsible for making the elimination of polishing and the substitution of nonmechanical methods of cleaning practical.

These three major improvements were: the substitution of lacquer disks for wax, the control of stress in nickel solutions, and the control of impurities in the nickel solutions. The last two are not entirely independent. However, each contributed its major share to quality improvement, independent of the other.

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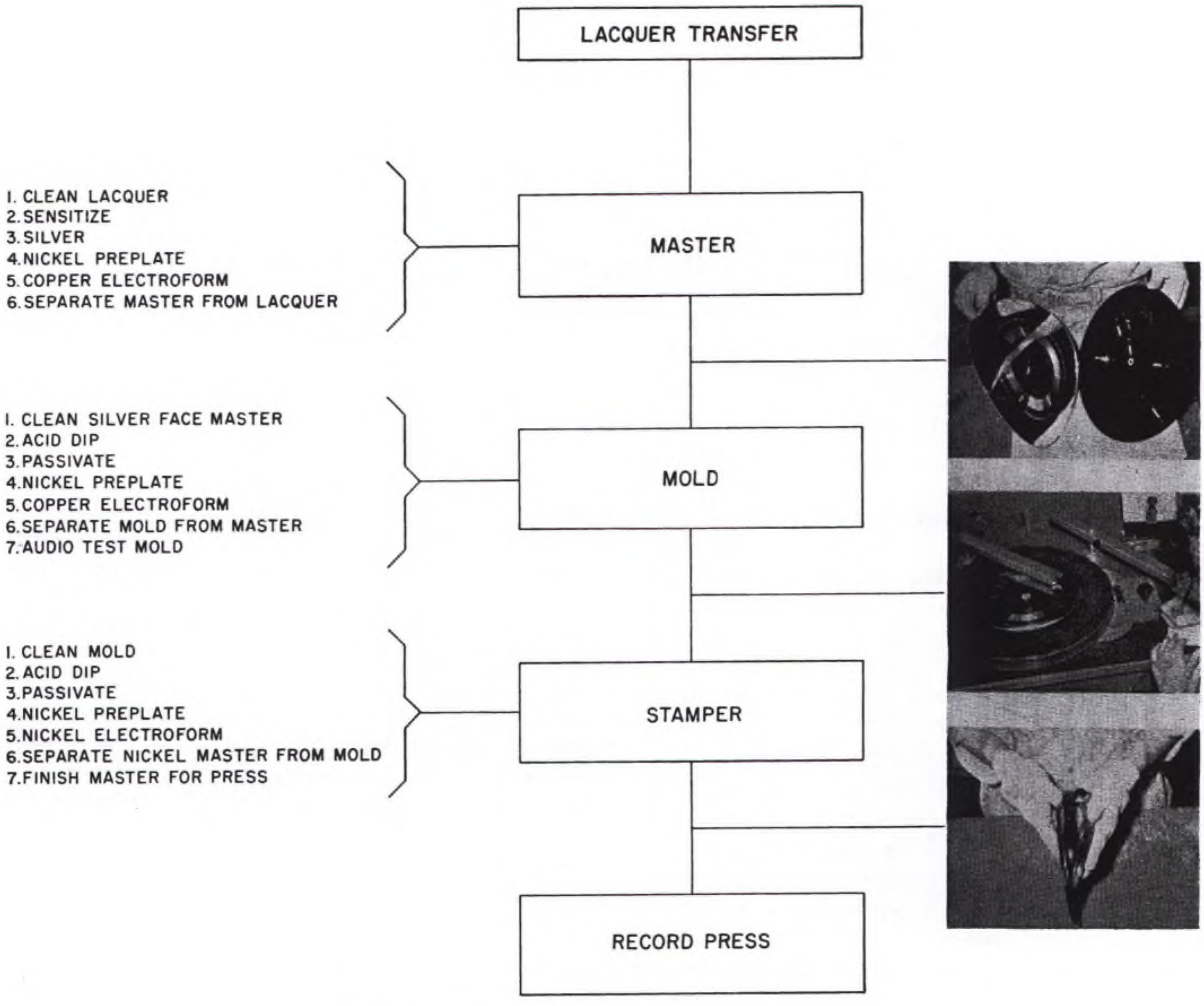

FIG. 1. Sequence of operations for disk record manufacturing.

For many years, the original recordings were made on a wax composition based on Montan wax. After the recording was made, the surface of the wax was made conductive by brushing graphite into the surface. The limitations of the process are apparent. First, the grain size of the graphite particles led to an inherent surface roughness and consequent surface noise. The danger of groove change by the brushing was always present. The results obtained depended on the skill of the operator.

In the late 1930's, Bell System engineers introduced gold sputtering. This step eliminated the crystallinity of the graphite particles and the dangers of brushing the grooves, but unfortunately other deficiencies appeared. These deficiencies were particularly noticeable with fine-groove work. They showed as small defects in the grooves which on playback manifested themselves as ear-splitting ticks. Furthermore, surface noise was not consistent. Considerable effort was expended in trying to eliminate the occurrence of these defects, but direct proof of their cause was never determined. Some conclusions were reached, however, among them the following:

1. The wax surface was not as smooth as recording engineers thought it to be. In some formulation work, for example, a correlation was found between graininess of the wax surface before cutting and surface noise of the grooves. This surface roughness is apparent in Figs. 2 and 3.

2. The wax formulation was not homogeneous. As a result, under the vacuum and heat of sputtering, localized outgassing occurred which prevented or reduced gold deposition at these spots. When the operation was viewed through

a glass porthole, this phenomenon became evident as localized high-density spots in the glow discharge. Part of the trouble was heat. Measurements of surface temperatures during a normal sputtering cycle showed that the wax surface temperature rose to 180 to 210°F with normal cooling of the mounting plate. Figures 4 and 5 illustrate these points.

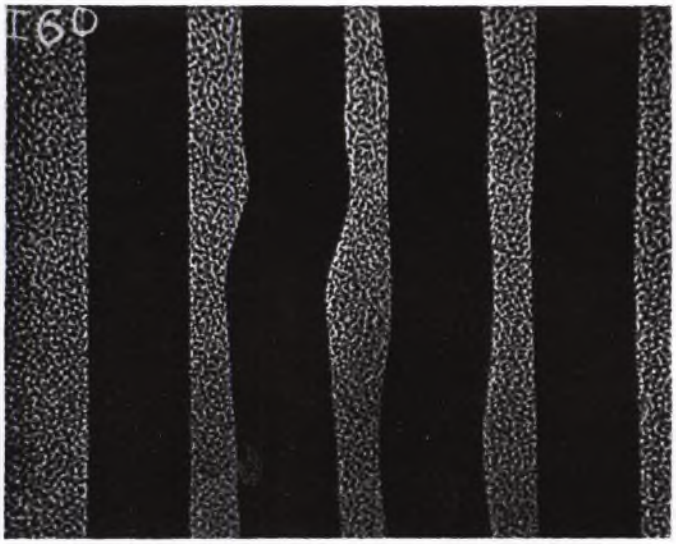

FIG. 2. Gold-sputtered wax recording, focused on land at 250 diameters magnification.

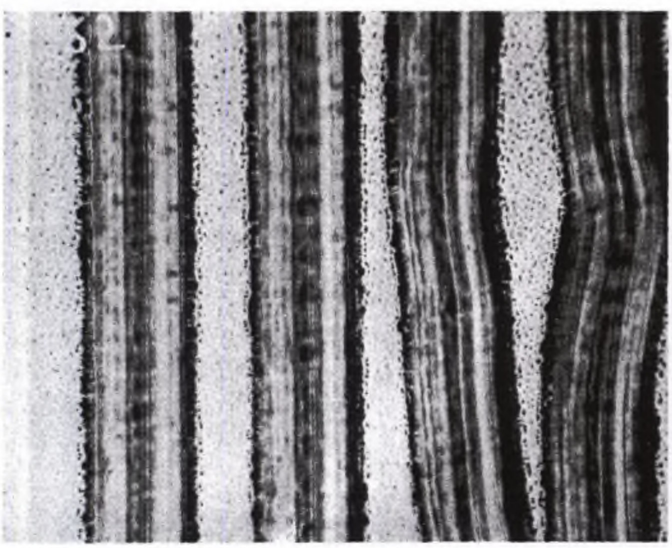

FIG. 3. Gold-sputtered wax recording, focused near the middle of the groove wall at 250 diameters magnification.

Concomitant with the development of gold sputtering of wax, the recording and processing of lacquer discs was being investigated. Lacquer discs have their deficiencies with respect to recording. The rheological properties are such that the groove does not cut smoothly. To compensate for this,

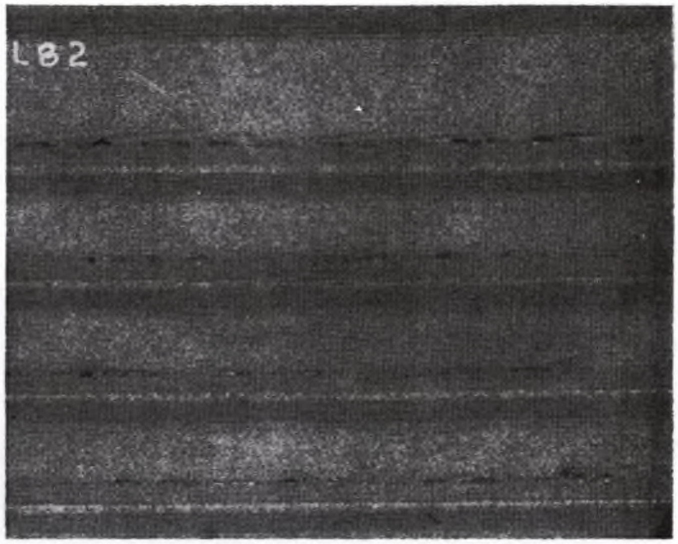

FIG. 4. Copper-preplated gold-sputtered wax, at 50 diameters. Black areas show pores in the copper.

burnishing facets[1] which polish the groove wall are used. These have the disadvantage of producing cross-modulation which shows up particularly as echo. The effect of the burnishing facets also varies with the angle the groove wall makes with the tangent to the radius. Both of these effects, cross-modulation and a rough side wall at sharp wave fronts, are shown in Fig. 6. Another disadvantage is the difference between input level and recorded level, which is a function of linear groove velocity and frequency. Even with these limitations, however, it was found that the lacquer disk resulted in a better record. More consistent manufacturing

FIG. 5. Copper-preplated gold-sputtered wax, at 50 diameters. Light areas show pores in the copper.

[1] I. L. Capps, "The Design of a New Lacquer Recording Stylus," *Audio Eng.*, 32, 18 (January, 1948).

results were obtained with proper metalizing and preplating. The surface noise was less variable, and there were fewer surface defects resulting in ticks, "grit," and "swish." The use of a heated stylus for lacquer recording has minimized the difficulties of cutting lacquers.

Metalizing is extremely important. Any discontinuities in the metal film will not cover over immediately in preplating. If small enough, the metal deposit will bridge over the gap. This bridge represents a pit in the master surface and an unwanted bump in the mold surface. Thus, it will result in a tick if it occurs in a groove. Fortunately, it is comparatively easy to check for adequate metalizing on lacquers. The mirror film can be examined visually under a magnifying glass. Any small discontinuities are easily visible as black spots.

Wet silvering methods are preferred for metalizing lacquers. The high vacuum and heat involved in sputtering was found to be incompatible with the relative volatility of residual solvent and plasticizer in the lacquer film. The method consists in cleaning the lacquer in trisodium phosphate solution or similar mild detergent solution for approximately 5 minutes. Strong cleaners, acids, or high temperatures tend to "dry out" the lacquer, roughen the surface, and thus raise the surface noise. Figure 7 illustrates, for example, the effect of temperature of cleaning on surface noise.

After being cleaned, the lacquer is spray-rinsed thoroughly and immersed in a stannous chloride sensitizer solution. The stannous chloride probably seeds the surface with an adsorbed layer of tin ions, which serve as nuclei for silver crystallization. The lacquer surface must then be rinsed thoroughly to remove excess chloride. The rinse water should be low in chlorides. We prefer to use a strong spray with the lacquer rotating under the spray and finish with a deionized water rinse. Any free chlorides in the water film will

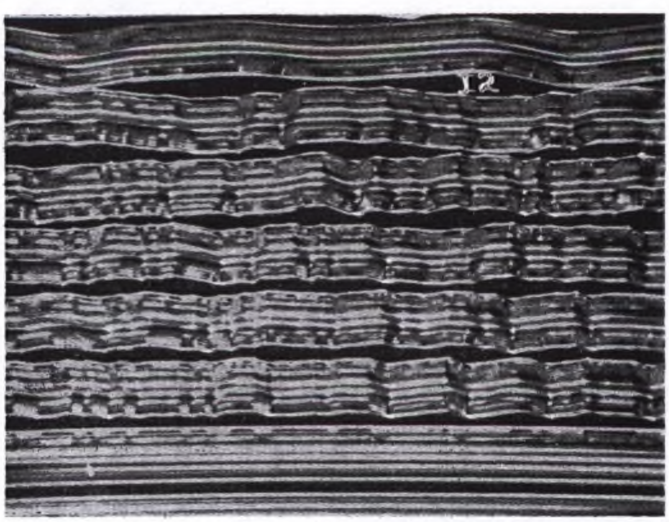

FIG. 6. Lacquer disk recording (1948).

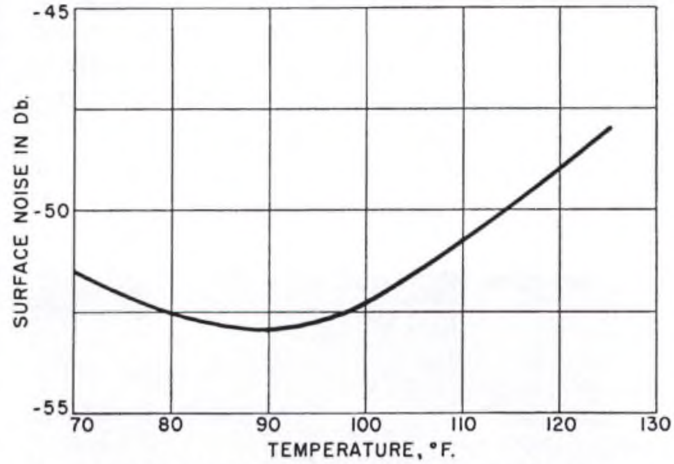

FIG. 7. Effect of cleaner temperature on the surface noise of molds.

cause the silver to be nonuniform, showing up as clouds or stains. After the thorough rinse the lacquer is put in a flat, round tray and clear ammoniacal silver nitrate solution and reducer solution are poured over it. The tray is rocked and rotated until silvering is complete. For tray silvering, a slow deposition rate is desirable. Consequently, chilled solutions and a slow reducer such as dextrose are used. Spray silvering with a two-nozzle gun, one nozzle for the silver solution and one for reducer, is also used. Here, the two sprays converge a short distance from the nozzle, and a fast reducer such as formaldehyde is necessary. After the silvering process, a strong water spray is desirable (replacing a camel's hair brush) to remove any coarse, loose particles of silver which would cause an unsound subsequent electrodeposit.

We have found the silvering operation itself to be relatively noncritical. Preparation of the lacquer, particularly the sensitizing operation, is most important. If, for example, a small air bubble persists in the groove during sensitizing, that spot will not cover during silvering. The sensitizing must be given enough time. If the surface is not sensitized sufficiently, silver may form but it will be easily washed off the lacquer surface. If the lacquer is oversensitized, the silver will stick tightly enough to damage the lacquer surface when the master is separated.

The silvering operation involves a number of steps. Between each step the lacquer must be handled. To reduce the hazards of damaging the lacquer surface and to ensure better timing control of the different phases of the operation, an automatic silvering machine was developed. The machine has spray guns for sensitizer solution, water rinsing, and the silver and reducer spray guns. The lacquer is put in the chamber (Fig. 8) on a turntable. A sequence timer then takes over and starts and stops the operations: rinse, sensitize, rinse, silver, and rinse.

Obviously, metalizing is the first step in the production of a negative replica of the recorded surface. The next is to build a sound structure on the silver, which in the master becomes the supporting structure of the silver surface. Any porosity or coarseness of structure will manifest itself in surface noise, "grit," or ticks. The effect of porosity is evident. In the case of a large-grained structure, the effect is not so apparent. An electrodeposit will begin with a crystal structure duplicating the structure on which it is being formed, in this case the metalized film. This initial layer is reported to be 1000 to 2000 Ångströms thick[2]

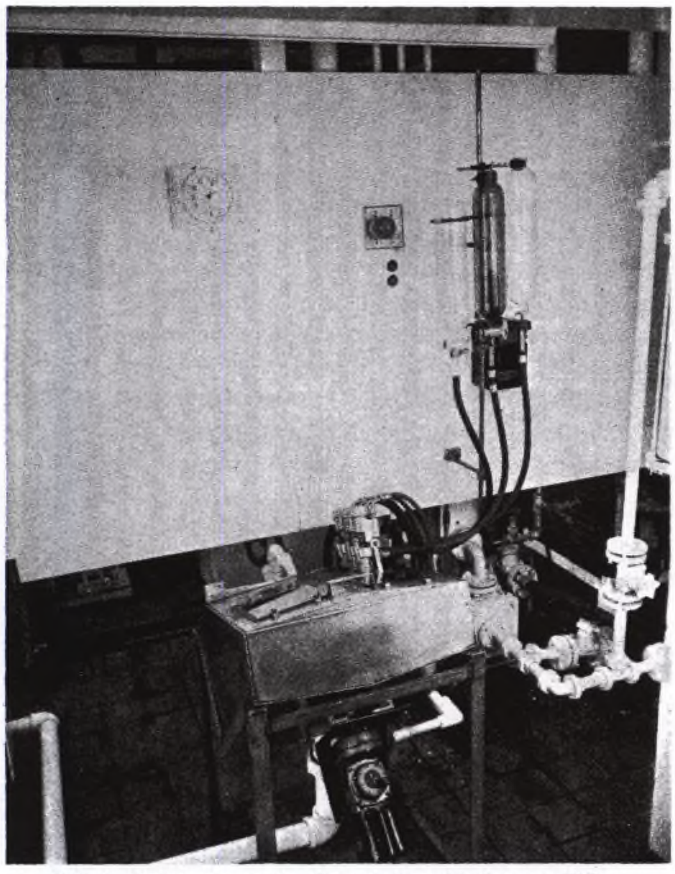

FIG. 8. Automatic spray silver system for lacquer disks.

(3 to 6 $\times 10^{-5}$ in.). Only after this initial layer is deposited does the crystal structure become characteristic of the solution and deposition conditions. However, apparently a coarse-grained structure produces enough localized stresses to impart a roughness to the surface which results in higher surface noise. The solution should, therefore, inherently deposit a fine-grained metal structure to support the silver film.

The initial layer of deposit must spread uniformly over the surface. Where lacquers are concerned, we are aided by the fact that the silver deposits are heavier than gold-sputtered deposits. For example, the best gold sputtering we could do resulted in a resistance of 6 ohms across a 12-in. disk, whereas silver will have a resistance of 2 ohms or less. The thicker film permits the use of higher currents with faster and more uniform coverage.

The requirement of a fine-grained deposit is usually contradictory to that of a good spreading deposit. In acid copper, for example, addition agents are used to obtain a fine crystal structure. These are conceived as interfering physically with crystal growth by adsorption or codeposition. Thus, they tend to interfere with the spreading of the deposit which, on the thin, comparatively high-resistance metalized films, tends initially toward lateral growth. To get good results from copper, a low-temperature (75 to 80°F) purified solution is needed.

With the development of a low-stress nickel solution of low impurity, it was found that nickel produced more dependable results than copper for the initial plating. For example, on tests of copper against nickel with six bands on twenty-five lacquers in each group, molds made from nickel-preplated silvered lacquers had an average surface noise of −54.6 db with a standard deviation of 2.32 db, compared with −50.5 db, 2.29 db standard deviation, for copper preplate. Temperature is important in obtaining best results. Control at 95°F gives best results for lacquer preplating with our nickel solution.

As implied in the above remarks, stress and impurity control are important in the nickel solution, particularly in the Watts type of solution which we use. A normal nickel deposit is stressed in such a way that the edges of the deposit tend to curl away from the starting surface. Obviously, this cannot be tolerated since, with lacquer, the silver is attached by only the most tenuous of forces.

Stress control of the nickel is also important in metal-to-metal duplication, e.g., making a mold from a master or a master from a mold. In the past it was customary to depend on a small amount of adhesion to keep the nickel in place to overcome the inherent nickel stress. If there was no adhesion, parts would split open in the tank and be ruined. If the adhesion was too good, the result might be a master and a mold surface stuck together. The ideal, as expressed by an old-time record foreman, was a squeal as parts were separated, just enough pin-point adhesion to keep things in place. This rupture of pin points roughened the surface sufficiently to increase the surface noise. The cleaning and passivating procedure was obviously very critical, and instructions and traditions were more representative of a voodoo ritual than a manufacturing operation. We can laugh now, but when a load of parts comes out of a tank after 8 hours or so and you find you can get only half of them

[2] G. T. Finch, H. Wilman, and L. Yang, "Crystal Growth at the Cathode," *Discussions Faraday Soc.*, No. 1, p. 144 (1947).

apart, you begin to wonder whether that extra wrist motion isn't necessary.

By tackling the problem at its source, the inherent stress of the nickel deposit, it was found that all the other problems disappeared. The passivating procedure is now drastic enough to ensure no adhesion and, consequently, we now find no noise increase in the music grooves through duplication. This is contrary to popular belief, we know.

In our first work in controlling stress we used the bow of an all-nickel part 0.010 in. thick as an indication of stress. With the development of the spiral contractometer,[3] it became possible to measure the stress. Duplicating solutions are now controlled at 10,000 to 15,000 psi stress. Lacquer preplating solutions are controlled at 1000 to 6000 psi stress. These values are obtained by means of addition agents.

Control of impurities in the nickel solution has been mentioned as important. Copper and iron are particularly harmful, not because they are the only possible deleterious impurities but because they happen to be present. Parts sometimes come off a rack, or nuts and bolts may get into the tanks. These are the obvious sources of impurities. What is commonly overlooked is that the purest form of commercial nickel anodes contains from five to ten times the proportion of these elements to nickel as is desired in the solution.

The effect of these two impurities is twofold. First, they interfere with covering; low spots, such as grooves in molds or "lands" in masters, tend to receive little plating. Second, these impurities tend to make the deposit more porous and brittle. As indicated previously, the sphere of influence of crystal interference is much larger than one would expect. The result is felt on the surface and shows up as a measurable increase in surface noise. Figure 9 shows how this effect can be seen. Heavy deposits over 0.005 in. thick tend to be rough.

In addition, copper and iron make the deposit more highly stressed. With normal equilibrium between anode and solution composition, for example, a stress value for nickel deposits from a Watts solution will average around 22,000 psi. With partial purification, or, more accurately, with present equipment, values between 15,000 and 20,000 psi are obtained. With more recent equipment designed to maintain the purest possible solution, we have achieved levels between 12,000 and 15,000 psi tensile stress without addition agents.

How are the impurities maintained at a low level? The principle of continuous dummying at low-current density is employed.[4] Solution is pumped from the plating tank through a purifying tank, then through a heat exchanger and filter which removes suspended particles, and then back to the plating tank. In the old design, gravity was depended on to settle the solution from the anodes, which has a higher specific gravity, to the bottom of the tank. Therefore, solution from the filter was pumped in at the top of the tank and the outlet was at the bottom. The heating of the solution with outside heat exchangers tends to assist in this downward flow of dirty solution.

Filtration is important. Suspended particles cause roughness which produce enough surface irregularities to raise the surface noise and cause ticks. In decorative plating, filtration equivalent to one turnover of the tank capacity in 2 to 3 hours is considered good practice. Faced with a more critical operation, we turn over the solution two to four times an hour.

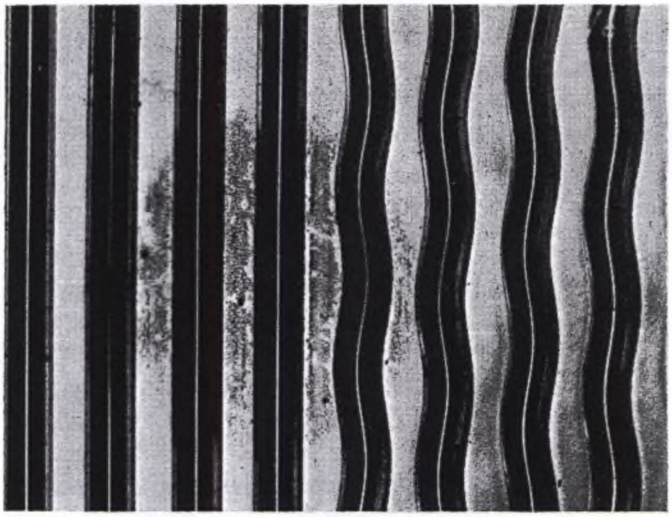

FIG. 9. Surface stain typical of duplication from an impure solution, at 50 diameters.

Newer equipment, presently being installed in one of the plants and designed after more than two years of testing in our laboratories, keeps impurities at the still lower levels. This design is the so-called diaphragm tank[5] in which the cathode solution is separated from the anode solution by a cloth diaphragm. Solution is pumped from the anode compartment through the purifying tank, heat exchanger, and filter, and then back into the cathode compartment. Flow from the cathode compartment to the anode compartment is by gravity through the cloth diaphragm. The trick is to choose a weave porous enough to maintain a difference in level between the two compartments of 1 to 2 in. This

[3] A. Brenner and S. Senderoff, "A Spiral Contractometer for Measuring Stress in Electrodeposits," *Proc. 35th Ann. Conv. Amer. Electroplaters' Soc.*, p. 53 (1948).

[4] B. C. Case, "Modern Applications of Electroplating Solution Purification," *Proc. 34th Ann. Conv. Amer. Electroplaters' Soc.*, p. 228 (1947).

[5] R. H. McCahan, C. E. MacKinnon, and D. A. Swalheim, "Diaphragm Tanks to Eliminate Roughness in Copper Plating," *Proc. 35th Ann. Conv. Amer. Electroplaters' Soc.*, p. 203 (1948).

design physically keeps the less pure solution from the anodes from mixing with the purified and filtered solution in the cathode compartment.

Now it is time to re-examine polishing. As long as various steps in matrix manufacturing contributed to a comparatively noisy surface, polishing was necessary to cover up defects. It is apparent that, if it can smooth over defects in the surface, it can smooth down some of the modulation in the grooves. High frequencies are especially susceptible (Fig. 10). The intermodulation tests made by Roys[6] show another manifestation of the ill effects of severe polishing. Figure 11, a microphotograph taken 8 years ago of some stampers rejected for "distortion," shows an extreme case. On the side walls, modulation appears which fades out at the top and bottom of the groove.

Polishing was a cumulative process. Because of the defects in the gold surface, the gold-faced master had to be polished. Then, to get a uniform surface, the surface was face-plated with a thin layer of nickel, which was again polished. This nickel surface was duplicated to make a mold, and the mold surface was again polished. After every stamper the mold was polished as well as the stamper. It was, therefore, not surprising, that, with every part made, further deterioration occurred. The common conception that, the closer the stamper was to the master, the more faithful was the reproduction of the record was true. Today this need not be true. In fact, we have made tests with present methods of matrix manufacture of as many as eight generations with no loss in frequency response and a negligible increase (1 to 2 db) in surface noise between the original mold and the eighth descendant of that mold.

Another common misconception is that the first pressing from a set of stampers is better than subsequent pressings. This may have been true with 78 rpm records where an abrasive filler is used in the plastic. With present unfilled plastics, wear of stampers is not a problem. No difference in frequency response or noise can be detected between the first and the thousandth record from a set of stampers.

It should be apparent that to cover the whole story of manufacturing changes in the past ten years would require considerable space and time. However, in this brief review, the key developments in reducing the spread between the recording process and the record ultimately delivered to the consumer have been presented. Obviously, these developments are not the work of any single individual or group, but represent contributions from the recording and manufacturing personnel as well as workers in development engineering, all working toward the goal of giving the record buyer better reproduction and more music for his money.

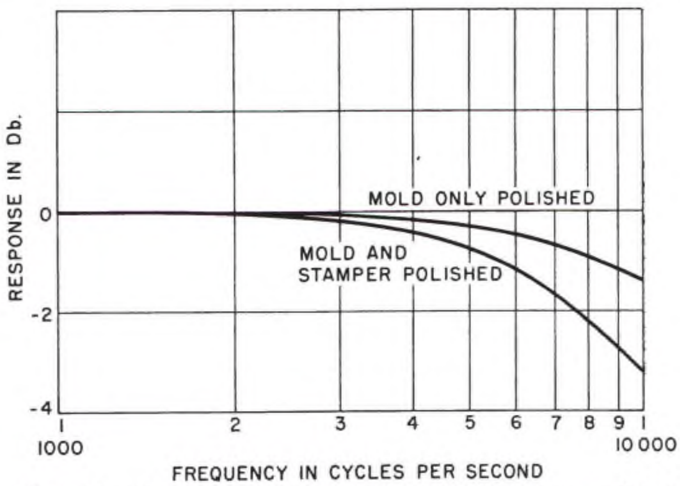

FIG. 10. Frequency response of records made from polished matrices.

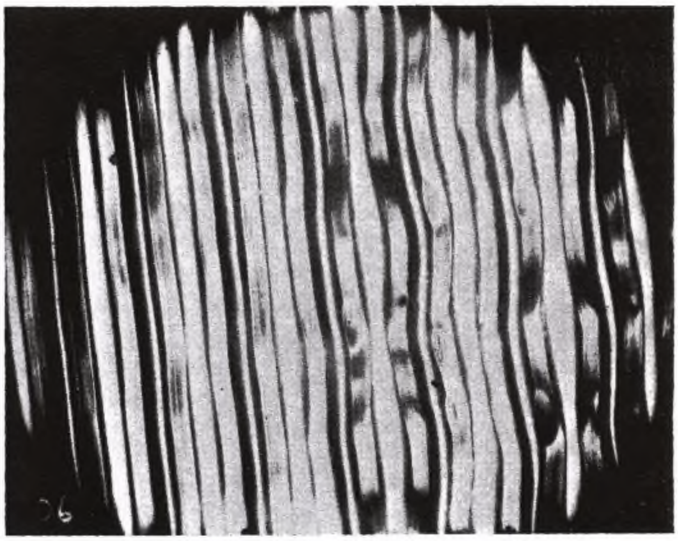

FIG. 11. Severely polished stamper.

[6] H. E. Roys, "Intermodulation Distortion Analysis as Applied to Disk Recording and Reproducing Equipment," *Proc. IRE,* 35, 1151 (1947).

Control of Static Electricity on a Phonograph Record*

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A method of making phonograph records permanently anti-static is disclosed. Quaternary ammonium salts as a family of compounds were observed to be quite effective in reducing surface resistivity, and vinyl plastic containing American Cyanamid's *Catanac SN* produced phonograph records which demonstrated practically complete absence of a static charge. A major quality improvement resulted from this process since static-free records do not attract damaging lint and dust particles.

Plastic materials normally employed in the manufacture of phonograph records can be given very high static charges: the surface resistivity of these plastics will usually measure more than 1×10^{13} ohms. Such static charges attract and hold damaging particles to the record surface. The problems associated with these dust particles have been all too apparent to the record connoisseur since the introduction of long-playing records, and each improvement to the product has accentuated this condition. Now a satisfactory method to reduce this static build-up has been developed. This has made available to the consumer high quality stereo records which will not be degraded by damaging static-held particles.

The normally highly charged surface of a phonograph record attracts many kinds of damaging particles. Generally they consist of silica particles, cellulose fibres, carbonaceous material such as soot, and other miscellaneous types of dirt and contamination. Because of the high static charge these particles cling to the record surface so tenaciously that conventional cleaning methods do not remove them.

The handling and playing of lint- and dirt-laden discs puts scratches into the playing surfaces. This damage changes the sound reproducing ability to the point that otherwise good records lose all aesthetic value to the listener.

By those not associated with disc quality problems, heterogeneous audio noises may be blamed upon the discharge of static electricity from the record surface; those well-versed in the profession, however, recognize that such noises are not caused by static electricity. Instead, these noises in part originate from groove defects resulting from attracted dirt particles being ground into the disc.

Records acquire their first strong charge during the molding operation. Hot plastic material is formed in a mold cavity under heat and pressure to create a record. Most of the heat is then removed from the mold cavity and the plastic disc, after which the mold opens releasing the record. At the instant when the record surface separates from the mold surface, a strong charge is induced upon the record.

Secondary charges, usually of a lower order of magnitude, are created on a record during the normal handling given it by the user. By merely pulling the record out of its jacket the user can sometimes create enough static buildup to attract damaging dust particles.

Over the years the art and science of record making progressed, resulting in monaural discs of improved quality. Finally, truly superior stereo discs were introduced. The problem of dust and lint attraction took on greater significance because of the narrower grooves encountered with these disc improvements. To make the problem still worse, refinements were made in record formulation to reduce the inherent surface noise. These changes tended to raise surface resistivity, making dust attraction worse.

It was felt that a permanent solution to the control of static electricity should be created to give the consumer the ultimate in record performance.

The problem was recognized by many people quite early in the LP program, even before stereo. This led to the commercial introduction of a number of materials designed to give at least a temporary relief from these difficulties.

First in this category was the large group of wetting agents or detergents which could be mixed in small quantities with water. This usually amounted to a mixture of one of these materials in water at the rate of about a teaspoonful to a quart, giving about a 1% solution. A lint-free cloth moistened with such a solution cleaned records quite effectively and imparted some anti-static effect by leaving a very thin conductive film on the record surface. The anti-static effectiveness of these coatings was limited to relative humidity values above 35%. They had no lasting value because they could be easily removed by wiping with a cloth moistened with plain tap water.

Other devices depended upon air ionization to reduce the static charge on a record. Materials like polonium were used to neutralize the static charge, permitting the dust and lint particles to be brushed from the disc.

* To be presented October 16, 1962, at the Fourteenth Annual Convention of the Audio Engineering Society, New York.

Work on anti-static records started at RCA soon after the introduction of LP's. The work was broken down into three categories: 1. improve surface conductivity by applying conductive films to record surfaces; 2. mix into the record material a conductive carbon black; 3. reduce surface resistivity by the addition into the plastic of a chemical additive which would impart anti-static properties to the molded discs.

Work on the first category led to experiments with conductive films of such materials as titanium tetrachloride, platinum chloride and tin chloride. None of these proved effective.

U. S. Patent 7,727,831 disclosed a method of sulfonating a polystyrene surface to decrease surface resistivity. Since polystyrene is a member of the vinyl family, the same techniques were investigated on normal copolymer vinyl resins used in record production. Sulfonation did not occur with short exposures. However, time intervals on the order of 20 minutes at temperatures of 120°F to 140°F reduced surface resistivity, indicating that sulfonation might have occurred. The copolymer vinyl resins, however, decomposed under this severe treatment of concentrated sulfuric acid. It is felt that the decomposition products may have made the contribution to the reduction in resistivity values. The process was obviously not considered to be practical.

Surface-active agents were given considerable attention as disc coating agents. Table I lists several materials evaluated. Records were treated by dipping them in 0.1, 1.0 and 10 percent solutions of the various materials, followed by spin drying. Surface resistance was measured using the method described in ASTM Designation D-257-52T.

TABLE I. Surface resistivity of records coated with various surface coating agents.

Product	Relative humidity	Concentration in water 0.1%	1%	10%
		Surface resistivity, ohm		
Niatex AG-2	10	5×10^{14}	6×10^{12}	9×10^{9}
	52	3×10^{13}	5×10^{11}	2×10^{9}
Tergitol nonionic TMN	10	7×10^{15}	3×10^{15}	8×10^{12}
	52	7×10^{14}	3×10^{14}	8×10^{12}
Tergitol nonionic NPX	10	8×10^{15}	4×10^{14}	6×10^{12}
	52	6×10^{15}	8×10^{14}	7×10^{11}
Tergitol nonionic NP-27	10	1×10^{16}	5×10^{15}	2×10^{13}
	52	1×10^{16}	6×10^{14}	3×10^{12}
Carbowax 200	10	1×10^{16}	1×10^{16}	1×10^{13}
	52	1×10^{16}	8×10^{14}	1×10^{13}
Propylene glycol	10	1×10^{16}	1×10^{16}	1×10^{16}
	52	1×10^{16}	1×10^{16}	1×10^{16}
Ucon lubricant 50-HB-3520	10	1×10^{16}	3×10^{14}	2×10^{13}
	52	8×10^{15}	1×10^{14}	3×10^{12}
Ucon lubricant 50-HB-55	10	1×10^{16}	1×10^{16}	8×10^{14}
	52	1×10^{16}	1×10^{16}	8×10^{14}
Zelec DP	10	1×10^{16}	1×10^{13}	4×10^{10}
	52	8×10^{15}	6×10^{10}	1×10^{9}
Zelec DX	10	4×10^{15}	3×10^{12}	5×10^{10}
	52	1×10^{14}	2×10^{11}	9×10^{8}
Zelec NE	10	6×10^{15}	2×10^{17}	3×10^{9}
	52	3×10^{15}	6×10^{11}	8×10^{8}
Zelec NK	10	1×10^{16}	7×10^{10}	4×10^{9}
	52	6×10^{13}	3×10^{10}	6×10^{8}

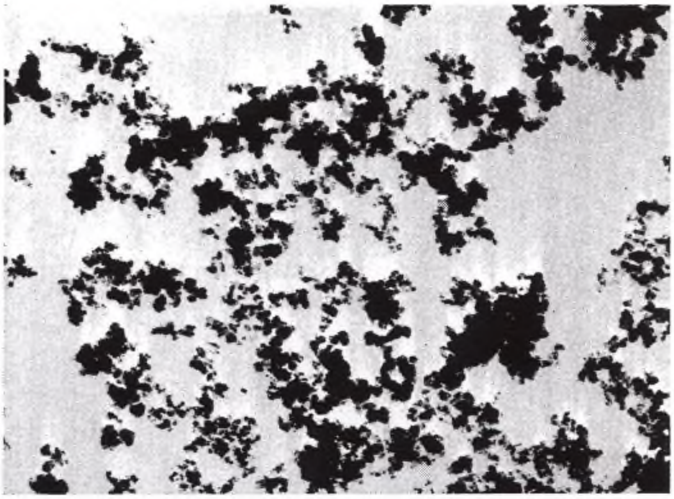


FIG. 1. Electron micrograph of high-structure carbon black (enlarged 50,000 times).

The surface resistance at which a coating is effectively anti-static at a given relative humidity is not specifically known. The literature contains references ranging from very low to fairly high values. We assumed for our work an arbitrary value of about 10^9 ohm as being low enough to produce desirable anti-static properties for phonograph records. Using this as a tentative standard, none of the materials listed in Table I were effective in 0.1 percent solutions at low humidity. Coatings of *Zelec NK* (E. I. duPont Company) from 1.0 percent solutions were anti-static. However, these coatings stained record surfaces badly and had no permanence since they were wiped off readily with a damp cloth. This work led us to believe that a coated record would not give the permanent high quality anti-static discs desired.

Method two attacked the problem by attempting to reduce the volume resistivity of the plastic through the incorporation of a conductive carbon black. Acetylene blacks or high-structure furnace blacks were employed in this work, as the chain-like structure of the black particles is believed to aid in bleeding off a static charge. An electron microscope view of this type of carbon black is shown in Fig. 1. Tests showed there is a threshold loading that must be reached before any conductivity can be obtained; this value appeared to be around 13 percent (by weight) loading in the plastic material. From this concentration upward, further loading reduced volume resistivity as shown in Fig. 2. Many problems developed with the use of this type of composition: 1. it was extremely difficult to mix in a banbury mixer or on a two-roll mill; 2. the presence of the carbon black reduced the strength and flexibility of the plastic composition below the normally accepted safe values; 3. the audio properties of discs made from such a mix were poor because of a steady hiss resulting from the presence of the carbon black particles or agglomerates in the plastic, and 4. it was extremely difficult to mold such a composition into perfect LP records.

Anti-static properties were very good. Surface resistivity values of 9×10^6 ohm were measured. In spite of the good anti-static properties, this method was rejected as not being commercially feasible.

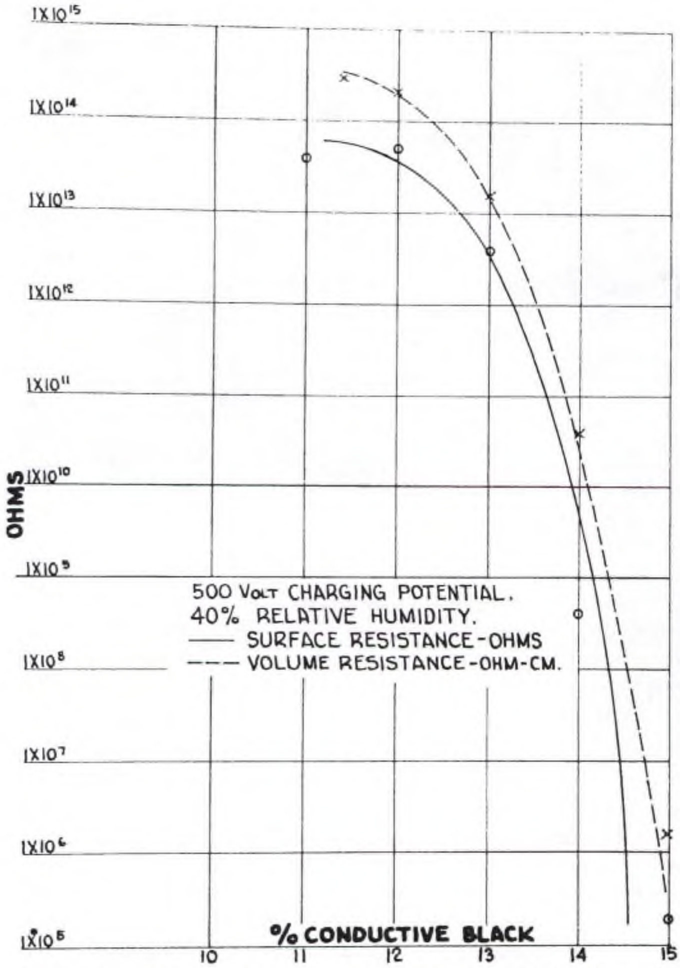
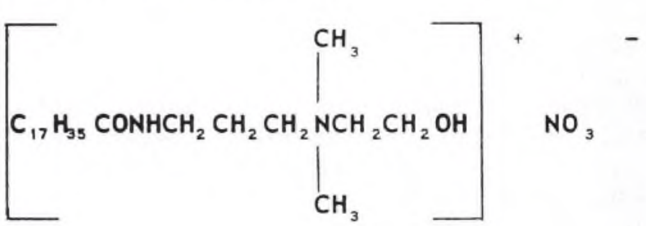


FIG. 2. Surface resistivity of record compound loaded with 11 to 15% conductive carbon black.

Method three showed the most likelihood of success. More than one hundred different materials were tested. Each material was mixed into the vinyl plastic in concentrations ranging from 1 to 5 percent. It was established rather early in the work that a quaternary ammonium salt gave the best compromise between compatibility and limited bleed-out to the disc surface to produce the anti-static effect.

Greatest success was observed with a family of cationic materials supplied by the American Cyanamid Company. *Catanac SN* was selected as being the most effective member of this family of compounds. American Cyanamid disclosed that its structure is as follows:



stearamidopropyldimethyl- β -hydroxyethylammonium nitrate

A concentration of about 1.3 percent by weight mixed into the resin seemed to give optimum results. Tests in the factory, however, showed it could not be added directly in the banburys to get a sufficiently uniform mix to produce good records. Stains were encountered on the record surface and very spotty noises in certain portions of the record indicated that a better mixing procedure was needed.

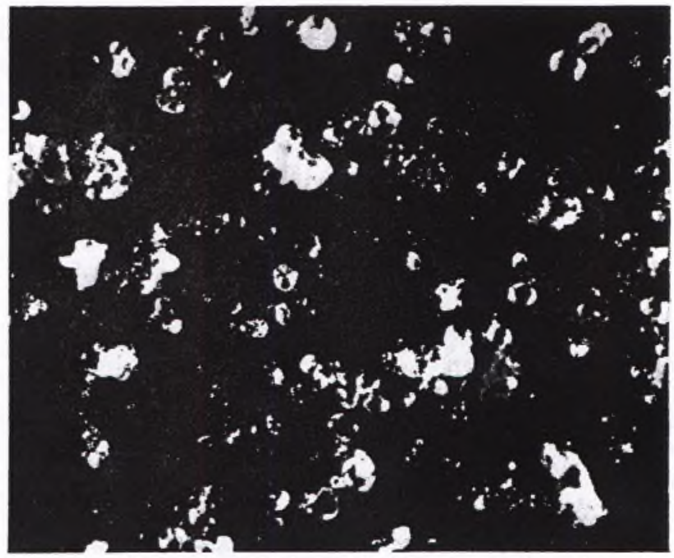


FIG. 3. Vinyl resin particles exhibiting poor surface absorption properties.

The leading vinyl suppliers were called in and shown the problem. We requested that they study their manufacturing process and determine whether or not they could coat their resin with the agent to get a uniform coating on each particle of material. In time, each manufacturer achieved a certain degree of success in this work.

It was found that particle shape played a very important part in the problem. Those resins having a glassy bead type of particle, as shown in Fig. 3, absorbed the agent on

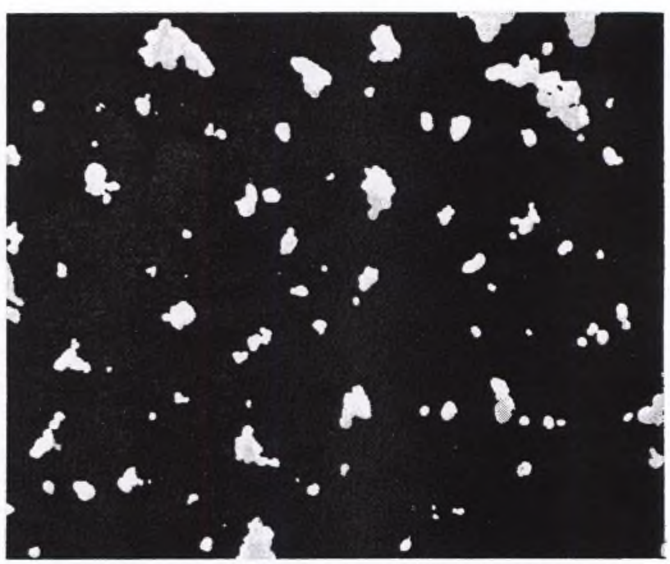


FIG. 4. Vinyl resin particles having a highly absorptive surface.

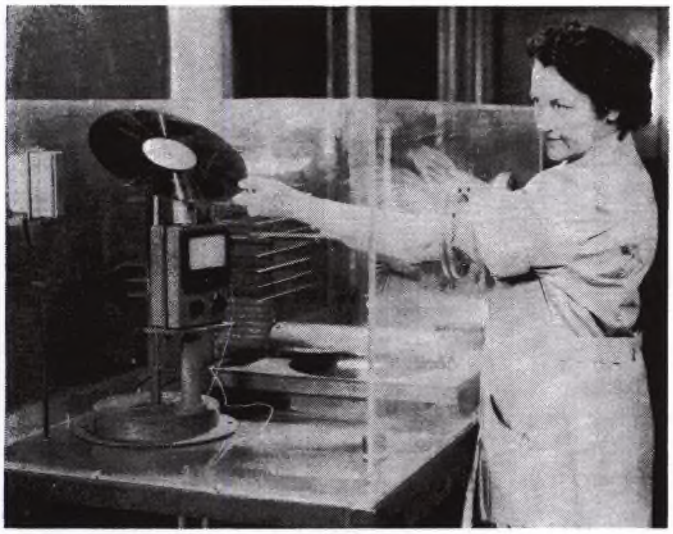


FIG. 5. Electrometer static record testing.

the surface rather poorly. This resulted in blocking or caking of the mix, causing it not to flow properly in bulk handling equipment. A more porous type of particle as shown in Fig. 4, however, produced a dry, free-flowing powder that processed without difficulty. Practically all of the major resin suppliers can now modify the particle size and shape of their resin to meet these requirements.

The first production lots of coated resin were made available for test in early 1959. Results were promising but refinements were needed to put the product on a commercial basis. A task force of three men was assigned to the project —one representing RCA, one representing the supplier of the anti-static agent, and one representing a vinyl supplier. Using the full support and know-how of each organization, a crash program was set up to solve the remaining problems and carry out refinements necessary to make a fully commercial product.

A series of runs were made to determine the upper and lower control limits of the *Catanac SN* content; banbury cycles and processing temperatures were studied to determine their effect upon the anti-static properties; formulation modifications were made to determine the effect of different types and concentrations of carbon black and heat stabilizers upon the anti-static properties.

Actual production started by July 1, 1959, and has continued to the present time.

Testing and evaluating the static properties of records must be carried out under carefully controlled conditions. Figure 5 shows the test chamber devised to accomplish this. A strong charge is induced on a record using a corona discharge; the strength of this induced charge is then measured with an electrometer and the decay time observed. An effective anti-static agent should cause a strong charge to dissipate within 2 or 3 minutes. Records produced from this product are quite anti-static over a range of relative humidity conditions from 20% on up. Figure 6 graphically demonstrates the improvement by comparing the difference in the attraction of cigarette ashes for an untreated and treated record.

FIG. 6. Cigarette ash test comparing treated and untreated records.

Thus, a record so treated will not attract and hold lint and dust during storage and use. Furthermore, a record may be easily cleaned of any lint and dust which might settle on it when left lying unprotected.

From a sound reproduction standpoint, anti-static discs are of the highest quality. If handled properly they will remain in that condition over a long period of time because of their ability to remain clean.

THE AUTHOR

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He joined RCA in 1946 as an engineer in the Record Compound group and in 1947 he was appointed to group leader. In 1956 he was made Manager of this group. He is a member of the American Chemical Society and a member of the board of directors of the Central Indiana section of the Society of Plastics Engineers.

Record Contamination: Causes and Cure*

PERCY WILSON
"The Gramophone," London, England

INTRODUCTION

THE modern high-grade stereo cartridge reacts to contaminated records in ways that went more or less unnoticed with lower-grade types. This is due basically to three characteristics: a diminutive stylus of low mass, a relatively high compliance in both planes, and a low tracking force arising primarily from the necessary relationship between stylus tip radius and the elastic limit of record material.

Trouble due to surface contamination shows up mainly in spurious noises becoming more obtrusive, one factor concerned being the improved transient response designed into the cartridge. Certain types of contamination also result in high-note loss, the same condition often causing styli to get so clogged with collected deposit that they eventually fail to transduce. Owing to high transient response, and if it is equally matched at the acoustic transducing stage, a tiny stylus colliding with a particle of grit in the outer grooving, where the impact velocity is greatest, can thus produce quite shattering noises at times.

Under the prevailing atmospheric conditions, high-note loss sets in the moment even a new record is played, becoming more identifiable at each subsequent playing. People are often disappointed when they change over to an expensive high-grade cartridge, perhaps with a new arm and other new components to match. A stereo record, first time out of its sleeve, may and often does sound brilliant and noise-free, but seldom is this experience repeated if the person is a city or town dweller. The odd noise or two usually appears on the second playing, more on the third, and so on. The owner tries to get rid of the noises, often succeeding only in making matters worse.

NATURE OF RESEARCH UNDERTAKEN

A detailed research has been undertaken by the author and his associates during the past three or four years to determine the physical nature of record contamination, how it arises, why existing attempts to avoid it, whether by ingenious brushing arrangements or by anti-static dopes, had failed. This research has produced some fascinating explanations of a number of apparent anomalies. It has also led to a firm conclusion that, provided a record is not scratched or otherwise ill-treated, its effective life can be extended indefinitely: it can now be played *ad infinitum* with a clarity, crispness and brilliance and an absence of surface noises that only a record straight off the press normally achieves.

The investigation started with the assumption that the requirements for cleanliness were: *1.* during the playing of the record there should be no unrelated and sporadic "pops and crackles" (a regular repetition of clicks would most probably be due to a scratch, though it is recalled that on one occasion it was found that a sliver of chewing gum was responsible); *2.* microscopic examination of the stylus after a record (or parts of it) had been played should reveal no collection of foreign matter on the tip.

It was also hoped to develop cleaning techniques that would reveal the nature of any contamination. This led to many frustrations but in the result it has become possible to distinguish accurately between various types even in minute degree. Thus, when someone blew three mouthfuls of tobacco smoke through a metal tube (so as to avoid spittle—never blow on a record, by the way, to remove fluff!) about 1 foot above a clean rotating record, our latest technique revealed the tobacco tar as a brown stain on a linen pad.

TYPES OF CONTAMINATION

Record contamination can be divided into three categories, viz., discrete particles such as soot, dust, grit, textile fluff, tobacco ash and cosmetic powders; fluids which, after evaporation, leave behind crusty deposits, such as household sprays and saliva deposits; and condensates formed from fumes, such as tobacco smoke, cooking vapors and automobile exhaust fumes. These three types of contamination are usually met in combination, thus resulting in mixtures that are anything but easy to remove safely. The mixing is in any case started by stylus traverse itself, but

* Presented October 13, 1964 at the Sixteenth Annual Fall Convention of the Audio Engineering Society, New York.

the greatest effect is when grooving is wiped with a cloth that is not quite dry, whether before or during playing.

STATIC AND ANTI-STATIC

Throughout, we have of course been fully apprised of the problems of electrostatic attraction and have taken full precautions to minimize it. One thing, however, became abundantly clear during the early part of the investigation: the treatment of the record surface *with any material whatsoever* that leaves a deposit increases the liability to noisiness within a few months. Detergents in particular (whether anionic or cationic) will produce a battery of pitted sections of the groove in about 12 months' time. We surmised, though we have not investigated this thoroughly, that the action was not so much on the vinyl chloride as on the lead stearate or other additives that go to make up the modern disc material.

An interesting experiment in this connection, which I mentioned before, is the following: I electrify both surfaces of a disc by rubbing with a piece of silk and show by this primitive but effective electroscope how powerful the static is. I now place a piece of aluminum foil on one surface and the net charge on the other surface is neutralized. I remove the foil and the charge flows back again.[1] Hence the value of a conductive turntable mat which has just been introduced in Britain. It has the same effect as the aluminum foil in neutralizing the surface field on the opposite side of the record, so that the problem of static attraction while the record is being played is solved.

Now a record should not be anti-static while it is removed from the turntable. Any static charge can be used to *protect* the record from finger marks while it is being handled or from scratches while it is being inserted into or withdrawn from its sleeve. The method is simple and needs only a piece of thin, light but stiff tissue 12 in. by 24 in., folded over to make a (double) 12-in. square and with the two corners cut off at the fold, with the cuts sealed on the angles.

The record can then be inserted within the fold without sliding, and the tissue will gently adhere to it on both faces, and so protect it from damage, even during the act of sliding the whole thing in and out of an inner record sleeve—folded end first.

The only safeguard that then needs to be added is a protection from finger marks while the record is being taken from the turntable and until the tissue is folded over it. The most convenient way of doing this is to use a small folded piece of card (old Christmas cards are the right thickness) between thumb and first finger.

AERODYNAMIC ACTION

The research to be described in this paper, however, goes far beyond static effects. The fundamental nature of the problem can best be illustrated by another simple experiment.

Set a record rotating at 33 rpm on a massive turntable, preferably one of considerable thickness. Then get a piece of rubber or plastic tubing and fix a metal tube in the end. Gently puff tobacco smoke through the rubber tubing and direct it (as cooled to room temperature by the metal) above the middle or thereabouts of the record. You will find that the smoke will be drawn down with a vortex-like suction towards the rotating disc, will swirl round with the disc and will then be drawn close down over the edge of the turntable.

That is what happens to the air whenever a record is played: any suspended particles are irresistibly drawn down to the record surface. I will mention some of the consequences of this, as affecting turntable design, later in this paper. At the moment, I want to call attention to the fact that this phenomenon completely explains why "pops and crackles" arise in a record mostly at the outside grooves and then at the middle of the recorded surface. Between the two there is commonly a quiet patch, as there is also nearer to the label. In confirmation, I may add that our cleaning process actually finds most contamination in those two locations. The reasons are that there the vortex lines of force, as it were, change their direction; and for the material particles there will be an overshoot—particularly if the surface is statically charged.

But it is not only the larger particles of grit and dust that are affected in this way, as the smoke film shows. All fume condensates, whether from smog or tobacco smoke or from household fumes—the stuff in fact that smears window panes and your car wind-screens—are pulled down with even greater effectiveness since their mass is smaller relative to the vortex forces.

One practical conclusion is immediately apparent: never leave a record rotating on a turntable any longer than is necessary.

FUME CONDENSATE

But there are more difficult things to follow. The fume condensate unfortunately readily adheres to the record surface, and because of the physical dimensions and minute turbulences involved, even to the walls of the groove and especially to the high-note modulations. It forms a sticky deposit which gradually hardens. Particles of grit of microscopic dimensions become embedded in it and eventually adhere so firmly that ordinary brushing action cannot dislodge them. But they *can* be dislodged, as we shall see.

This fume condensate is the most difficult form of contamination to combat. Ideally, the right course is never to allow fumes of any sort to come in contact with record surfaces, but this is more or less impracticable certainly so far as external causes are concerned. Even in air-conditioned houses, fumes do creep in from outside at times. As for internal causes, it is surprising how much condensed tobacco tar can form and build up on a repeatedly-played record: a change from a larger to a smaller stylus can result in the latter getting clogged on one playing only.

The worst condensates form in big industrial cities and other smoggy areas where there is usually a high proportion of soot in the air. In these cases, one does not have to

1. I am indebted to my son, Professor Richard Wilson of Harvard University, for a theoretical explanation of this effect, which is presented in the *Appendix*.

change from a larger to a smaller stylus for trouble to evidence itself. Stylus traverse alone mixes the two contaminates together, producing deposits on the shank that defy dry-brushing and which only some wet-cleaning process will remove.

HIGH-GRADE AND LOW-GRADE CARTRIDGES

An interesting point arises here. Why do such troubles evidence themselves more with high-grade than with lower-grade equipment? One reason is that with the latter the speaker or amplifying equipment may mask the effect, but usually this is because of the cartridge. First, its inherently poor transient response makes it relatively deaf to spurious noises, or at least it takes the "bite" out of clicks caused by grit. Secondly, the ratio of condensate film collected to stylus mass is so relatively low that it does not become a bugbear: with its higher tracking force, the stylus just ploughs through the film each time, so maintaining permanent contact with the groove walls. The deposit keeps being pushed up the stylus shank which, being so relatively long, can store quite an amount before anything untoward happens. This same ploughing action also overcomes a lot of sporadic noise trouble; what with higher tracking force and lower compliance, particles are pushed out of contact with walls, likewise becoming stored up at the back end of the shank.

The difference in mechanical behavior between low-grade and high-grade cartridges then is that, with the latter, the stylus tends to jump every particle encountered no matter its size, so producing noises of various types; and because of the combination of high compliance and low tracking force, the stylus tends to ride up and into blobs of deposit collected along the groove walls.

Smog film in particular is responsible for a lot of trouble when, for example, a person buys a high-grade cartridge and uses it on old mono records in order to get the best out of them. They often do sound better to begin with, but, as likely as not, the cartridge is put out of action well before the end of playing. This is due to a dust and film mixture never before disturbed by a larger-radius stylus and which past cleaning processes had never extracted.

DRY CLEANING AND WET CLEANING

An unfortunate aspect of all films (when allowed to build up) is that no form of dry-cleaning will remove them without damaging the grooving; wet-cleaning has to be resorted to, in a way that ensures every vestige of resultant sludge being extracted.

For maintaining records in first-class condition so that justice can be done to them with good modern equipment, new techniques for both dry-cleaning and wet-cleaning have therefore had to be evolved.

DEFICIENCIES OF EXISTING TECHNIQUES

The evolution of these techniques was helped largely by first studying existing general methods, their shortcomings, and the design rules they prompted. Typical findings were:

1. Special cloths or tissues of the dry-cleaning type can seriously impair record surfaces. Grit held in the fabric can produce scratches that a high-grade cartridge will register and impregnations in the cloth can result in particle mixtures becoming bonded to the grooving. Moreover, seldom will a cloth of any type remove every vestige of loose particles in the grooving, let alone particles that have become stuck to the walls.

2. Though highly effective in some cases, dry-cleaning devices of the "tracking brush" type have their shortcomings. So far as dry loose contaminate is concerned, they do collect and hold it but when their collection capacity is exceeded, they start shedding some of it back on to the surface. As for film contaminate, they do pick up some of it, which is why brushes develop black tips, but they also leave some of it behind, at the same time mixing it up with any particles remaining in the grooving. This is the reason why records sometimes sound noisier when such devices are used with highly sensitive stereo cartridges.

3. A repeatedly-used damp pad or like device can become a positive menace. Under circumstances by no means infrequent, it can consistently make surfaces far noisier than without such treatment, and can result in severe stylus-clogging. These devices themselves get clogged with grit, dust and film, transferring back more than they ever pick up. Periodic washing is no remedy.

4. Some users, in attempts to get pads clean again, resort to soap or detergent. It is extremely difficult to remove all soap or like residue and more often than not traces of it remain in the pad. The result is that a certain amount gets transferred back to records then treated. After playing them, a tiny stylus can become obliterated by fine soap shavings scraped off the groove walls.

5. Certain types of detergent are particularly troublesome. They leave a deposit on styli which, if not soon removed, hardens and defies subsequent removal. I have already commented on the long-term effect on records.

6. Much research has gone into suitable fluids for wet-cleaning, particularly for cases where records have been allowed to get into a bad state. For the majority of cases no more than distilled water is needed. For the tougher cases, a variety of fluids has been tried, but up to now the only safe and satisfactory one found is a dilution of ethyl alcohol (say 25%) in distilled water. There may, of course, be other fluids equally safe and satisfactory, but they have not yet come to light. As for cleaning smog and like deposits off styli, pure ethyl alcohol applied with a miniature "stroking" brush has been found effective and safe, certainly with all modern ways of fixing styli to cantilevers.

7. Special problems arise with records given anti-static treatment after leaving the factory. It sometimes gets applied to surfaces not spotlessly clean, having been subjected to fumes in the meanwhile or the surfaces handled with the bare fingers. Imperfect bonding of the anti-static film thus occurs and, in time, it will start to flake off here and there under stylus action, so producing sporadic noises and collecting around the stylus and up the cantilever. When this happens, there is no option but to remove the film, an operation involving considerable time and patience if groov-

ing is not to be impaired by rough or careless treatment.

8. Records sometimes develop loud clicks which do not respond to any normal cleaning treatment, the owner usually attributing them to factory defects. This sometimes is the case but it is often due to a grit particle with a bonding area so relatively large that loosening fluid cannot penetrate right through it. The grit thus remains stuck. A way has been developed for dealing with such cases without damaging the groove walls. The principle is to apply a nearly-lateral force to the particle, with next to no vertical force applied to the record surface.

PRACTICAL CONDITIONS

The next stage is to consider the principles and rules arising from the findings just quoted. To begin with, the odds are against a record remaining in pristine factory condition right up to the stage of final purchase. This is on account of the handling and subjection to fumes between leaving the factory and coming to the owner's possession. It is the reason why no valuable record should be played without the user first examining it carefully under a strong light, and until in any case he has at least dry-cleaned it properly. The only way records can be assured of reaching purchasers without risk of "intermediate" impairment is for them to be packed in sealed sleevings, such as polyethylene bags, before leaving the factory. The purchaser would then be the first to break the seal. There is rather a counterpart to this procedure in that, when young people buy a current popular record, then often rush into a shop, buy the record without examining it, and then go straight home to try it. This is the reason why most pop records, even when played on high-grade equipment, are usually completely noise-free on the first playing. No better way of buying a record could be devised. It provides a moral to the more serious enthusiast who often examines his record in the shop and tries it out on damaged shop equipment.

Considerable attention has been given to ways of preventing fumes condensing on record surfaces. A little headway has been made, involving amongst other things a revised type of turntable as referred to later, but the problem is likely to remain for some time ahead. Much the same applies as regards airborne particles. In the meanwhile, the only way out is periodic cleaning, the "dry" method for removing loose particles and the "wet" one for removing stuck particles and condensed film. The simpler of the two of course is the "dry" process, where grooving in the stylus path can be progressively made clean. Our more recent experiments show that this can be made 90% effective.

CLEANING DURING PLAYBACK

The requirements for an ideal particle-removing device on the playing deck are as follows:

1. The method should be proof against groove-wall damage in any way, not even impairing to the slightest degree the surface sheen of modern pressings.

2. Every vestige of removable contaminate should be extracted.

3. The device should never shed any collections back onto the surface no matter how much is encountered, a requirement particularly relevant to dusty areas where there are high winds.

4. The device should have no effect whatever on record speed.

5. The action of the device should never produce noise in the pick-up.

The most effective way of applying these requirements is with a tracking-type suction head with certain ancillaries. Evolving the design was no simple and straightforward matter. For instance, mere suction, regardless of the air-flow rate, is not enough; it has to be supplemented by other actions. Perhaps the greatest problem of all was in keeping the loading low and also the frictional characteristics. The pressure exerted on the record surface is of the order of 0.25 gram, the force remaining stable even when the device is applied to severely-warped records.

The device has certainly come up to expectations. The most effective tests were made during the past London winter in a district notoriously prone to high soot content. The device embodies a filter, of course, so that extractions do not get back into the suction generator. The filter, initially a wad of white material, turned almost jet black on occasions after only a few hours of record-playing, the stylus emerging at the end of each run with a deposit consisting only of smog film and not a soot mixture. The difference amounted to a slightly-blobbed tip instead of a collection extending almost to the end of the cantilever, which regularly happens under such atmospheric conditions without the suction device in operation. The main criterion, of course, is whether it does effectively stop or at least appreciably reduce sporadic noises caused by grit, dust and such harder types of particles. It does, to a degree that sets a new standard of listening pleasure.

Another and perhaps more significant result of this cleaning technique should be noted. A record containing some grit particles was played with a stereo cartridge having a .0004-in. tip radius and with an arm of very low frictional characteristics. Without the suction device in action, the tracking force was gradually lowered until the phenomenon of mistracking occurred; it happened just as the (1 gram) minimum rating for mistracking was reached. It ceased immediately the suction device was put in operation, the tracking force then being progressively lowered further still. The "misbehavior" point then become solely dependent on the modulation excursions, in the case concerned the force being reducible to 0.2 g with reproduction still being as it should.

NEED FOR WET CLEANING

As mentioned earlier, there comes a time when most records played in polluted atmospheres have to be given wet-cleaning treatment if brilliance is to be retained and the stylus kept reasonably free from clogging. This procedure bristles with problems which fortunately have now been overcome.

One conclusion soon reached was that any manual form of wet-cleaning was far too haphazard a method for producing consistently good results; they were achievable only by a "machine" method, with manual operations confined to preparatory actions. The reason will become clear after considering the basic requirements involved:

a. After a surface has been wetted and cleaned, sleeving and then storing must not take place unless and until the surface, right down to the groove roots, is bone dry. This is a common trouble with manual wet-cleaning: damp pads are applied, the user not realizing that, particularly in a humid atmosphere, moisture may remain in the root area for up to half an hour or more. However, he does not wait that long, only to find later that he has to pry the record away from the polyethylene lining of the sleeve, then finding on playing it that he gets a whole battery of noises.

b. So, in order to avoid long waits before being able to sleeve cleaned records, there should be means for drying them quickly.

c. The quick-drying process must be such that airborne dust will not sweep the surface and stick to grooving when it is not yet dry.

d. The quick-drying procedure should comply, however, with a certain rule, which is that the drying device must leave the surface just passed with a remanent degree of dampness that evaporates within a few seconds afterwards. This is known as the "damp persistence" time. It must be short to minimize the chances of random airborne particles meeting damp grooving and thus sticking, and it has to be long enough to ensure that the condition still does apply when ambient air is warm and dry. The need for "damp persistence" itself is to safeguard against the generation of electrostatic charges and their consequent attractive effects.

e. The machine should embody means for dislodging firmly-stuck particles.

PROBLEMS OF DRYING

The most difficult to fulfill of these requirements proved to be the proper drying of the *grooves* in a record. The moisture adheres so pertinaciously there that drying with a pad or a suction mop is not entirely effective. The requirement was eventually met by the inclusion in the wet cleaning machine of a spinning camel-hair brush operating after a suction mop.

When such a brush is rotated at a high speed (say 3000 rpm) the bristles fly out laterally and then assume a flat circular contour. Along its axis there is an air suction inwards to the contour; laterally there is a blowing effect of considerable velocity. Incidentally, this is the same effect, only in an exaggerated degree, of the vortex motion already noted for the spinning turntable.

If, therefore, one tracks a spinning brush across the record following a suction mop, so that the bristles just touch the surface, the blowing effect will dry the moisture out of the grooves, together with any loose grit or dust that has escaped the mop. A high polish results, quite comparable to the original off-the-press polish, whatever the previous contamination.

The ideal "wet-cleaning" machine should therefore consist of: *1.* a liquid applicator preferably operated with scouring bristles so arranged as to penetrate the bottom of the groove; *2.* a mop with a suction device to remove the body of the liquid together with the contaminate; *3.* a spinning brush to perform the final drying and polishing.

A PRACTICAL DESIGN

A prototype machine has been evolved which meets these requirements. As a measure of what it can do, the following is typical of conditions which are sometimes met and which, in the past, have usually defied completely satisfactory remedial action. A record can be in such a filthy state that the music is partly drowned by sporadic noises, and be so infested with smog, built up after many exposures to it, that the stylus is put out of action by "blobbing." It is then wet-cleaned on the machine, perhaps involving several runs owing to the severity of the case, and then played again. The usual result is reproduction at least not far short of what it sounded like when the record was new.

The case just taken is exceptional; in all normal cases where records are used and stored with care but which require occasional treatment to remove film, finger marks, stuck grit and so forth, only one cleaning is involved, taking only a few minutes.

The machine, now come to be known as the Record Doctor, has proved invaluable for reconditioning old mono records so that they can be played properly with modern stereo-type equipment. In many cases the musical content of early recordings has been a revelation. When they were new, the relatively massive styli then in vogue were incapable of transducing the full content, and when they became old, encrustations of contaminate barred the modern cartridge from functioning properly.

The mechanical principle used for wet-cleaning a record is its rotation at a relatively high speed, with various devices then engaging the upper surface radially inwards. Once preparatory actions are completed and the machine started, all subsequent actions are automatic. The design takes into account the differing rates of groove speed between inner and outer grooving. Fluid is never applied in "dousing" quantities which means, if nothing else, that the procedure is never a messy operation. Precautions are embodied against labels ever getting wet.

An important feature is that the record is not supported on the usual type of turntable but on a miniature one the same size as the label; it is made stable by means of a clamping weight. It completely solves the "dirty mat" problem, so ensuring that a cleaned record remains clean on both sides for immediate sleeving and storage. This, however, was not the primary reason for adopting it.

TURNTABLE DESIDERATA

At this point some further comments on the subject of turntable mats are called for. They suffer, of course, from the same sort of contamination as records, by gravitational collection in some cases and by collection from polluted records in others. Though few people seem to realize it, dirty mats are a prolific source of trouble with sensitive cartridges, and all the care and attention bestowed on a favorite record can be nullified by placing it on a mat that merely receives a "wipe-over" now and then.

Here again two types of contaminate are involved: discrete particles and tacky film. Take particles first. There is an interesting psychological effect arising from the very low ratio at times of rib width to inter-rib troughing. Dust readily shows up in troughing, usually producing the urge, on that ground alone, for people to get rid of it. So they assiduously go over each section, making sure that nothing is left in the crevices, but often ending up by giving the ribbing only a cursory wipe. It is the ribbing, of course, that needs the meticulous attention, and provided this part of the mat is clean, the state of the trough area hardly matters.

The trouble, of course, is that dust on the rib crests gets pressed into record grooving, thus producing noises at intervals governed by the ribbing pitch. This transfer action is particularly marked when the "record-to-mat" pressure is not even all over, which it seldom is, and the age of the mat often has some bearing on the matter. If too heavy and abrasive a cleaning pressure is repeatedly exerted, rib crests become roughened and more spongy, thus improving the conditions for more pick-up and then more transfer.

Dust sufficient to bring on noise troubles is not readily detectable by the naked eye, the ribs themselves being so narrow, and it should never be assumed they are clean because they look clean, not even when the troughing itself looks in perfect condition. A simple test is to wind a piece of dark velvet around a pencil, breathe on it for a few seconds, and apply the device radially to the rotating mat. With ribbing that appeared clean at the start, the velvet will often emerge with spaced patterns of dust visible from several feet away.

Cleaning action naturally means relative movement between cleaning device and disc. It can be done with a stationary device and a moving disc, or vice versa, or a combination of both. The dual movement principle is the one used but it became practicable only after reducing the "centrifugal draught" effect to negligible proportions—otherwise it would have been a case, with fumes about, of polluting the surface almost as fast as the previous pollution was being removed.

Three factors entered into the solution. First, the rotation time was reduced to two minutes, being only about one-tenth of the time for a 12-in. record played at 33 rpm. Secondly, speeding up the rate of rotation for wet-cleaning means that air moves faster horizontally over the surface, thus making it less prone to downward deflection. This, however, is not necessarily so at the rim, if something there has a contour which causes turbulence.

By dispensing with the normal turntable, its rim-contour problems ceased to exist, leaving only the record edge itself to be considered. This resolved into the design of a rotating unit in which, for all practical purposes, the "free air" conditions below the record matched those above it. This produced a more or less balanced "centrifugal draught" action on both sides, causing the air to fly past the rim with a nearly-flat trajectory.

The points just mentioned lead naturally to the question of turntables. All normal ones have rubber mats, relatively long rotation times, and rim contours that produce turbulence to some degree or other—all conducive to the contamination of records in other than perfectly clean atmospheres. A case thus arises for miniature turntables on the playing deck. Experiments have shown that the advantages are far from hypothetical, so much so that practical points were soon considered. This is where the difficulties lie. To apply the principle would mean drastic changes to playing-deck design, one being the provision of the necessary air space below the record, and the other the location elsewhere of the dynamically-balanced mass represented by the existing type of turntable.

The main functional problem is that an unsupported disc like an LP record can be stimulated into vertical oscillation, and abnormal precautions against this are necessary. For instance, rumble has to be kept very low and an auxiliary like the suction device must never generate vibration that can get to the cartridge laterally through the cross-section of the record.

The simplest arrangement is merely to use a disc fitted to the final spindle, its diameter being that of a record label, with a rubber ring to stop slip and a clamping weight to make the record stable. However, the tendency these days is for records to become thinner and thinner, now veering towards the stage where they must be supported nearly to the rim to stop them from drooping during playing. This has been forestalled by two alternative arrangements. One is the same miniature turntable but with spider-like arms extending radially outwards and carrying a rubberized annulus that supports the record close to the rim. The other is a thin, stiff, non-ferrous plate reaching nearly to the rim and processed in a way that relieves all internal strains. Both would entail careful handling, of course, to avoid being knocked out of true alignment. With good aerodynamic design, both would come close, in characteristics, to the simple turntable first mentioned.

Because of the practical difficulties mentioned, it may be a long time before miniature "balanced draught" playing turntables appear, but when they do, it will help appreciably towards record cleanliness.

IDEAL AND FAULTY PLAYING CONDITIONS

Studies over the past two years included a subject seldom aired in problems of this nature, the tackling of contaminate at the source and the general conditions that apply in rooms where phonograms are operated. The subject is best introduced by quoting the best place for operating the modern specialized phonograph—by an Eskimo in an igloo near the North Pole during the summer when he does not have to light his whale oil lamp. His own central-heating system is body heat so that he does not have to resort to man-made devices which generate fumes. He also has no vehicular traffic passing his front door, nor is he troubled to any appreciable degree with dust and grit.

So far as our own habitats are concerned, contaminates are all around us. Full air-conditioning in homes has gone a long way towards keeping out sources of trouble but dust is still tracked into rooms on shoes and brought in on clothing, and people still smoke, to name only two things to be reckoned with. However, even under these conditions the most sensitive equipment can be made to work satisfactorily with a minimum of record maintenance, but only subject to certain things *not* being done. It is because so many people do these things that the results sometimes leave them in despair.

Various "home" happenings have been investigated, covering all seasons of the year and a wide variety of contaminates, and the findings leave no doubt about the importance of this aspect of record care. Perhaps the best way is to refer to them subjectively with a typical example.

Friends are invited for an evening of serious record-playing, perhaps on a new phonograph recently acquired. To help create the impression intended, everything must, of course, be in perfect order, so great activity commences in getting the music room "straight." The vacuum cleaner is brought out and applied to everything it can deal with, which so far as extraction is concerned is an excellent procedure; like the turntable suction device, particles are collected and stored. However, few people ever think about the exhaust end of the cleaner; carelessness in this direction can result in the air being filled with fine dust blown out of inaccessible places. Cloth dusters are then brought to bear and are usually operated on the basis that the visitors are bound to examine everything in the room from floor to ceiling. The final action, usually, is to "puff up" the cushions by vigorously operating on them like a pair of bellows.

So far so good, but the man of the house then does his bit, by making sure that the playing deck, which the visitors are also bound to examine, is perfectly clean. He brushes or wipes every visible surface, often accompanied by a certain amount of blowing. He may also lay out a few new records, still in their covers, in readiness for the session.

Apart from deliberately setting out to dirty the air and to make things difficult for playing deck and record, nothing more inimical to good noise-free reproduction could be devised. This, in fact, was a typical home procedure specially studied, to see what concentrations of airborne particles resulted, and the rates at which the concentrations declined under various atmospheric conditions conjointly with such household plant in action as central heating and air conditioning. Sometimes room air returned to the status quo only after a matter of hours.

Similar tests were made under winter smog conditions and with heavy diesel traffic during the summer. These tests were not made for seeing if fumes did condense on records: that was already well known, right from the advent of the low-mass lightweight-tracking cartridge. They were to establish the relationship between particular concentrations and the effects on both records and styli.

A matter equally important as room air conditions is phonograph design itself, in respect to shape, disposition of components and other such characteristics; many a searcher after better results has found himself handicapped at the start by the arrangement he unwittingly went for. This applies particularly to the low-slung type of cabinet where the playing deck is close to the floor. It may look modern and fashionable but, apart from that, nothing good can be said for it from the viewpoint of consistently good reproduction, easy operation and easy maintenance.

IMPLICATIONS ON PLAYER DESIGN

Where low-mass, lightweight-tracking cartridges are concerned, optimum results depend on three basic conditions: *1.* Location of the playing deck where it is least prone to contamination, and certainly away from central heating airduct exhausts and like sources of draught. *2.* The protection of records during playing by means that do not themselves accentuate contamination troubles. *3.* The ability to view records and styli *in situ* under a good light that will clearly reveal blemishes.

The ideal scheme is a playing deck at not lower than waist level, not only to get it well away from the floor but to facilitate inspection and to get components into the safest position for handling and maintenance operations. The deck should be sealed off from all sources of heat from within the phonograph, thus safeguarding against convection currents and the dirt they often carry along with them, and the covering arrangements during playing should produce no air turbulence when they are operated. The biggest offender in this last respect is the hinged lid which when raised and lowered acts somewhat like bellows, producing variations in air pressure which not only cause airborne dust and fluff to swirl around but also replenish the supply from external sources each time the lid is lifted.

The examination of styli *in situ* needs explaining, being particularly relevant to those who operate high-class cartridges in certain atmospheres during the winter, when record surfaces are apt to be colder than ambient room air. If there is smog about, it is often necessary to remove collected film on the stylus *at the end of each playing of a 12-inch side,* otherwise the stylus tip may not register properly at the next playing. Film collects in a state of slight compression all bunched around the tip; when the stylus is lifted, the blob "eases out" slightly and so prevents full re-entry into grooving on the next playing.

Another practical point emerges clearly from this analysis: the most important time to clean the stylus is at the end of a playing session. Otherwise the collected condensate will have time to harden and removal will be more difficult.

WHAT OF THE FUTURE?

At the time this paper is being written, both the suction cleaner, which operates while the record is being played, and the Record Doctor which is used before playing to ensure that dirty records are restored to their pristine condition, are being developed and simplified. Our original devices were most complicated and even frightful affairs. They were built up, step by step, to meet specific requirements. The past year has been spent in rationalization, a process that is not yet finished. But since the analysis we have embarked upon seems to be an entirely new and promising development for the record industry, I have thought it worthwhile to report our interim conclusions. Perhaps next year I may be able to present a description of finalized apparatus.

APPENDIX

Let us restate the observation mentioned under *Static and Anti-Static* above. A thin insulator of large area, which has a charge on it, is placed on a conducting surface. Then the electric field on the outer surface of the insulator vanishes identically. This problem is the inverse of the problem of the electrophorus, and we see that this must happen physically as follows.

The negative charge on the insulator attracts an equal positive charge on the conductor. Then the charges appear as shown in Fig. 1. The positive charge on the conductor and the negative charge on the insulator are close together and produce no net field on the outside of the insulator.

We can prove this rigorously as follows. Let us draw an imaginary cylinder perpendicular to the surface with one end inside the conductor and one outside both the conductor and insulator. By a famous law of electricity—Gauss' Law—the flux of the electric field through the surface of this cylinder is equal to the charge enclosed —which is zero. (This is the law: $\vec{\nabla}\cdot\vec{E} = \rho/\epsilon_0 = 0$.)

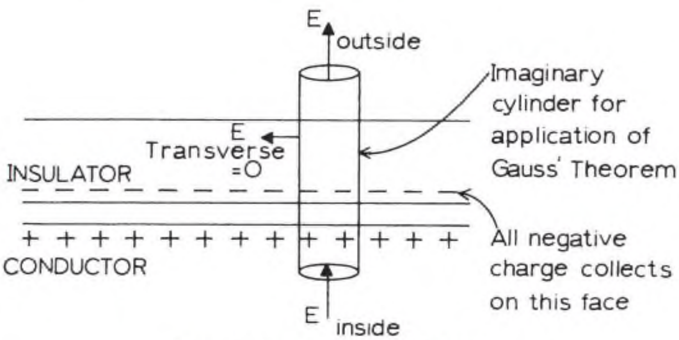

FIG. 1. Electrical charge diagram.

Now by the symmetry of the problem there can be no electric field normal to the sides of the cylinder. It can only be normal to the surface at the ends. We thus get the relation that the electric field outside the insulator equals the electric field inside the conductor with the directions as shown in the figure ($E_{out} = E_{in}$). But inside a conductor the electric field is zero, or else a current would flow. Therefore, outside the insulator the electric field is also zero and in the problem as originally stated the phonograph record will not pick up dust while it is on the conducting surface.

Note that the insulator is not *discharged* by the conductor. The negative charge appears on the surface of the insulator closest to the conductor by polarization of the individual atoms comprising the insulator. But they remain *bound* charges. Thus when the insulator is removed from the conducting surface there will again be an electric field on the surface, and it will again be able to pick up dust. The conductor will have a negative charge on the opposite side from the insulator—but we don't mind it picking up dust on this side.

In case the statement that a charge near the surface of a conductor attracts an equal positive charge is doubted, a proof follows (it is given in most elementary electricity texts).

Consider the electrostatic problem of finding the fields and equipotentials for a positive and negative charge at a distance $2d$ apart. There is a zero potential plane half-way in between. The field on this plane can be calculated.

The total field is the vector sum of the fields due to the positive and negative charges, $E = E_+ + E_-$ (see Fig. 2).

$$|E| = \frac{q}{4\pi\epsilon_0(r^2+d^2)}\cos\theta + \frac{q}{4\pi\epsilon_0(r^2+d^2)}\cos\theta = \frac{2qd}{4\pi\epsilon_0(r^2+d^2)^{3/2}}.$$

Now this, by symmetry, is normal to the midplane. If we place a conductor at the midplane, and remove the bottom charge, the field above the conductor remains the same.

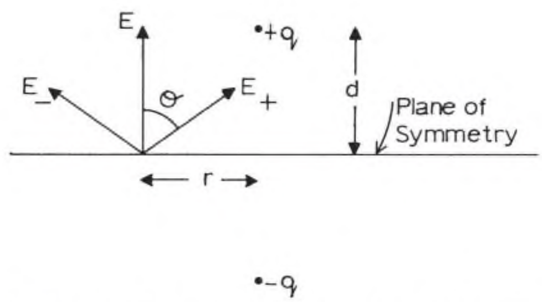

FIG. 2. Vector addition of the fields due to positive and negative charges.

The surface charge density induced on the conductor to give this field is found by another application of Gauss' theorem: $\sigma = -|E|\epsilon_0$.

We can integrate this surface charge over the surface of the conductor to get the total charge induced on the surface.

$$Q = -\epsilon_0 \int_0^\infty \frac{2\pi r dr}{4\pi\epsilon_0} \cdot \frac{2qd}{(r^2+d^2)^{3/2}} = -q.$$

This proves the statement for a single charge. Since, by the principle of superposition, the fields and forces due to each charge are linearly *added*, the surface charges induced in a nearby plate by a distribution of charges always add to give a charge equal and opposite to the sum of the charges.

THE AUTHOR

In 1893 Percy Wilson was born in Halifax, Yorkshire, England. Receiving a B.A. in mathematics with first class honors from Oxford in 1915 and an M.A. in 1918, he served as a Naval Instructor in the Royal Navy from 1915 to 1919. During the latter part of this period, he was Lecturer in Applied Mathematics at the Royal Naval Engineering College.

From 1919 to 1938 Mr. Wilson was an administrative officer in the Ministry of Education after which he became Principal Assistant Secretary in charge of the Roads Department of the Ministry of Transport. As a hobby, he acted as Technical Adviser to the British magazine *The Gramophone* from 1924 to 1938. After his retirement from public service in 1953, Mr. Wilson became Technical Editor of that magazine.

Author of *Modern Gramophones and Electrical Reproducers* (1929) and *The Gramophone Handbook* (1957), Mr. Wilson is a member of the Audio Engineering Society and a Founder Member of the British Sound Recording Association.

LETTERS TO THE EDITOR

Note to Members: This is your column. It is designed for the discussion of papers published in the Journal *and other pertinent topics about which you feel strongly.*

INSTRUMENTS FOR RECORD CLEANING*

PERCY WILSON

"The Gramophone" Magazine, Middlesex, England

THIS paper describes several minor techniques and two instruments developed on the principles published in the *J.A.E.S.*, April, 1965. One instrument is automatic in operation and (wet) cleans soiled records (1 minute per side). The other tracks in front of the stylus and maintains cleanliness.

Record Contamination

The previous paper gave the conclusions reached from a prolonged research into record contamination. Three forms of contamination were identified: relatively hard discrete particles such as dust, grit and fluff; deposits from household sprays (including spittle from blowing), soot and preparations such as anti-static dopes and detergents deliberately applied; and condensates from air-borne fumes of an oily nature such as cooking vapors, tobacco smoke and smog loaded with diesel fumes.

So long as it is not attached to the groove the first type can readily be removed by a cleaning device which precedes the stylus. The second and third, if not of long standing, can also be removed in this way: they are soft at the beginning but gradually become sticky and then harden and trap the discrete particles; the whole mass then becomes more or less firmly attached to the groove. At this stage only a wet cleaning process can remove the contamination completely. Even then, if the record has been repeatedly played while the particles were firmly stuck, they may have been pressed into the record material and their removal may leave slight pits. It is in fact easy to distinguish between the "pops" caused by particles and the fainter noises of the pits left behind.

Dry and Wet Cleaners

Two instruments have been developed for dealing with this situation: the Record Player Cleaner (R.P.C.) and the Record Doctor (R.D.). Both depend on the use of a degree of suction from a separate suction generator (S.G.).

The R.P.C. is normally used without application of liquid, though a small amount of humidity (which can readily be provided) is an advantage. It is a self-tracking device carried along by the groove and it is of importance that the tracking force should be kept low if damage to the groove is to be avoided. It is also important that the disturbed contaminate should be instantly removed and not run the risk of being re-deposited: hence the value of suction operation. A suction of about 4 to 6 liters per minute through a tube of ¼-in. decreasing to ⅛-in. bore has been found to be adequate. It should be noted, however, that the air impedance is high, and therefore the ordinary run of suction generators, such as domestic vacuum cleaners, is ruled out. Special types such as "solder gobblers" are suitable but they are apt to be noisy.

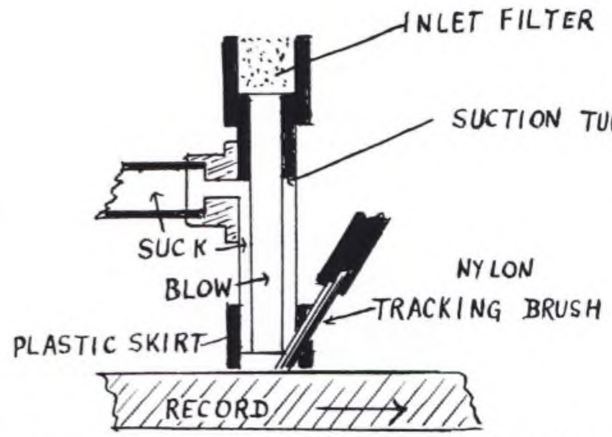

FIG. 1. Blow-suck-scrub nozzle of the record-player cleaner.

The R.D. operates through the application of a suitable liquid, followed by a groove scrubber to loosen the contaminate, then a suction mop to remove the liquid sludge and finally a spinning brush which, by contact and by its blowing action, completely dries the surface of the groove and leaves it in a highly polished condition. For this instrument a higher degree of suction is advisable: from 8 to 10 liters per minute is aimed at.

The Record Player Cleaner

The R.P.C. operates through a forward-facing brush with 2 dozen or so nylon bristles inclined at an angle of about 45°. This loosens the contaminate, which is then removed through suction. To assist in the loosening process the brush is arranged to work at the mouth of a nozzle where both blow and suck actions are operating together so as to create turbulence. This nozzle is shown in Fig. 1. The suction is through the annulus of the larger tube and automatically draws the air down through the inner tube. There is a possibility of a leak effect by air drawn in parallel to the record surface, but this is minimized by having a plastic skirt round the tube which is arranged to be close to the record surface. By having it actually

* To be presented October 14, 1965 at the Seventeenth Annual Fall Convention of the Audio Engineering Society, New York.

in contact in the first instance and then dropping the nozzle (on its arm) gently onto the surface, the change of noise from a sharp tap to a soft thud indicates that the brush has taken charge of the contact in place of the skirt. By this simple technique a clearance of as little as 0.002 in. is easily realizable.

The nozzle is carried by an arm in the form of a plastic tube which houses a cotton-batting filter through which the suction is drawn. It is continued through a perspex section mounted on a single pivot situated in the center of the suction channel. The counterbalance arrangement at the end of the arm operates for both longitudinal and lateral balance. A picture of the device is shown in Fig. 2 and its mounting in relation to a sophisticated playing-deck arrangement in Fig. 3. It will be noticed that an additional filter has been provided at the base of the arm. This was found desirable to prevent any contaminate which had escaped past the edge of the floating filter from being drawn into the suction generator. This, as well as the state of the floating filter, which becomes quite dirty after only a few record sides have been played, is ample indication that the nozzle captures microscopic particles as well as larger ones from the record surface.

A number of important considerations dictated the adoption of this design rather than several alternatives, including the one exhibited at the 1964 Fall Convention of the A.E.S.

1. A forward-facing brush has been found essential for dislodging particles stuck to the groove by fume contaminate even in their softer state. If damage by such a brush

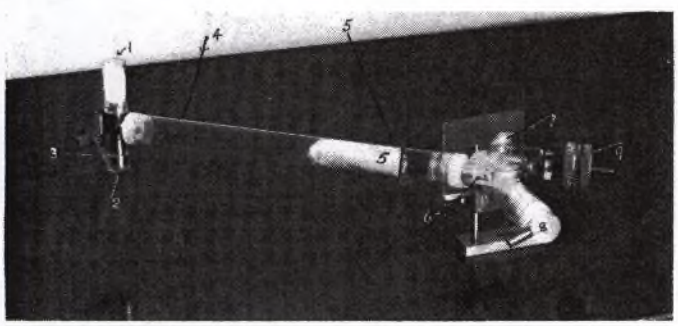

FIG. 2. Photograph of the record-player cleaner. (*1*) Input air filter (also acts as tracking weight). (*2*) Blow/suck nozzle. (*3*) Tracking (forward-facing) brush. (*4*) Transparent tube as carrying arm. (*5*) Suction filter. (*6*) Perspex suction channels. (*7*) Universal pivot member operating in suction channel. (*8*) Suction-out through final filter. (*9*) Counterbalance: longitudinal and lateral.

to the finer groove modulations is to be avoided the lightest possible tracking force is to be aimed at. In the present design this is represented by the mass of the inlet filter at the top of the nozzle, the arm being balanced on the pivot, in the first instance, with the filter removed. Actually, a total tracking force of 0.25 gram has been found feasible, and since this is applied through some 2 dozen bristles the force between each bristle and the groove is of the order of 0.01 g. This is an ample safeguard against wall damage even if the groove is dry. It is also possible to arrange for a modicum of lubrication by putting a drop of distilled water onto the loosely compacted cotton batting composing

FIG. 3. Photograph of record-player cleaner in operation. (*1*) *through* (*9*), as in Fig. 2. (*10*) Stylus microscope (swivelling to preset position). (*11*) Arm stop positioned for microscope. (*12*) Lamp to illuminate stylus.

the input filter; the air drawn down the nozzle will then be slightly moist.

2. To have so low a tracking force the arm must be of low mass and be perfectly balanced. Bearing friction and tracking error must also be at a minimum. These considerations dictated that the suction tube should itself form the carrying arm. The tube is made of transparent acetate, or thin-walled perspex, so that the degree of collection of contamination can be regularly observed. In some dirty areas it is seen to be desirable to change the filter after only 10 record sides have been played.

The use of transparent tubing has another advantage because of its lightness: the arm inertia becomes so low that quite substantial record warps, even when combined with "swingers" (eccentric discs), have no effect on the operation.

Figure 3 also shows another desirable feature: the pivoting, balancing and counterweight system have been so designed that the mounting can be made quite comfortably in the triangular space left between a 12-in. turntable and a $13\frac{1}{2}$-in.-square motor board. This is done by giving the arm a linear offset at the pivot end, so reducing the amount of offset required at the nozzle end to give proper tracking. With this arrangement the arm, when not tracking, sits back along the parallel to the player cabinet. Incidentally, the same sort of system might be used with advantage in pickup arm design.

To facilitate the cleaning which is periodically required, the R.P.C. can be quickly taken to pieces into separate parts, though it should never be necessary to dismantle the nozzle.

The prototype has now been in regular use since December, 1964, the filters having been renewed many times. One can in fact become rather scared at the amount of dirt picked up, even in the comparatively clean air of the residential part of Oxford, where the author lives. Another prototype has been in use by an associate in a residential area on the outskirts of London where smog, diesel fumes

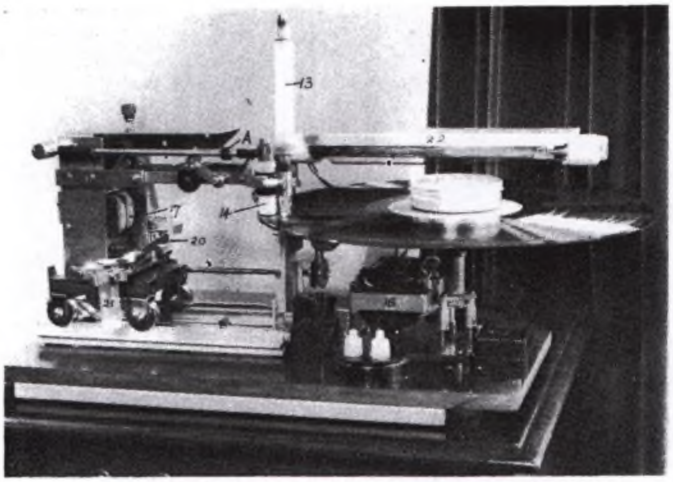

FIG. 4. The Record Doctor. (*13*) Liquid applicator. (*14*) Suction mop: suction through tubes (*A*) and (*B*). (*15*) Forward-facing groove scrubber. (*16*) Spinning brush operated by (*17*) Motor mounted in traverse carriage. (*18*) Rim-drive turntable (miniature) motor. (*19*) Belt drive of screw traverse mechanism, operated from turntable spindle. (*20*) Twin trigger mechanism for lifting half-nut clear of traverse screw at end of run. (*21*) Container for applicator at end of run, linked to trigger mechanism to permit suction mop and spinning brush to proceed up to record label. (*22*) Strip lamp, to provide sufficient heat to control humidity and rate of drying, adjustable as to height on (*23*) Post to carry strip light.

from a nearby motorway, and even factory fumes from an industrial area within a few miles give rise to substantial air pollution. In his case, the arm filter becomes black within a few hours' playing time.

In both cases, the use of the R.P.C. has resulted in a remarkable improvement in high-note and transient reproduction.

Figure 3 also shows another feature of interest besides the special adaptation of the pickup arm (which is another story). This is the permanent fixing of a long-focus stylus microscope in such a way that when the arm is set back against a pre-set stop, the microscope can be swivelled into position for viewing the stylus without any fumbling adjustments. This arrangement has made it simple and quick to determine the condition of the record (and the efficiency of the cleaning techniques) by the amount of muck picked up by the stylus after playing any chosen annulus of the record. It is even possible to determine by the contamination left on each side of the stylus whether greater pressure has been exerted on one wall of the groove than on the other. (Hence the delicate anti-skating devices at the back of the arm, and the cleaning brushes kept on the playing deck in convenient positions.) All this, of course, is highly sophisticated and non-commercial. But it has provided a wealth of valuable information in this research.

FIG. 5. The Record Doctor. (*13*) *through* (*23*), as in Fig. 4.

The "Record Doctor"

If the household spray, finger mark, fume condensate films have been left undisturbed for some time and been allowed to harden, dry cleaning with the R.P.C. becomes no longer effective. The more drastic measures of wet cleaning are needed to remove the contamination. Because of the lubrication afforded by the applied liquid, considerably greater pressures between tracking devices and record become permissible without risk of damage. Scrubbing action can therefore be greatly improved and this, combined with the loosening effect of the liquid itself, makes possible the removal of even hard stuck material.

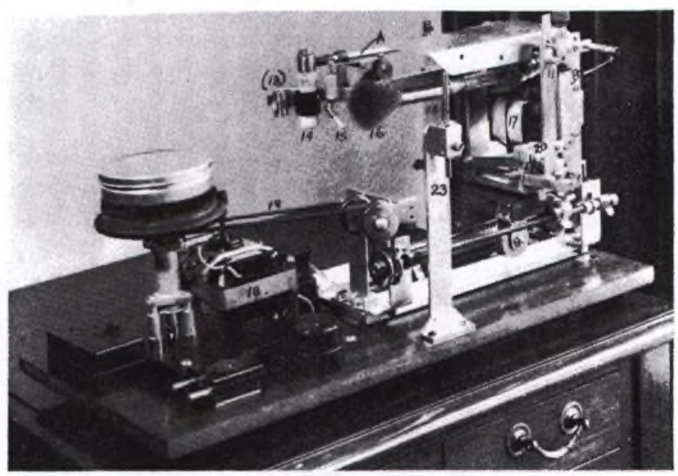

FIG. 6. The Record Doctor. (*13*) *through* (*23*), as in Fig. 4

The main problem has been to maintain the cleaning efficiency while expediting the process and leaving the record grooves quite dry.

As explained in the previous paper, a miniature turntable is used, rotating at a high speed—now 150 rpm—so as to minimize the vortex action of the air on the record and the onset of new contamination during the process of removing the old.

The cleaning operation is done by 4 devices mounted on a carriage which is drawn radially towards the record by a half-nut on a screwed rod. The pitch and rotation speed of the rod has now been arranged so that the 4-in. traverse takes place in one minute.

Figures 4, 5 and 6 show the details. Of the four devices involved, the last to come into operation is the spinning brush (*16*) which does the final drying and polishing. This must move radially across the record; its motor drive (*17*) is mounted in, and moves with, the traversing carriage. Preceding that comes the suction mop (*14*) with suction provided through tubes *A* and *B*. This comes into operation just after the forward-facing groove scrubber (*15*) whose bristles are inclined at an angle of 30° to 45° to the record surface. First in operation is the liquid applicator (*13*) which consists of a nozzle containing a brush with nylon bristles, backed by a plastic foam sponge through which liquid percolates from a tubular container, the rate of flow being controllable by a simple valve at the top.

Elements (*13*),(*14*) and (*15*) are each mounted in tubular items attached to a carrying arm which is fixed to the face of the traverse carriage, about 1½ in. from the radial line followed by the spinning brush. In this way they are given a freedom of vertical motion, thus maintaining independent contact with the record whether the latter runs true or not.

The more important thing to notice about them, however, is that each has a path across the record along a different chord. Their traverse lengths and their contact with successive grooves are therefore quite different. Quite a tricky geometrical problem has therefore been presented to ensure that *1.* At the beginning of the transit the applicator (*13*) starts at the run-in groove, and at the end it is clear of the record label. *2.* Elements (*15*) and (*14*) are lowered on to the record in such a way that (*14*) is just on the rim when (*15*) is in the first groove a little after (*13*) has passed it. *3.* Item (*15*) never overtakes (*13*) but keeps ahead of (*14*) throughout the transit. *4.* Neither (*14*) nor (*15*) touch the label but do reach the run-out groove. *5.* The carriage stops before the spinning brush touches the label, but goes far enough to allow the brush to clear any remanent liquid from the blank space outside the label.

In the model illustrated, adjustable, independent mountings were provided for (*13*), (*14*) and (*15*), so that these adjustments could be made. The appropriate positions having been determined, a later model has all the positional holes cut in a perspex block attached to the carrying arm. In both cases (*14*) and (*15*) can be raised and lowered onto the record together.

The operation was much facilitated by the provision of the trigger mechanism shown at (*20*). This was a double stop. The first triggers the catch, permitting the spiral spring to lift the half-nut clear of the screwed rod and therefore stopping the traverse. Applicator (*13*) is then removed from its transit position and placed through a ring into the little bottle (*21*) mounted on the carriage. Its weight on the ring releases the first catch and permits the carriage to proceed further until the mop and spinning brush have finished their transit. Then the second trigger catch comes into action and stops the traverse again. In this way it is ensured that no liquid deposited by the applicator (*13*) is left on the record beyond the recorded surface.

The rim drive of the miniature turntable should be noticed. The motor, which runs at 1450 rpm, is mounted on two pillars and rests by its own weight, with its rubber-encased spindle resting against a rubber ring on the turntable rim. The dimensions are such as to produce the turntable speed of 150 rpm so that the belt drive to give the required rotational speed to the screwed rod is a simple affair.

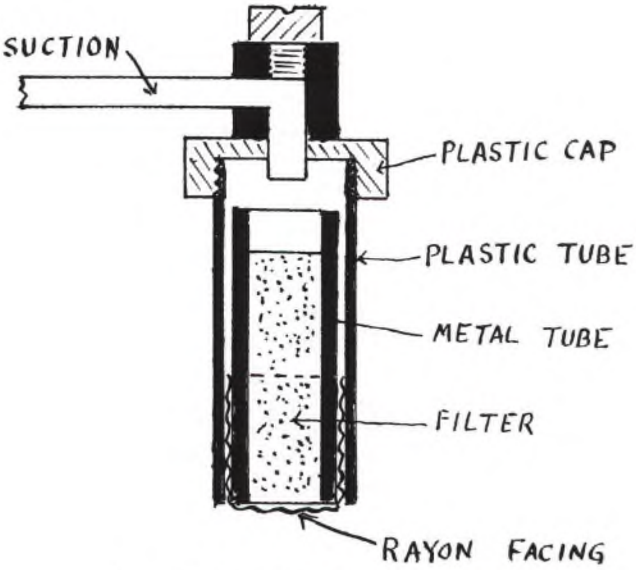

FIG. 7. Detail sketch of the suction mop.

Particular attention has been paid to the design of the suction mop. In the early experimental models this was just a hand-rolled cylinder of cotton batting faced with a piece of linen and pushed into the suction tube so that the linen face would rest on the record. This worked surprisingly well, but the finished design is much more efficient in that it maintains a flat instead of a bulging surface on the record, and there is thus less suction leakage.

The details are as shown in Fig. 7. There are two tubes, the outer being plastic and the inner of aluminum with a machined end-face. Inside the inner tube is a standard cigarette filter. Over the end of the tube a piece of rayon is placed, and with that it is pushed into the outer tube—a tight fit—until the rayon face is just clear of the end. The outer tube is then screwed into the holder which is attached to the suction tube (*A*). Rayon is better for the purpose than linen, being of a more open weave and with no fluffy threads: the suction is therefore more effective in pulling the liquid sludge through into the filter which often goes black with the cleaning of one side of a record, the rayon facing remaining fairly clean. The suction generator is connected through a vacuum flask by flexible tube to the inlet at (*B*).

An adjustable strip lamp (*22*) has been provided so as to improve the drying effect during days of high humidity. This was an afterthought, but it was found to be worth retention not only for that primary purpose but also because it was found, by adjusting the height so as to give a greater heating effect, that during a few minutes of record rotation it would warm a warped or dished record sufficiently to permit it to be nicely flattened by insertion between two pieces of plate glass.

The liquid usually used is water, preferably distilled water or filtered rain water. It is more effective at a temperature of 120°F to 150°F than at a lower temperature; at a higher, there is a risk of causing some deformation. Three cc in the applicator (*13*) is enough. If the record is badly soiled, and especially if it is greasy, even if only with finger marks, a preliminary treatment with a mixture of 25% ethyl alcohol and 75% water can be carried out with advantage. (If ethyl alcohol is not available methylated spirit can be used.) It can be applied by means of a soft (baby's) hairbrush. But remember the caution in the previous paper: *do not wet a record surface unless an efficient method of instant drying is available.*

A special suction nozzle has also been devised to be used in place of the mop for obliterating light scratches. This consists of a glass tube with a fused flat end, surrounding a suction tube into which nylon bristles are inserted at the record end, the glass tube being capable of up-and-down motion independently of the suction tube. The combination of brush and suction with the burnishing action of the glass tube will rub out many a light surface scratch —"paper scratches" one used to call them. Deep scratches nothing can remove, at least so far as is known at present.

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Most of the development work described in this paper was carried out by one of the author's associates who desires to remain anonymous. But the author thinks it only proper that his indebtedness should be acknowledged.

Since the photographs were taken this associate has elaborated the R.D. even further so as to eliminate all the manual control except the preliminary charging and setting. As many as 20 sides can now be cleaned in succession without changing the mop filter or recharging the liquid applicator.

Further experiments are proceeding to ascertain whether the machine could be made more effective, or simpler in operation, by reversing the motion, i.e., by starting the cleaning at the inner grooves and letting the cleaning devices drop off in succession at the rim of the record.

Deformation Distortion in Disc Records*

TAKEO SHIGA

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The distortion produced by groove-wall deformation in accordance with the elastic theory of Hertz, combined with the distortion produced by tracing error, is calculated using an extension of the power-series approach of Lewis and Hunt in their treatment of tracing error. The dynamical differential equations for the several components of deformation error representing resonant systems are derived and solved for the distortion components. Numerical results for certain parameters are presented graphically, and the relative importance of various terms is appraised. It is seen that the second harmonic term, for example, arising from tracing error, is reduced by the action of deformation error, away from resonance. For an isotropic stylus impedance there is no cross-coupling between stereo channels in the absence of friction, but a beginning at an analysis of friction suggests a mode of coupling.

INTRODUCTION

WHEN a stylus of a phonograph pickup traces the groove of a disc recording, the groove wall is subjected to a variety of forces. If the disc material is assumed to be ideally rigid, groove-wall deformation produced by such forces may be neglected, and tracing distortion may be calculated from geometric considerations alone. A full analysis, based on this approach, was first given by Lewis and Hunt.[1] Actual disc records are not, however, ideally rigid, and the stylus forces do produce deformations resulting in further distortions of the reproduced waveform. Several analyses, taking the elasticity of the groove wall into account, have been made by Kornei and others,[2,3,5] and, in particular, an anlysis of Miller,[5] was significant for its discussion of resonance involving the elasticity of the groove wall and the equivalent mass of the pickup stylus.

The present work is aimed at extending these analyses, to provide a close and extensive examination of the relations between these stylus forces and their effect upon the waveforms reproduced from the elastic groove wall. The approach of Lewis and Hunt is followed, but both the pressure and mechanical impedance presented by the stylus are taken into account as supplying the forces imposed on the groove wall.

MOTION OF A SPHERICAL STYLUS

In the 45-45 system of making stereo disc recordings, when a signal is recorded in one channel alone, the groove wall on one side is cut in varying depth according to the amplitude of the signal, while the other wall remains smooth.

Let us simulate the stylus tip by means of a sphere, having the same radius as the tip, rolling in the groove. Then, the locus of the center of the sphere will be a plane curve, parallel to the unmodulated groove wall, and lying at a distance $r-\delta_o$ from it, for r being the radius of the sphere, and δ_o being the penetration into the unmodulated wall due to stylus pressure. See Fig. 1. (In this analysis, the deformations of the two groove walls, because of applied forces, will be assumed to be uncorrelated.)

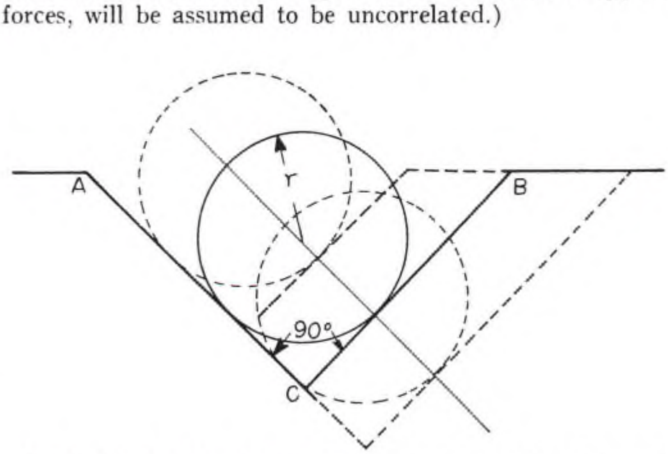

FIG. 1. Motion of a sphere in the groove of a 45-45 stereo record for a signal recorded in only one channel.

On the other hand, as viewed from the modulated wall, the locus of the center of the sphere will appear to differ from the modulation waveform shown in Fig. 2. The locus of the center of the sphere can be given by a relation of the general form

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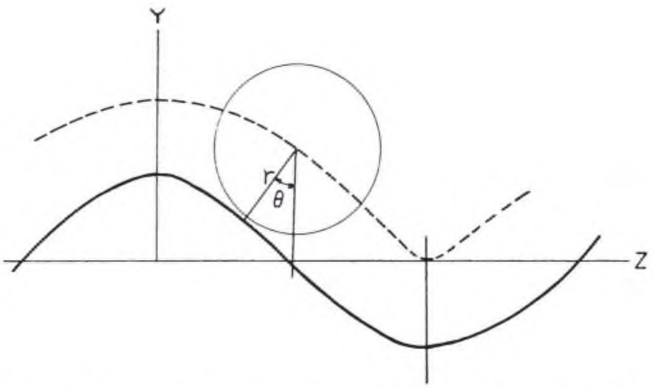

FIG. 2. Locus of the center of a sphere in tracing a sinusoidally modulated groove wall.

$$S(t) = S(q,q',q'',\dots q^{(n)},\dots,\delta), \tag{1}$$

where q describes the modulation waveform on the groove wall, and $q',q'',\dots,q^{(n)},\dots$ are the derivatives

$$q' = dq/d(Vt), \quad q'' = d^2q/d(Vt)^2,$$

etc., δ is the penetration due to the elasticity of the groove wall, and V is the groove speed, so that Vt would be a distance Z along the groove.

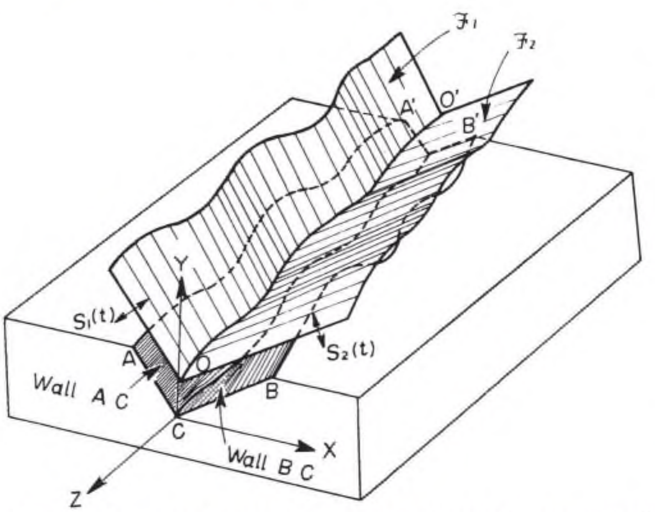

FIG. 3. Locus of the center of a sphere for both groove walls being modulated.

When signals are recorded in both channels, as shown in Fig. 3, the locus of the center of the sphere is described as the intersection of surfaces $\mathfrak{J}_1$ and $\mathfrak{J}_2$ containing the individual trajectories $S_1(t)$ and $S_2(t)$, corresponding to groove walls AC and BC, given in Eq. (1). The surface $\mathfrak{J}_1$ contains the locus S_1 when the sphere rolls on groove wall AC, and the surface $\mathfrak{J}_2$ contains the locus as the sphere rolls on BC. The space curve 00′ denotes the intersection of $\mathfrak{J}_1$ with $\mathfrak{J}_2$. In this way, it is seen that the mutually perpendicular groove walls act independently of each other.

The motion $S_1(t)$ of the sphere, in response to undulations in wall AC, lies in a plane at right angles to every line in the groove parallel to AC so that $\mathfrak{J}_1$ is given by

$$\mathfrak{J}_1 = y + x - [S_1(t) + r]\sqrt{2} = 0, \tag{2}$$

and in a similar manner, the motion in response to BC is given by

$$\mathfrak{J}_2 = y - x - [S_2(t) + r]\sqrt{2} = 0. \tag{3}$$

From these, one obtains the intersection as having the coordinates†

$$\begin{aligned} x &= [S_1(t) - S_2(t)]/\sqrt{2}, \\ y &= r\sqrt{2} + [S_1(t) + S_2(t)]/\sqrt{2}. \end{aligned} \tag{4}$$

This set of equations shows how the motion of the sphere can be presented in a rectangular system of coordinates in terms of the loci of centers individually obtained for spheres rolling on the two groove walls.

ELASTIC DEFORMATION OF THE GROOVE WALL

When a pickup stylus traces a groove wall, the groove wall is subjected to various forces. These forces include not only the static tracking force, but also inertial forces deriving from the equivalent masses of the pickup stylus and arm, together with the elastic restoring force for the stylus, as well as a force (skating) which develops as a result of an angular difference between the groove direction and the direction to the arm pivot. All of these forces develop in a vertical plane normal to the direction of tracing, or groove direction. The static tracking force is normal to the disc; inertial forces are in the same direction as the stylus motion; the elastic stylus restoring force is in a direction determined by the ratio of vertical and lateral compliances of the stylus, and the skating force also develops laterally in a vertical plane.

The tracking force F_G can be decomposed into two forces, each normal to a groove wall. The force in the direction of the Y-axis in Fig. 4 is, therefore, $F_G/\sqrt{2}$. When the vertical and lateral stylus compliances are equal, the elastic restoring force is also aligned with the Y-axis. Otherwise, the independency of S_1 and S_2 would be perturbed. The inertial forces for both stylus and arm lie in the same direction as the stylus motion, i.e., the Y-axis of Fig. 4. If the arm were of finite length, the independency of S_1 and S_2 would again be perturbed. An infinite length is, therefore,

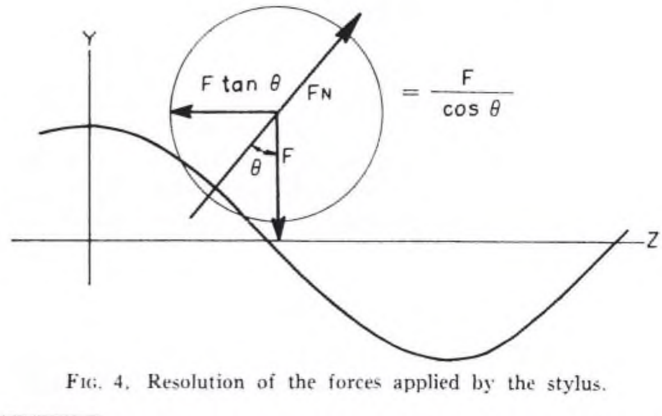

FIG. 4. Resolution of the forces applied by the stylus.

† The positive sense is referred to the outward wall normal for both S_1 and S_2. In another convention, the inward normal is the reference for one of the two walls, so that x would involve the sum of S_1 and S_2, and y would involve the difference.

assumed here. The above three forces are taken into account in the derivation which follows, and they are all assumed to act in the Y direction.

In rolling on the groove wall, as shown in Fig. 4, with a force F applied in the Y direction, the locus of the sphere's center may be perturbed by the elastic deformation of the groove wall. In the playback equipment, the pickup stylus is held fixed at a point on the Z axis by the arm, so that it may not move in that direction while the groove motion is towards $-Z$ with speed V. The Z component of the force resulting from this condition is borne entirely by the arm. This requires a force $F(\tan\theta)$ be applied to the sphere by the arm to prevent it from rolling further down the slope, shown as negative in Fig. 4. Meanwhile, the groove wall is deformed by the force F_N acting along the local wall normal, and the sphere penetrates the wall. According to Hertz, when a rigid-body sphere, of radius r, is pressed on a cylindrically curved surface, of local radius of curvature ρ, with a force F_N, the total deformation δ_N is

$$\delta_N = [(1+r/2\rho)F_N{}^2/(rH^2)]^{1/3}, \tag{5}$$

in which H, which may be called the Hertzian modulus, is given by

$$H = 4E/[3(1-\nu^2)],$$

in which E is Young's modulus, and ν is the Poisson ratio. Since Eq. (5) is not applicable for wavelengths shorter than the radius of the sphere, the applications to be derived must be restricted to regions of validity for (5). The magnitude of F_N will vary with θ, as must δ_N.

In beginning an extension of the rigid-body approach of Lewis and Hunt, so that the output locus deriving from elastic walls may be given, reference is made to Fig. 5.

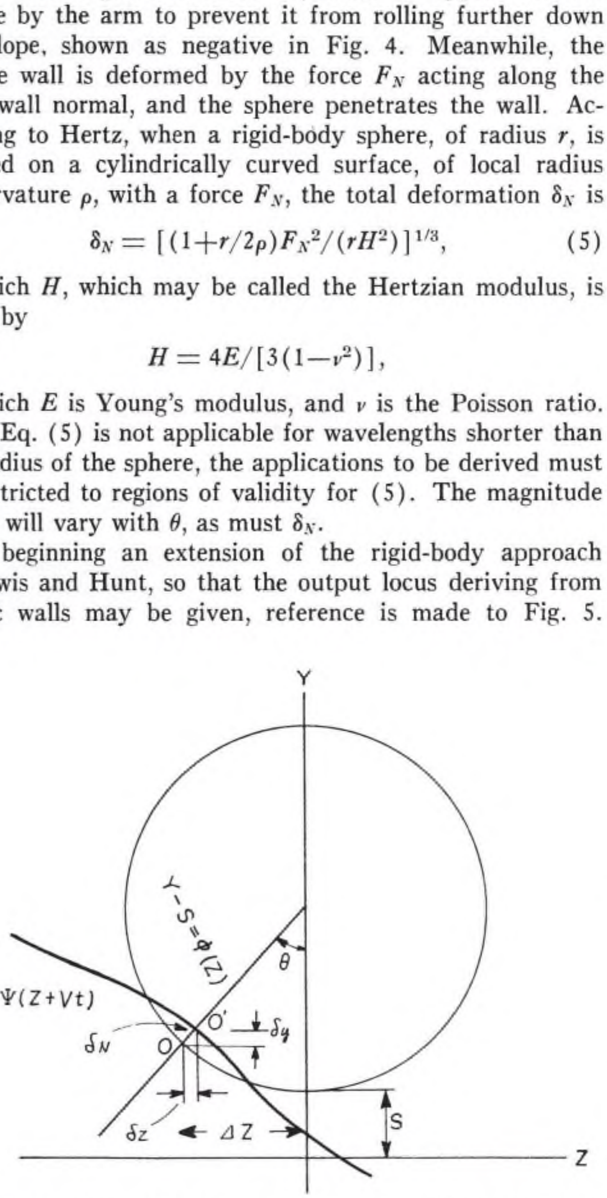

FIG. 5. Displacement of the spherical center resulting from elastic deformation of the groove wall.

There, the modulation of the groove wall is expressed by the waveform $a\psi(Z+Vt)$ which is moving toward $-Z$ at a speed V, and the shape of the profile of the sphere is expressed as the function $\phi(Z)$, which is swinging in the Y direction in response to the modulation. Then, with a contact force F_N being applied, the sphere penetrates along the local wall normal to a depth δ_N.

When the sphere is displaced a distance S from the origin of Y, the vertical position of a point 0 at a position ΔZ from the center is just $\phi(\Delta Z)+S$ so that the equation of the circular profile is

$$Y-S = \phi(\Delta Z), \tag{6}$$

and, for the sphere, one also has

$$\begin{aligned}\phi(0) = \phi'(0) = \phi'''(0) = 0,\\ \phi''(0) = 1/r, \phi^{IV}(0) = 3/r^3,\end{aligned} \tag{7}$$

etc. The Y coordinate of the point 0′ is

$$a\psi(\Delta Z-\delta_z+Vt),$$

so that for 0 one has

$$Y = S+\phi(\Delta Z) = a\psi(\Delta Z-\delta_z+Vt)-\delta_y. \tag{8}$$

Since the tangents to the curves $\phi(Z)$ and $a\psi(Z+Vt)$ at 0 and 0′ are parallel, one also has the relation

$$\phi'(\Delta Z) = a\psi'(\Delta Z-\delta_z+Vt). \tag{9}$$

The force F_N is

$$F_N = F/\cos\theta = F[1+(\phi'(\Delta Z))^2]^{1/2},$$

so that from Eq. (5) one obtains

$$\delta_N = \delta[1+(\phi'(\Delta Z))^2]^{1/3}, \tag{10}$$

in which δ is now given by

$$\delta = [(1+r/2\rho)F^2/(rH^2)]^{1/3}, \tag{11}$$

the same form as (5), but with F replacing F_N. The expressions for δ_z and δ_y then are

$$\begin{aligned}\delta_z &= \phi'(\Delta Z)[1+(\phi'(\Delta Z))^2]^{-1/2}\delta_N\\ &= \delta[\phi'(\Delta Z)-(1/6)(\phi'(\Delta Z))^3+\dots],\end{aligned} \tag{12}$$

$$\begin{aligned}\delta_y &= [1+(\phi'(\Delta Z))^2]^{-1/2}\delta_N\\ &= \delta[1-(1/6)(\phi'(\Delta Z))^2+\dots].\end{aligned} \tag{13}$$

These are to be substituted into Eqs. (8) and (9) to obtain the pair of equations

$$\begin{aligned}S+\phi(\Delta Z) = {}&\\ &a\psi(\Delta Z-\delta\{\phi'(\Delta Z)-(1/6)(\phi'(\Delta Z))^3+\dots\}+Vt)\\ &-\delta[1-(1/6)(\phi'(\Delta Z))^2+\dots],\end{aligned} \tag{14}$$

and

$$\begin{aligned}\phi'(\Delta Z) = {}&\\ &a\psi'(\Delta Z-\delta\{\phi'(\Delta Z)-(1/6)(\phi'(\Delta Z))^3+\dots\}+Vt).\end{aligned} \tag{15}$$

With the help of Eq. (7) an expansion around $Z=0$ is to be made of Eq. (15) to obtain

$$\begin{aligned}&\Delta Z\phi''(0)+(1/6)(\Delta Z)^3\phi^{IV}(0)+\dots\\ &= a\psi'(Vt)+[\Delta Z-\delta\{\Delta Z\phi''(0)+(1/6)(\Delta Z)^3\phi^{IV}(0)+\dots\\ &\quad-(1/6)(\Delta Z\phi''(0)+(1/6)(\Delta Z)^3\phi^{IV}(0)+\dots)^3\}]\\ &\quad\times a\psi''(Vt)+(1/2)[\Delta Z-\delta\{\Delta Z\phi''(0)\\ &\quad+(1/6)(\Delta Z)^3\phi^{IV}(0)+\dots-(1/6)(\Delta Z\phi''(0)\\ &\quad+(1/6)(\Delta Z)^3\phi^{IV}(0)+\dots)^3\}]^2a\psi'''(Vt)+\dots.\end{aligned} \tag{16}$$

The assumption that $\Delta Z=0$, and $\delta=\delta_o$, for $a=0$, and that ΔZ is near zero, with δ near δ_o, for small a, is made to allowing the writing of

$$\begin{aligned}\Delta Z &= a\Delta_1+a^2\Delta_2+a^3\Delta_3+\ldots \\ \delta &= \delta_o+a\delta_1+a^2\delta_2+\ldots .\end{aligned} \tag{17}$$

Then, (16) and (17) may be combined to write

$$\begin{aligned}0 = {} & a[\Delta_1\phi''(0)-\psi'(Vt)] \\ & +a^2[\Delta_2\phi''(0)-\Delta_1\psi''(Vt)+\delta_o\Delta_1\phi''(0)\psi''(Vt)] \\ & +a^3[\Delta_3\phi''(0)+\ldots]+\ldots .\end{aligned} \tag{18}$$

In this equation, each coefficient of a given power of a should vanish separately. This observation results in

$$\begin{aligned}\Delta_1 &= \psi'(Vt)/\phi''(0)=r\psi' \\ \Delta_2 &= \psi'(Vt)\psi''(Vt)[1-\delta_o\phi''(0)]/[\phi''(0)]^2 \\ &= r^2\psi'\psi''[1-\delta_o/r.\end{aligned} \tag{19}$$

The locus S of the center of the sphere is similarly obtained by expanding (14) in a power series around $Z=0$. The expansion for ΔZ is then inserted, using (19). Rewriting $a\psi(Vt)$ as $q(Vt)$, one then obtains

$$\begin{aligned}S(t) = {} & -\delta+q(Vt)+[(r/2)-(5/6)\delta](q')^2 \\ & +[(r^2/2)-(5r/3)\delta+(1/2)\delta^2+(2/3)\delta_o\delta] \\ & \times(q')^2q''+\ldots ,\end{aligned} \tag{20}$$

which becomes, for $\delta=\delta_o=0$, the rigid-body result

$$S(t)=q(Vt)+(r/2)(q')^2+(r^2/2)(q'^2)q''+\ldots , \tag{20'}$$

as obtained by Lewis and Hunt.

EFFECT OF LOCAL CURVATURE IN THE GROOVE WALL

In Eq. (11) it is seen that the penetration δ depends upon the radius of local curvature in the groove wall. The positive sign for ρ is taken when the wall is convex (center of curvature within the material), and negative when concave, as required in the Hertzian formula. This is

$$\rho=-[1+(dY/dZ)^2]^{3/2}/d^2Y/dZ^2. \tag{21}$$

In this, on puts $Z=Vt$ and $a\psi(Vt)=a\psi(Z)$, so that, regarding $a^2\psi'^2(Z)$ to be small, one writes

$$\begin{aligned}(1/\rho) &= -a\psi''(1+a^2\psi'^2)^{-3/2} \\ &= -a\psi''+(3/2)a^3\psi'^2\psi''-\ldots .\end{aligned} \tag{22}$$

Combining this with (11), one has

$$\begin{aligned}\delta = {} & [F^2/rH^2]^{1/3}[1-(r/6)a\psi''-(r/36)a\psi''^2 \\ & +(r/4)a^3\psi'^2\psi''-(5r^3/648)a^2\psi''^3+\ldots].\end{aligned} \tag{23}$$

In this, $a\psi''$ should be understood to be $a\psi''(\Delta Z-\delta_z+Vt)$. Thus, in (23) one should write

$$\begin{aligned}a\psi''(\Delta Z-\delta_z+Vt) = {} & \\ & a\psi''+a^2(r-\delta)\psi'\psi'' \\ & +a^3[\{r^2-r(\delta_o-\delta)+\delta_o\delta\}\psi'\psi''\psi''' \\ & +(1/2)(r-\delta)^2\psi'^2\psi^{IV}]+\ldots .\end{aligned} \tag{24}$$

The effect upon $S(t)$, however, in using (24) instead of simply $a\psi''$ will appear only in the a^4 and higher power terms of the expansion of (8) into a power series. This is because (24) already contains δ multiplied by a^2 and higher powers, and the principal appearance of δ in the expansion will be in a^2 and higher power terms.

EFFECTS OF ELASTICITY AND INERTIA

Consider a model consisting of a sphere having mass M constrained to move along only the Y axis while the wall moves to the left, as in Fig. 6, at speed V. It is assumed

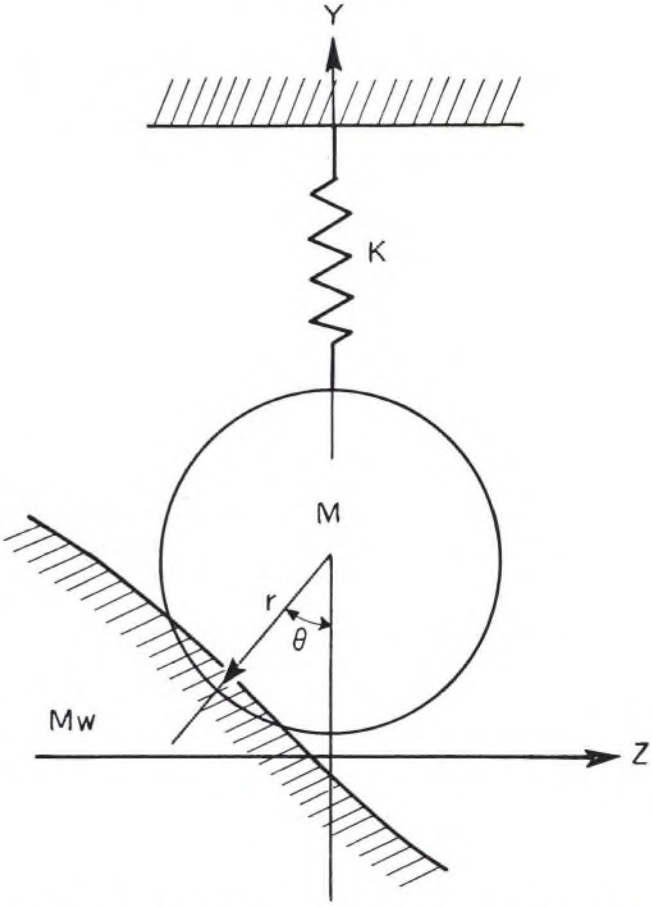

FIG. 6. Model of a spherical stylus and the groove wall as constituting an oscillatory system.

that the wall is elastic, but that its surface is so smooth that the friction opposing the movement is negligible, that a deformation of the type shown in Fig. 7 need not be

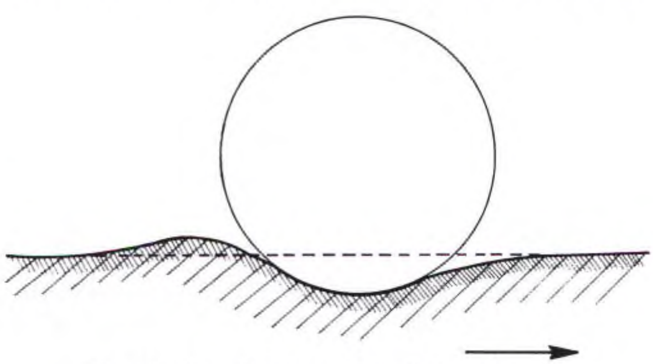

FIG. 7. A possible type of elastic deformation.

considered, and that there is no energy loss within the wall.

Assigning a mass M_w to the wall and writing $S_f(t)$ for an oscillatory term in $S(t)$, one finds the kinetic energy to be

$$T=(1/2)MS_f'^2(t)+(1/2)M_wV^2, \tag{25}$$

and the potential energy to be

$$U=(F_G/\sqrt{2})S(t)+(1/2)KS_f^2(t) \tag{26}$$
$$+\frac{2}{5}\left(\frac{r}{C}\right)^{1/2}\left(\frac{\delta_y^{5/2}}{(\cos\theta)^{1/2}}+\frac{\delta_z^{5/2}}{(\sin\theta)^{1/2}}\right).$$

In (26) the elastic constant of the spring suspending the sphere is K, the tracking force is F_G, and the combination $(1-r/2\rho)/H^2$ is represented by C. Letting Q denote a force in the Z direction applied to the wall, the set of differential equations below

$$\frac{d}{dt}\frac{\partial T}{\partial S'(t)}+\frac{\partial U}{\partial S(t)}=0, \tag{27}$$
$$\frac{d}{dt}\frac{\partial T}{\partial V}+\frac{\partial U}{\partial Z}=Q,$$

is obtained.

Upon insertion of (25) and (26) into (27),

$$\frac{F_G}{\sqrt{2}}+MS_f''(t)+KS_f(t)+ \tag{28}$$
$$\frac{(r/C)^{1/2}\delta_y^{3/2}}{(\cos\theta)^{1/2}}\frac{\partial\delta_y}{\partial S(t)}=0,$$
$$M_w\frac{dV}{dt}+\frac{(r/C)^{1/2}\delta_z^{3/2}}{(\sin\theta)^{1/2}}\frac{\partial\delta_z}{\partial Z}=Q.$$

In (28), $\partial\delta_y/\partial S(t)=-1$, and $\partial\delta_z/\partial Z=1$, and $dV/dt=0$, because the velocity of the wall is constant. With the help of these observations, there is obtained

$$\delta_y=(C/r)^{1/3}(\cos\theta)^{1/3} \tag{29}$$
$$\times[MS_f''(t)+KS_f(t)+F_G/\sqrt{2}]^{2/3},$$
$$\delta_z=(C/r)^{1/3}(\sin\theta)^{1/3}Q,$$

or putting $\delta_z/\delta_y=\tan\theta$, just

$$Q=(MS_f''(t)+KS_f(t)+F_G/\sqrt{2})^{2/3}(\tan\theta)^{2/3}, \tag{30}$$
$$\delta_z=(C/r)^{1/3}\sin\theta(\cos\theta)^{-2/3}$$
$$\times(MS_f''(t)+KS_f(t)+F_G/\sqrt{2})^{2/3}.$$

As given by Eqs. (29) and (30), δ_y and δ_z should be equal to those same components given by Eqs. (12) and (13), in which $F=MS_f''(t)+KS_f(t)+F_G/\sqrt{2}$. For this reason, it must be understood that the relation derived in Section 3 is valid only when deformations of the kind shown in Fig. 7 need not be considered, and the static storage of energy in the wall may be carried over without change to the case of the moving wall. In Eq. (30), Q relates to a torque to be supplied by the turntable motor in rotating the disc recording.

In reproduction, the spherical stylus should always make contact with the wall, requiring $F>0$. This requires

$$|KS_f(t)+MS_f''(t)|<F_G/\sqrt{2}. \tag{31}$$

Using this relation, we obtain

$$F^{2/3}=(F_G/\sqrt{2})^{2/3}\{1+(2/3)[KS_f(t)+MS_f''(t)] \tag{32}$$
$$\div(F_G/\sqrt{2})-(1/9)[KS_f(t)+MS_f''(t)^2$$
$$\div(F_G^2/\sqrt{2})+\dots\}.$$

Rewriting $F_G/\sqrt{2}$ as f, and using the relation

$$\delta_o=[f^2/rH^2]^{1/3},$$

then, with the help of Eqs. (17), (23), (24), and (25), one expands (20) into a series as follows: One denotes

$$S(t)=-\delta_o+S_f(t),$$

and the series for $S_f(t)$ is

$$S_f(t)= \tag{33}$$
$$-\delta_o\{(2/3)[KS_f(t)+MS_f''(t)]/f-(1/9)$$
$$\times[KS_f(t)+MS_f''(t)]^2/f^2+\dots\}$$
$$+a[\psi(Vt)+(r\delta_o/6)\{1+(2/3)[KS_f(t)+MS_f''(t)]/f$$
$$-(1/9)[KS_f(t)+MS_f''(t)^2/f^2+\dots\}\psi'']$$
$$+a^2[\{(r/2)-(5\delta_o/6)(1+(2/3)[KS_f(t)$$
$$+MS_f''(t)]/f+\dots)\}\psi'^2+r(\delta_o/6)(1+(2/3)$$
$$\times[KS_f(t)+MS_f''(t)]/f-\dots)$$
$$\times\{r-\delta_o(1+(2/3)[KS_f(t)+MS_f''(t)]/f-\dots)\}$$
$$\times\psi'\psi'''+r^2(\delta_o/36)(1+(2/3)[KS_f(t)$$
$$+MS_f''(t)]/f-\dots)\psi''^2]$$
$$+a^3[\{(r^2/2)-(5r\delta_o/3)(1+\dots)+(\delta_o^2/2)(1+\dots)^2$$
$$+(2\delta_o^2/3)(1+\dots)\}\psi'^2\psi''+(r\delta_o/6)(1+\dots)^2$$
$$\times\{(r-\delta_o)(r-\delta_o(1+\dots))\}\psi'\psi''\psi'''$$
$$+(r\delta_o/12)(1+\dots)\{r-\delta_o(1+\dots)\}^2\psi'^2\psi^{IV}$$
$$+(r^2\delta_o/18)(1+\dots)\{r-(\delta_o/2)(1+\dots)\}$$
$$\times\psi'\psi''\psi'''-(r\delta_o/9)(1+\dots)\psi'^2\psi''$$
$$+(5r^3\delta_o/648)(1+\dots)\psi''^3]$$
$$+a^4[\dots]+\dots.$$

As a power series in a, the solution $S_f(t)$ is as shown in Eq. (33). Denoting terms of first, second, etc., degrees as $S_{1f}(t)$, $S_{2f}(t)$, . . . , the solution may be written

$$S_f(t)=S_{1f}(t)+S_{2f}(t)+\dots. \tag{34}$$

If we first extract the $S_{1f}(t)$ term, we obtain

$$S_{1f}(t)=-(2\delta_o/3f)[KS_{1f}(t)+MS_{1f}''(t)] \tag{35}$$
$$+a[\psi(Vt)+(r/6)\delta_o\psi''(Vt)],$$

a differential equation for $S_{1f}(t)$. Denoting $2\delta_o/3f$ as $1/k$, we can write it as

$$MS_{1f}''(t)+(K+k)S_{1f}(t)=ka[\psi+(r/6)\delta_o\psi'']. \tag{36}$$

This equation describes the situation, shown in Fig. 8, for which the wall B is in oscillation, with a possibility for resonance, as given by the term $a[\psi+(r/6)\delta_o\psi'']$.

The solution of the differential equation (36) is

$$S_{1f}(t)=A\cos nt+B\sin nt,$$

in which

$$A=-(a/n)\int[\psi+(r/6)\delta_o\psi'']\sin nt\,dt+C_1, \qquad (37)$$
$$B=(a/n)\int[\psi+(r/6)\delta_o\psi'']\cos nt\,dt+C_2,$$
$$n=\sqrt{[(k+K)/M]}.$$

Assuming $k >> K$, and $k\psi >> M\psi''$, and performing the integrals for A and B, we obtain

$$S_{1f}(t)=a[\psi+(r/6)\delta_o\psi''- (1/k)(K\psi+MV^2\psi'') -(1/k^2)(K\psi''+MV^2\psi^{IV})+\ldots]. \qquad (38)$$

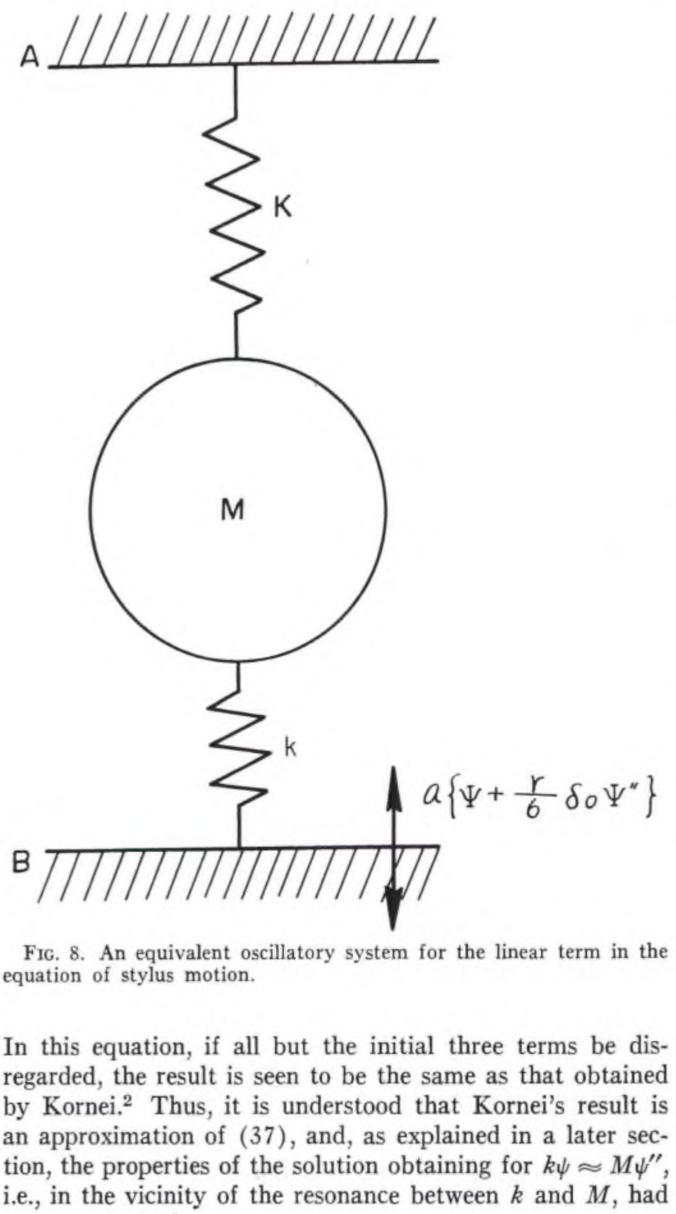

FIG. 8. An equivalent oscillatory system for the linear term in the equation of stylus motion.

In this equation, if all but the initial three terms be disregarded, the result is seen to be the same as that obtained by Kornei.[2] Thus, it is understood that Kornei's result is an approximation of (37), and, as explained in a later section, the properties of the solution obtaining for $k\psi \approx M\psi''$, i.e., in the vicinity of the resonance between k and M, had been overlooked.

Secondly, if the term $S_{2f}(t)$ be extracted, there is obtained

$$\begin{aligned}S_{2f}(t)=&-\delta_o\{(2/3f)[KS_{2f}(t)+MS_{2f}''(t)] \\ &-(1/9f^2)[KS_{1f}(t)+MS_{1f}''(t)]^2\} \\ &+a\{(r\delta_o/9f)[KS_{1f}(t)+MS_{1f}''(t)]\}\psi'' \\ &+a^2\{[(r/2)-(5/6)\delta_o]\psi'^2+(r^2/6)\delta_o\psi'\psi''' \\ &+(r^2/36)\delta_o\psi''^2\}.\end{aligned} \qquad (39)$$

By substituting Eq. (38) in the above, and eliminating terms of order δ_o^2, for the conditions $k >> K$ and $k\psi >> M\psi''$, one obtains

$$\begin{aligned}S_{2f}(t)=a^2\{&(r/2)\psi'^2-\delta_o[(5/6)\psi'^2-(r^2/6)\psi'\psi'' \\ &-(r_2/36)\psi''^2+(r/3f)(K\psi'^2+MV^2 \\ &\times(2\psi''^2+2\psi'\psi''))-(1/9f^2) \\ &\times(K\psi+MV^2\psi'')^2-(r/9f) \\ &\times(K\psi+MV^2\psi'')\psi'']\}.\end{aligned} \qquad (40)$$

In a similar manner, an equation for $S_{3f}(t)$, involving a^3, can be obtained.

The solution (37) may be used to represent general kinds of wave motion, but we now seek the solution for ψ being a sine wave. For this it will be better practice to express the solution as a Fourier series, i.e., in terms of the magnitudes of the fundamental and harmonic component waves, rather than leaving it as a series of ascending powers of a. For this let there be adopted the notation

$$\psi=\cos\omega t \qquad (41)$$
$$S_f(t)=S_1\cos\omega t+S_2\cos 2\omega t+S_3\cos 3\omega t+\ldots.$$

Then, S_1 includes the terms of order a, a^3, a^5, . . . ,
S_2 includes the terms of order a^2, a^4, a^6, . . . ,
S_3 includes the terms of order a^3, a^5, a^7, . . . ,

etc. However, for $a << 1$, then, in S_1, the terms of order a^3 and higher will be negligible compared to that of order a, and, in S_2, the terms of order a^4 and higher will be negligible, etc. As actual modulation amplitudes in disc records, such terms should be very small.

Inserting Eq. (41) into (33), the reproduced fundamental component is found to be

$$S_1=a[1-(r/6)\delta_o\omega^2/V^2][1-j\omega Z_{m1}/k], \qquad (42)$$

in which

$$j\omega Z_{m1}=k[M\omega^2-K]/[M\omega^2-(k+K)], \qquad (43)$$

and Z_{m1} is the mechanical impedance presented to the wall in the model of Fig. 8, and is equivalent to the impedance presented by the network shown in Fig. 9. Figure 10 shows the frequency plot of this impedance for $K << k$. It is,

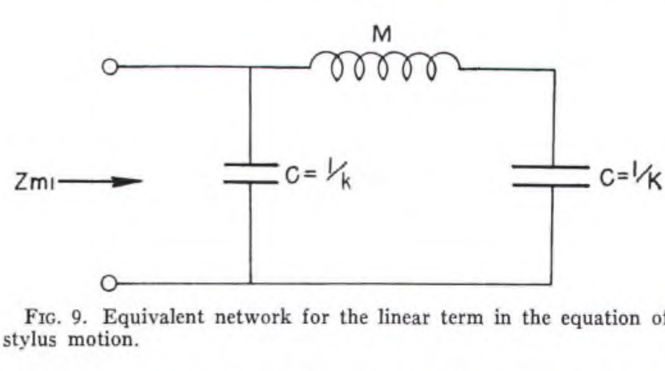

FIG. 9. Equivalent network for the linear term in the equation of stylus motion.

of course, essential in actual playback that the wall stiffness greatly exceed the stylus suspension stiffness, placing the value of ω_c, shown in Fig. 10, in the range between 10 and 20 kHz.

Condition (31) may be expressed, if harmonic components are disregarded, as

$$f > |a(1-r\delta_o\omega^2/6V^2)j\omega Z_{m1}|,$$

which implies

$$|j\omega Z_{m1}| < f/A, \text{ for } A = a[1-r\delta_o\omega^2/6V^2], \quad (44)$$

a condition for the avoidance of intermittent tracing, so there will always be contact with the groove wall.

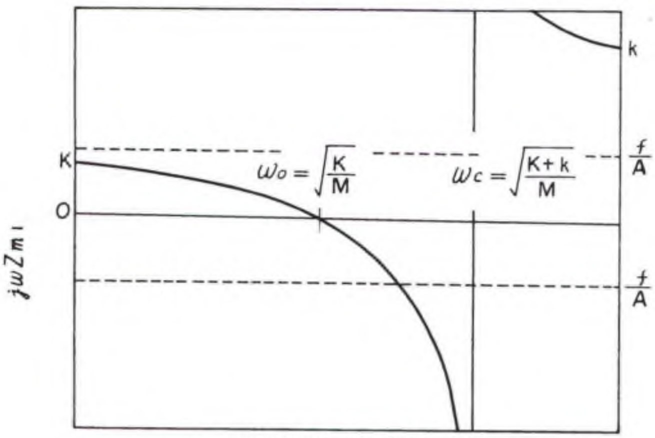

FIG. 10. Frequency plot of the impedance $j\omega Z_{m1}$.

Figure 11 exhibits a frequency plot of $|1-j\omega Z_{m1}/k|$. The response at the resonant frequency ω_c, in actual playback, is always finite, as shown, because of energy dissipation or losses in the resonant system. If the loss term is proportional to velocity, the equation

$$F = f + K_1S_f(t) + R_mS_f'(t) + MS_f''(t)$$

obtains, in which R_m is the resistive coefficient. As shown in Fig. 11, the resistive coefficient controls the blunting of the resonance peak at ω_c, the more so as $\epsilon = R_m/M\omega_c$ is being increased.

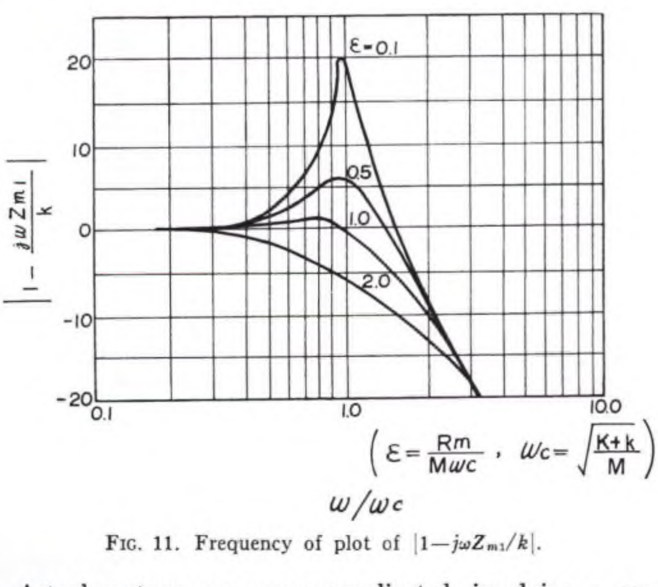

FIG. 11. Frequency of plot of $|1-j\omega Z_{m1}/k|$.

Actual systems are more complicated, involving a cartridge and arm, as well. One should also combine the impedances of these elements with the combined groove-wall-stylus impedance to obtain a more realistic response. In the course of actual measurements, one often finds several other resonance peaks, identifiable as an arm resonance, usually at 100 Hz or lower, together with resonances within the cartridge, depending upon the actual style of construction.

The second-harmonic component S_2 can be obtained in a similar manner from Eqs. (33) and (41). It is

$$\begin{aligned} S_2 = &-(a^2\omega^2/2V^2)\{(r/2)-\delta_o[(5/6)+(7r^2/36) \\ &\times(\omega/V)^2+(1/9f^2)(V/\omega)^2(1-r\delta_o\omega^2/6V^2)^2 \\ &\times(j\omega Z_{m1})^2-(r/9f)(1-r\delta_o\omega^2/6V^2)j\omega Z_{m1}]\} \\ &\times(1-j\omega Z_{m2}/k), \end{aligned} \quad (45)$$

in which

$$j\omega Z_{m2} = k[4M\omega^2-K]/[4M\omega^2-(k+K)]$$

is the mechanical impedance for the second harmonic, with a resonance at $\omega^2 = (k+K)/4M$, i.e., at $\omega = \omega_c/2$.

The various terms in Eq. (45) may be qualitatively described, and estimates obtained (quoted in parentheses) for the relative magnitudes of the contributions from each:

1) $-(a^2r/4)(\omega/V)^2$, a term representing a geometrical distortion present even for a perfectly rigid wall. (6%)

2) $(5a^2/12)\delta_o\omega^2/V^2$, a term representing a reduction in the geometrical distortion because of stylus force. Figure 12 illustrates this action providing for a more faithful reproduction in the elastic-wall case. (1.3%)

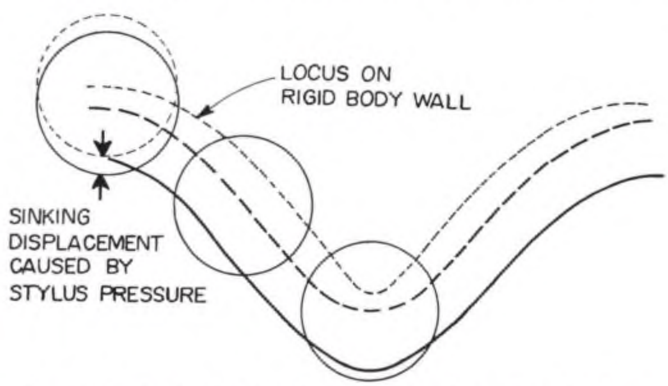

FIG. 12. Reduction in distortion caused by the static stylus force.

3) $(7a^2r^2/72)\delta_o\omega^4/V^4$, a term representing a distortion generated by stylus pressure, also acting to reduce the geometrical distortion, because of a greater deformation when the wall curvature is the less. (0.06%)

4) $(a^2/18f^2)(1-r\delta_o\omega^2/6V^2)^2(j\omega Z_{m1})^2\delta_o$, a term representing a distortion generated by the nonlinearity in the wall stiffness, as shown in Fig. 13, also acting to reduce the geometrical distortion. Upon the application of an elastic force, the drop at A would be less and the rise at B would be greater. Upon the application of an inertial force the rise at A would be greater and the drop at B would be less. Thus, in any case, the sphere traces the locus shown dashed. (0.13%)

5) $-(a^2r/18f)(\omega^2/V^2)(1-r\delta_o\omega^2/6V^2)j\omega Z_{m1}\delta_o$, a term representing a distortion generated by elastic and/or inertial forces acting on the wall, and resulting from a deeper deformation where the wall curvature is the less. It is negative for elastic forces and positive for inertial. (0.006%)

The percentages indicated above are derived from the following set of parameter values:

$a = 2 \times 10^{-3}$ cm	$M = 5 \times 10^{-3}$ gm
$K = 1 \times 10^{6}$ dyn/cm	$\delta_o = 2.1 \times 10^{-4}$ cm
$f = 5 \times 10^{3}$ dyn	$r = 1.7 \times 10^{-3}$ cm
$k = 4 \times 10^{7}$ dyn/cm	$\omega = 4800/s$

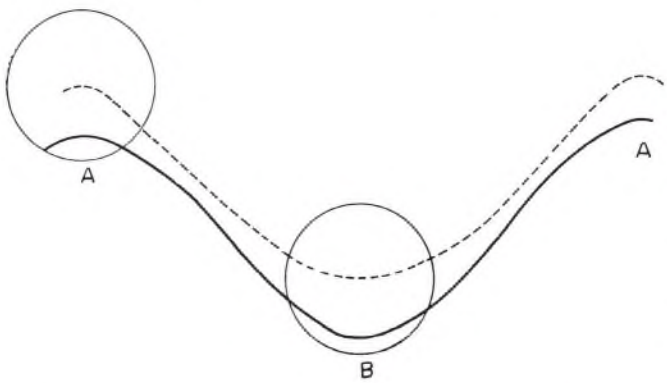

FIG. 13. Distortion caused by the nonlinearity of the wall stiffness.

The ω value corresponds to 800 Hz. The groove-speed value used was $V = 18$ cm/s. These values approximate those for a typical magnetic pickup. The figures quoted in the parentheses are, of course, percentages of the "a" value. These same parameter values have been used to make a plot, shown as Fig. 14, of second-harmonic content *versus* groove speed.

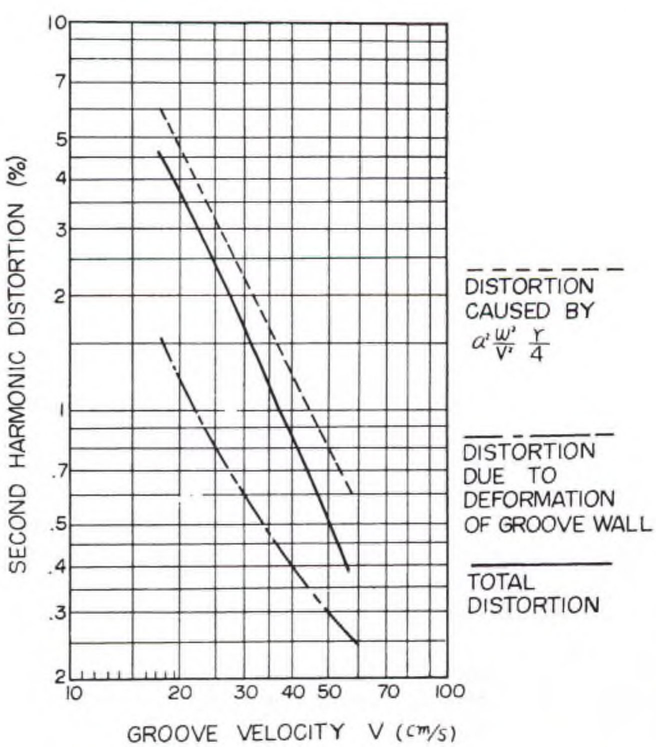

FIG. 14. Second-harmonic content *versus* groove speed.

The third-harmonic component may also be obtained, proceeding in the same manner as before. It is

$$\begin{aligned} S_3 = {} & (a^3/4)(\omega/V)^4\{(r^2/2)-\delta_o[(16r/9) \\ & +(203r^3/648)(\omega/V)^2-(2/9f^2)(V/\omega)^4 \\ & \times(1-r\delta_o\omega^2/6V^2)S_2'(j\omega Z_{m1})(j\omega Z_{m2}) \\ & +(4/81f^3)(V/\omega)^4(1-r\delta_o\omega^2/6V^2)^3(j\omega Z_{m1})^3 \\ & -(5/9f)(V/\omega)^2(1-r\delta_o^2/6V^2)j\omega Z_{m1} \\ & +(r/9f)(V/\omega)^2S_2'j\omega Z_{m2}-(7r^2/54f) \\ & \times(1-r\delta_o\omega^2/6V^2)j\omega Z_{m1}]\}(1-j\omega Z_{m3}/k), \end{aligned} \tag{46}$$

in which

$$S_2' = (S_2/a^2)/(1-j\omega Z_{m2}/k),$$
$$j\omega Z_{m3} = k[9M\omega^2-K]/[9M\omega^2-(k+K)],$$

the latter being the impedance for the third harmonic, with a resonance at $\omega^2 = (k+K)/9M$, i.e., at $\omega = \omega_c/3$. In Eq. (46) the first term represents the geometrical distortion, and the others represent deformation distortions. A plot of third harmonic content *versus* groove speed is shown in Fig. 15 for the same parameter values cited above.

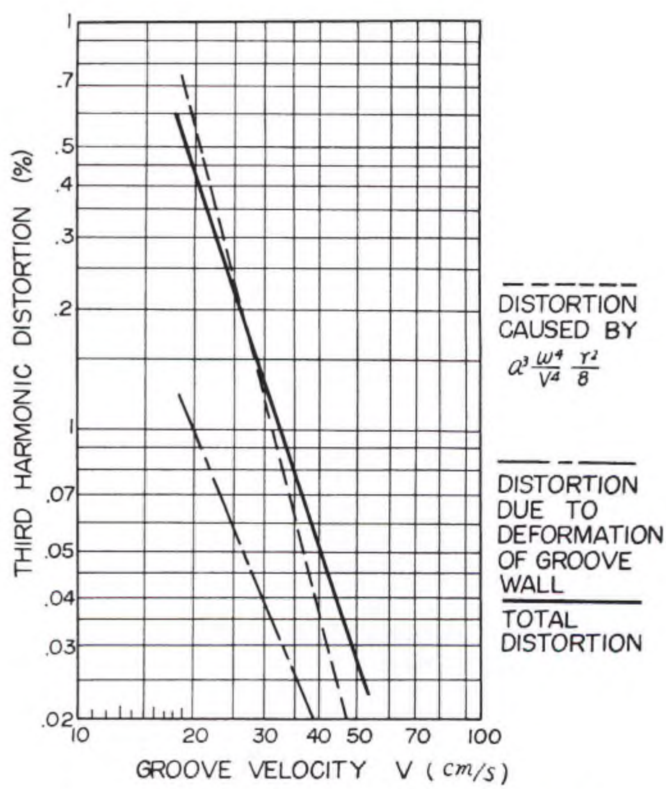

FIG. 15. Third-harmonic content *versus* groove speed.

In addition to showing plots of harmonic content *versus* groove speed, as in Figs. 14 and 15, for a fixed frequency, 800 Hz, it is also possible to show the dependence of harmonic content upon frequency at a fixed groove speed. For this, $V = 50$ cm/s was chosen, together with the fixed amplitude value, $a = 0.4 \times 10^{-3}$, for all fundamental frequencies from 400 Hz to 20 kHz, to prepare the plot shown as Fig. 16.

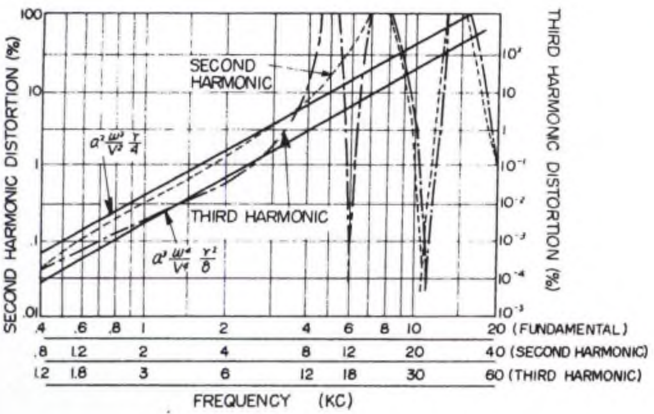

FIG. 16. Variation of harmonic content with frequency for $a = 0.4 \times 10^{-3}$ cm and $V = 50$ cm/sec.

FRICTION BETWEEN A SPHERICAL STYLUS AND THE GROOVE WALL

In considering the friction between a spherical stylus and a groove wall, the forces are assumed to act at equilibrium as shown in Fig. 17. Taking the frictional force to

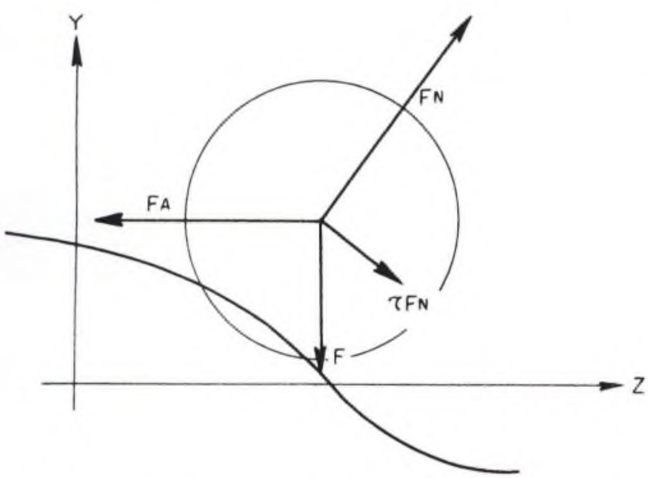

FIG. 17. Resolution of forces including the frictional force.

act in a direction parallel to the tangent to the wall, and denoting the frictional coefficient by τ, one writes the equilibrium condition as

$$\frac{F_N}{\sqrt{[1+(\phi'(\Delta Z))^2]}}\,[1+\tau\phi'(\Delta Z)] = F, \qquad (47)$$
$$\frac{F_N}{\sqrt{[1+(\phi'(\Delta Z))^2]}}\,[\tau-\phi'(\Delta Z)] = F_A.$$

The deformation δ_N, and its components δ_y and δ_z, produced by the force F_N are then given by

$$\begin{aligned}\delta_N &= \delta[1+(\phi'(\Delta Z))^2]^{1/3}[1+\tau\phi'(\Delta Z)]^{-2/3}, \qquad (48)\\ \delta_z &= \delta[\phi'(\Delta Z)-(2/3)\tau(\phi'(\Delta Z))^2+\dots],\\ \delta_y &= \delta[1-(2/3)\tau\phi'(\Delta Z)+(5/9)\tau^2(\phi'(\Delta Z))^2\\ &\quad -(1/6)(\phi'(\Delta Z))^2+\dots].\end{aligned}$$

From Eqs. (8) and (9), one obtains

$$\begin{aligned}S(t) = a\psi\{\Delta Z-\delta[\phi'(\Delta Z)-(2/3)\tau(\phi'(\Delta Z))^2 \qquad (49)\\ +\dots]+Vt\}-\phi(\Delta Z)-\delta_y.\end{aligned}$$

Then, from Eqs. (9) and (48) one obtains

$$\begin{aligned}\phi'(\Delta Z) = a\psi'\{\Delta Z-\delta[\phi'\Delta Z) \qquad (50)\\ -(2/3)\tau(\phi'(\Delta Z))^2+\dots]+Vt\}.\end{aligned}$$

In a manner similar to that by which (14) and (15) were expanded into the series (20), equations (49) and (50) may also be expanded into the series

$$\begin{aligned}S(t) = -\delta+q(Vt)+\{(r/2)-\delta[(5/6)+(5/9)\tau^2]\}q'^2 \qquad (51)\\ +(2\tau/3)\delta q'+(2\tau/3)\delta(r-\delta_o)q'q''+\dots.\end{aligned}$$

This series appears to be identical with (20), except that it has the additional terms involving τ with q', $q'q''$, $q'q''^2$, q'^3, etc., arising because of the friction. Letting $q = a\cos kVt$, the derivatives of q give rise to $a^2\sin 2kVt$, $a^3\sin 3kVt$, etc., all showing a phase shift of $\pi/2$.

The extension of these considerations would demand the oscillator model of Fig. 8 to contain a frictional resistance. Because the friction would also vary in accordance with the modulation of the other groove wall, an additional distortion could be incurred, which would be significant for frequencies near ω_c, where the elasticity of the groove wall plays a dominant role. In this way, a mechanism, due to friction, may be found by which the motions caused by the two groove walls may be shown to be strongly correlated at resonance.

REMARKS

The above represents a part of a paper originally published in Japanese in the *Journal of the Acoustical Society of Japan* in January, 1962. The English-language manuscript was prepared by Professor Duane H. Cooper, at the University of Illinois, from a translation furnished by the author. For this help, the author wishes to express his thanks. Since its original publication, some of the material has fallen out of date, so that it was felt that republishing certain sections of the original article would not be worthwhile at this time. The various sections omitted were concerned with:

1. An analysis of the two-dimensional stylus motions under the influence of tracing error, using a complex-variable approach to obtain a compact representation of the two degrees of freedom. The results obtained were similar to those recently published by Cooper,[8] but were somewhat more extensive.

2. An analysis of intermodulation distortion for tracing error in the absence of deformation. The results agree with those published by Roys[9] and Cooper.[10]

3. An analysis of the effects of the overall curvature of the groove track as it spirals around the center hole of the disc. In effect, each of the two groove walls has a different groove speed. The consequence was a source of distortion that would be the more serious at low frequencies and at inner grooves, and one which conceivably could be enhanced by arm resonance. Present-day arm resonances

in the better systems are below the audio band, however, so that exploration of this question was thought to be somewhat less timely than the deformation problem.

4. A summary for which these remarks will equally well serve was the final deletion.

Apart from these deletions, no attempt has been made to bring this material up to date, or to correlate it with more recent references. The reader may wish to consult the 1963 paper by Kantrowitz.[11]

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THE AUTHOR

Takeo Shiga was born in Tokyo, Japan, in 1924. He graduated from Tokyo University receiving a degree in engineering physics in 1946, and from the same university, a doctorate in applied physics in 1962.

In 1946 he joined Nippon Columbia, Ltd., a company which is affiliated with Columbia Records of CBS, and which is a manufacturer of phonograph records, stereo phonographs, and television receivers, where he has been responsible for the development of loudspeakers, phonograph pickups, and stereo phonographs. In recent years his studies have been of stereophonic sound, the analysis of musical and vocal sound, and the development of acoustical transducers at the Research Laboratories of Nippon Columbia, where he is presently Director of Acoustical Research Development.

Dr. Shiga is a member of the Acoustical Society of Japan and of the Audio Engineering Society.

LETTERS TO THE EDITOR

COMMENTS ON DEFORMATION DISTORTION IN DISC RECORDS

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THE paper "Deformation Distortion in Disc Records" by Takeo Shiga which appeared in the July 1966 issue of the *Journal* (**14**, 208) still appears to be the most nearly definitive work on the subject to date. Although it is more than two years since publication, it would seem that the corrections as given below would help in maintaining the paper's usefulness to interested readers.

Considering the complexity of the original article and the circumstances of its original publication it is a tribute to the staff of The Journal of the Audio Engineering Society that so few errors were found.

CORRECTIONS

Eq. (16), second line: for superscript VI read IV.

Eq. (19): the last line should read $r^2\psi'\psi''[1-\delta_0/r]$.

Eq. (23): for $(r/36)a\psi''^2$ read $(r^2/36)a^2\psi''^2$, and for $(5r^3/648)a^2\psi''^3$ read $(5r^3/648)a^3\psi''^3$.

Eq. (24): for $a^2(r-\delta)\psi'\psi''$ read $a^2(r-\delta)\psi'\psi'''$.

Eq. (26), third line following: for $(1-r/2\rho)/H^2$ read $(1+r/2\rho)/H^2$.

Eq. (33), fourth line on the right-hand side: for $MS_f''(t)^2/f^2$ read $MS_f''(t)]^2/f^2$ (also *second line of 32*); *twelfth line on the right-hand side:* for $(r\delta_0/6)(1+\ldots)^2$ read $(r\delta_0/6)(1+\ldots)$.

Eq. (40), first line: for $(r^2/6)\psi'\psi''$ read $(r^2/6)\psi'\psi'''$; *second line:* for $(r_2/36)\psi''^2$ read $(r^2/36)\psi''^2$; *third line:* for $2\psi'\psi''$ read $2\psi'\psi'''$.

Eq. (44), displayed equation immediately following: for K_1 read K.

Eq. (46), fifth line: for $(1-r\delta_0{}^2/6V^2)$ read $(1-r\delta_0\omega^2/6V^2)$.

Performance Characteristics of the Commercial Stereo Disc*

JOHN EARGLE

RCA Records, New York, New York

This study examines the state of the art in processing master lacquers through the intermediate stages of metal plating on through to vinyl pressing. Detailed noise and distortion measurements at each interface are described, and the effects of normal variations in processes as well as changes in materials are noted.

INTRODUCTION All too often, the audio engineer's concern with phonograph records ends with the cutting of the master lacquer disc, the point at which the disciplines of metal plating and plastics molding take over. The purpose of the present study is to fill this void somewhat by examining these subsequent processes for their effects upon the performance parameters which are important to the audio engineer: noise, distortion, and frequency response. Measurements made on master lacquers, metal positives (molds), and vinyl pressings, will be described, and changes in system input, materials, and processes will be observed. In addition, gross mechanical features which affect performance will be studied. Finally, fundamental process limitations will be discussed with recommendations for improving performance.

TAPE-TO-LACQUER TRANSFER

Although this operation has benefited from the sophistication of tracing distortion simulation, continuing development work in the areas of cutter and lathe design, and a good deal of penetrating mathematical analysis and speculation, there is still much about it that is poorly understood. The cutting action itself, a combination of shearing, embossing, and burnishing, varies with groove and stylus velocities, depth of cut, and the temperature of the heated cutting stylus. The effective vertical cutting angle is an elusive thing; different measurement methods yield different values for it. The effects of lacquer deformation and elasticity during cutting are only partially understood, as are the effects of groove wall deformation during playback of either a lacquer or vinyl record.

It has long been observed that a properly cut master lacquer exhibits a dynamic range greater than any other storage medium used in recording. Indeed, the master lacquer may tax the abilities of a state-of-the-art phonograph preamp. The low noise is probably the result of stylus burnishing action as well as of the stylus heat. The recording stylus does not normally have a feather edge; rather, it is blunted on each edge to form what are called burnishing facets. These facets are angled out a few degrees from the direction of disc travel so that they perform a wiping, or burnishing, action on the freshly cut surface. Too wide a burnishing facet gives excellent burnishing and improved stylus life, but poor high-frequency response. Too small a facet gives insufficient

* Presented April 30, 1969 at the 36th Convention of the Audio Engineering Society, Hollywood, California.

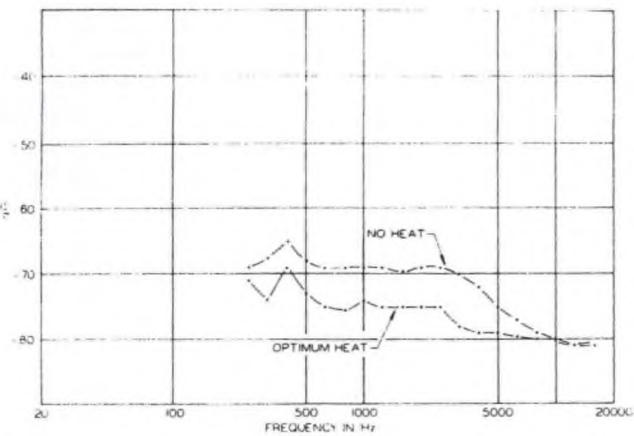

Fig. 1. Effect of stylus heat on playback noise. Recording diameter = 10 in.

burnishing, reduced stylus life, and superior high-frequency response. Obviously a compromise must be made; burnishing facet widths are in the range of .00015 to .0002 in.

Heating current is passed through a coil just above the stylus tip. The amount of current is usually determined empirically; a common rule is to use no more current than necessary to reduce the playback noise to some standard operating level. Figure 1 shows the effect of stylus heat on the playback noise spectrum at a recording diameter of 10 in. Depending on the condition of the lacquer material and the cutting diameter, the effect can be far greater than shown here. Figure 1 may be considered typical for a "well-behaved" lacquer-stylus combination. The noise reduction likely extends out to the highest frequencies of interest, but this effect appears to be wiped out by the finite area of lacquer-stylus contact. This is of course due to deformation of the relatively soft lacquer material.

The measurements shown in Figs. 1 to 5 were made from a single-channel output of a Stanton 681 cartridge with a nominal 0.7 mil tip radius. Melcor 1731 operational amplifiers, connected for RIAA response, were fed into a General Radio 1554-A Sound and Vibration analyzer set for third-octave response. The analyzer was connected to a General Radio 1521-B Graphic Level Recorder. The tone arm was mounted on a Neumann AM-32 lathe. The ordinate on all graphs represents the

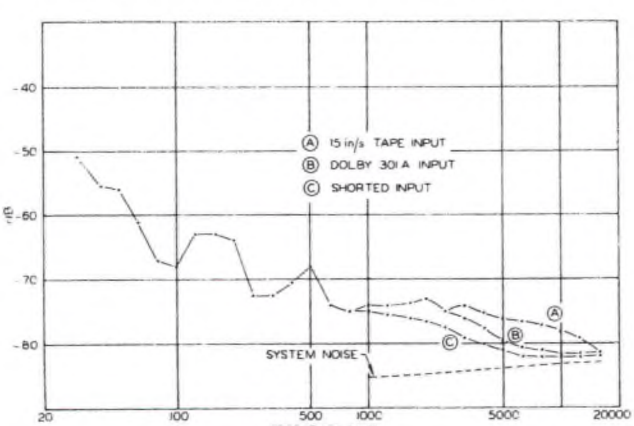

Fig. 2. Noise spectra of a master lacquer at a diameter of 8 in. **Curve A:** 15 ips tape input. **Curve B:** Dolby 301 input. **Curve C:** Shorted input.

level below a peak velocity of 5.5 cm/sec at 1 kHz.

Figure 2 shows typical noise spectra for a master lacquer at a recording diameter of 8 in. The effects of various system inputs can be clearly seen. A single generation of biased low-noise formulation tape with NAB 15 ips playback (50 μ) is shown as Curve A. System gain was set for "normal" transfer level, that is, at a setting such that an Ampex standard reference level 1 kHz tone yields a lateral cut on the disc of 5.5 cm/sec peak velocity. Curve B is the same tape played through a Dolby 301-A noise reduction system, and its effect is quite noticeable above 3 kHz. The difference between Curves B and C is clearly the contribution of associated circuitry in the channel up to the input of the cutting amplifier. Curves A and B represent "ideal" cases of only one generation of input noise. Normally, there would be several generations and the effects of noise reduction would be more pronounced.

Below 1 kHz the noise spectrum turns upward. The modes at 500 Hz and 150 Hz seem to be characteristic of the playback environment and are made obvious by the selective nature of the playback system.

The Metal Mold

Figure 3 shows the noise spectra of a metal mold made from the lacquer of Fig. 2. This metal part is

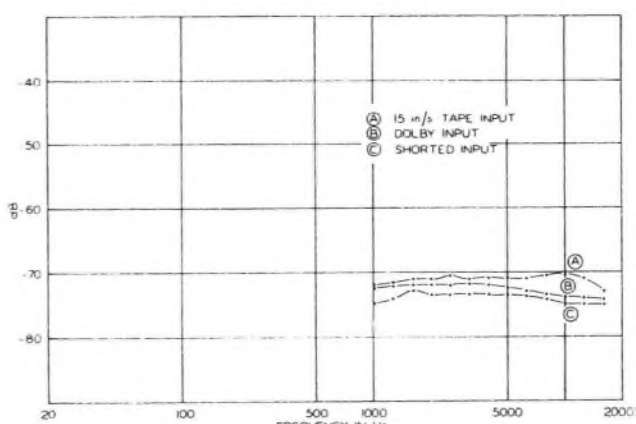

Fig. 3. Noise spectra of a metal mold. **Curve A:** 15 ips tape input. **Curve B:** Dolby 301 input. **Curve C:** Shorted input.

twice removed from the master lacquer. The intermediate steps were: silvering of the lacquer surface to make it conductive; plating this surface with nickel; stripping the nickel-silver combination; passivation of the metal surface; replating with nickel; and, finally, stripping metal from metal. It is obvious that successive replications of the original lacquer surface are alternately positive and negative; the metal mold is the first positive part after the lacquer.

The most striking aspect of Fig. 3 is the apparent rise in noise above 1 kHz relative to the lacquer. Note also that the same distinctions as before can be made regarding input noise. If the apparent increase in high-frequency noise on the mold were the result of processing, then the distinctions of system input noise would surely tend to be swamped out. This is obviously not the case, since the input conditions are equally evident in

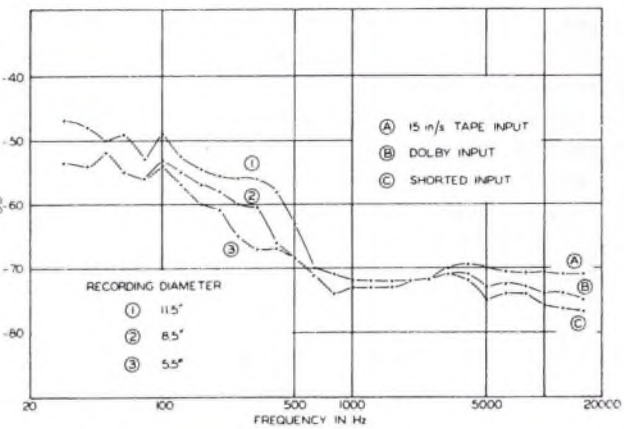

Fig. 4. Noise spectra of a vinyl pressing. **Curve A:** 15 ips input tape. **Curve B:** Dolby 301 input. **Curve C:** Shorted input. Recording diameter at **1** = 11.5 in., at **2** = 8.5 in., at **3** = 5.5 in.

Fig. 2 and 3. What begins to be obvious here is that the transfer to metal positive has added very little noise, and that the apparent increase is due to the lack of groove wall deformation which would tend to roll off high frequencies as in Fig. 2. In other words, we are probably getting our first good look at what was put into the groove. A final note on the metal mold: The parts can never be made to lie flat on a turntable. Therefore, no noise measurements could be made at low frequencies.

The Vinyl Pressing

A third replication is made in nickel, resulting in the stamper which will be used in the pressing operation. The stamper is centered and its edges crimped to give the characteristic contour to the pressing. In addition, the back of the stamper is ground down to remove the many metallic crystals which would otherwise prevent the stamper from making intimate contact with the die in the press. The grinding operation on the stamper, as well as the face of the die itself, result in the familiar "orange peel" look, or mold grain, noticeable on most pressings.

Figure 4 shows the various noise spectra of a typical pressing. Notice that contributions of the various input conditions are still evident. As in Figs. 2 and 3, the high-frequency noise measurements were made at a diameter of 8 in. Notice the effect of diameter on noise below 500 Hz; this is certainly due to mold grain, and may easily be seen to be an inverse function of diameter.

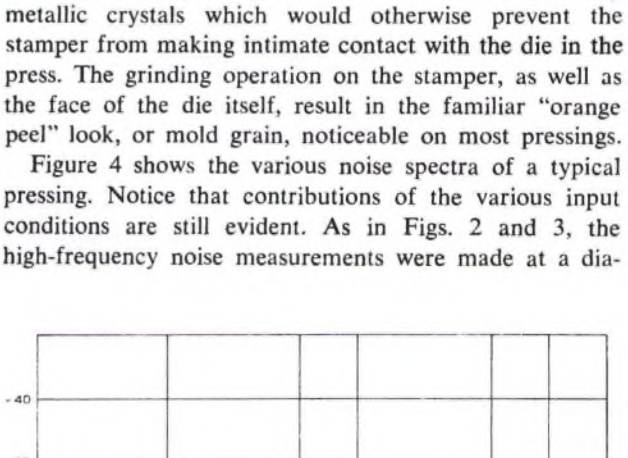

Fig. 5. Noise spectra of 7 in. "filled" compound pressing. Recording diameter for **Curve A** = 11.5 in., for **Curve B** = 8.5 in., for **Curve C** = 5.5 in.

Figure 5 shows the noise spectra of a 12 in. record pressed on a "filled" 7-in. type compound. Both organic and inorganic fillers were used, and the drastic increase in high-frequency noise is clearly evident. Note that, like mold grain, the noise level tends to fall off at inner diameters, very likely for the same reason that there are fewer disturbances passing the stylus per unit time at the inner as opposed to the outer diameter.

At this point it is interesting to look at representative wideband noise measurements. The NAB standard for noise measurement of discs examines the velocity output over the band from 500 to 15,000 Hz. By this method the data of Curve C in Fig. 4 yield a value of -64.5 dB referred to 5.5 cm/sec peak velocity. Since the NAB reference level is 7 cm/sec peak velocity, the noise level is thus -66.5 dB below reference. This is considerably better than the NAB standard of -55 dB. By a similar process the NAB noise reading for Curve A in Fig. 4 is -62.5 db.

DIAMETER LOSSES

From the very beginning of the disc art, technicians have observed a reduction in high-frequency response at reduced recording diameters. With electrical recording it has been possible to correct for this loss to some extent. At the present time, however, there is no standard way of dealing with the problem, complicated as it is by the present-day variety of playback styli, the use of tracing correlation, and, most important, the power limitations of the amplifiers used in disc cutting. Diameter losses are of three types: cutting losses, tracing losses, and deformation losses.

Cutting losses are the result of the finite width of the burnishing facets of the cutting stylus. At the short wavelengths which occur when high frequencies are recorded at reduced diameters, there is a certain amount of "self-erasure" of the signal due to the burnishing facet. There is no direct playback measurement of this loss; only optical measurement by way of light patterns can isolate it.

Tracing losses are fundamental to the disc playback process. They would be present even if cutting and deformation losses were non-existent. They are due simply to the fact that tracing distortion, as wavelengths grow short, gives rise to a *reduction* of signal fundamental as more of the signal is converted into harmonics.

Deformation losses far outweigh tracing and cutting losses. They result from the yielding of the vinyl or lacquer material under the force of the playback stylus. At present there does not exist a clear mathematical model of this process.

Figures 6 and 7 show diameter losses with a 0.7 mil radius stylus as well as with an elliptical stylus with a 0.2 or 0.3 mil bearing radius. The signals measured in this set were cut with a constant current at all diameters; that is, there was no attempt to compensate for the losses either by tracing correlation or by direct electrical boosting. Note the remarkable similarity, in both Figs. 6 and 7, between the diameter losses for both lacquer and

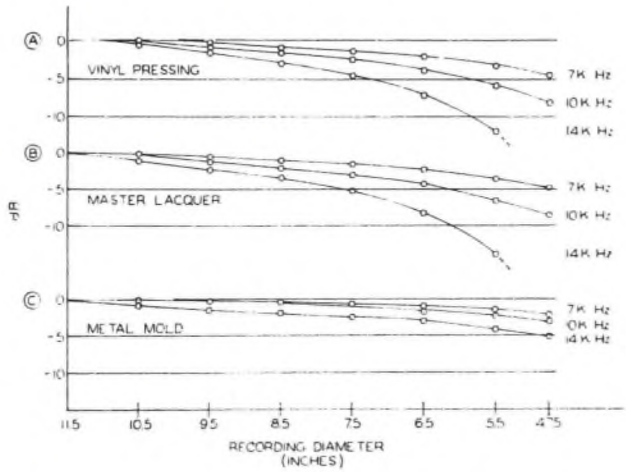

Fig. 6. Diameter losses with 0.7 mil radius playback. **Curves A:** Vinyl pressing. **Curves B:** Master lacquer. **Curves C:** Metal mold.

pressing. Apparently there is ample justification for the standard practice in the phonograph record industry of using reference lacquers in making musical judgments of program material at all steps prior to pressing.

Curves C in both Figs. 6 and 7 show playback losses associated with metal molds. Here there are no deformation losses, and the drop in high-frequency response at inner diameters can be ascribed solely to cutting and tracing losses. The tracing losses, in particular, are a function of the recorded level. In this case the signals are all recorded nominally at 15 dB below the 5.5 cm/sec reference level on an RIAA basis.

These two sets of curves raise several questions regarding the use of diameter compensation in disc recording. For many years most record companies have used some degree of diameter equalization in the cutting of masters. Tracing correlation itself provides some degree of compensation, and one could additionally compensate electrically for the cutting losses caused by the burnishing facets. However, any attempt to correct for deformation losses, especially for 0.7 mil radius playback, is out of the question. The combination of high-frequency pre-emphasis and high levels common in today's cutting are already taxing the power output capabilities of cutting amplifiers. There is very little room left unless one resorts to some kind of high-frequency limit-

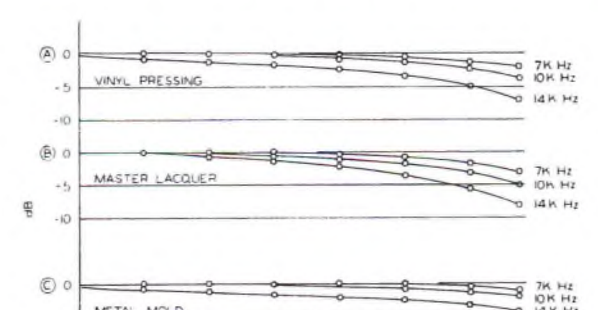

Fig. 7. Diameter losses with elliptical playback. **Curves A:** Vinyl pressing. **Curves B:** Master lacquer. **Curves C:** Metal mold.

ing. It should be obvious that small burnishing facets reduce the degree of cutting loss at short wavelengths. (The cuts measured in Figs. 6 and 7 were made with a fresh stylus with a nominal .00016 in. burnishing facet width.) One could easily monitor the size of the burnishing facet indirectly by comparing inside and outside cuts at some high frequency for each new stylus and observing the spread between them. This of course requires a playback stylus of known dimensions.

RADIUS OF GROOVE BOTTOM

Normally the radius at the bottom of the groove has no effect on the performance of the record since it is sufficiently smaller than the playback radius. For safety's sake it is wise to keep the nominal bottom radius no greater than about half the smallest nominal playback radius likely to be encountered. The reason is simply to avoid any overlap of the respective frequency distributions of radii, with consequent bottoming of the stylus in the groove. The most economical and effective method for controlling this is not by direct monitoring of the stylus radius during use but rather through establishing an arbitrary lifetime for styli. An earlier study showed that most styli, whatever their starting radius, tended to end up at about .00035 in. radius after about 25 hours' use. The results of a subsequent study are shown in Fig. 8. Here, the styli were used for no longer than 15 hours, and the result is a distribution that agrees favorably with the industry standard of .00025 in. radius.

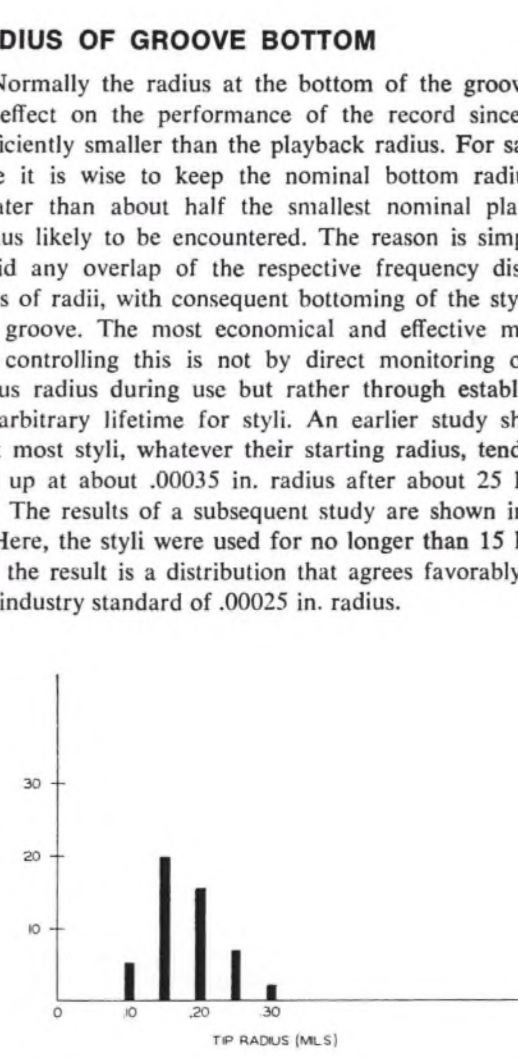

Fig. 8. Frequency distribution of tip radii for a stylus life of 15 hours (47 samples).

DISTORTION

Distortion measurements on lacquer, mold, and pressing were made by examining first-order sidebands produced by the intermodulation of 4 kHz by 400 Hz. Readings are presented for one channel only from a vertical cut. Distortion is expressed as the *sum* of the sidebands as a percentage of the carrier; the sidebands measured in this way are the result of both AM and FM. The 400/4 kHz combination was cut at a 4:1 velocity ratio, with the recorded level 4 dB above NAB reference level on an RIAA basis. (NAB reference level is 7 cm/sec @ 1 kHz.) Thus, the cut represents a normal maximum cutting level for today's LP records. These distortion figures appear fairly high, but they are normal for mea-

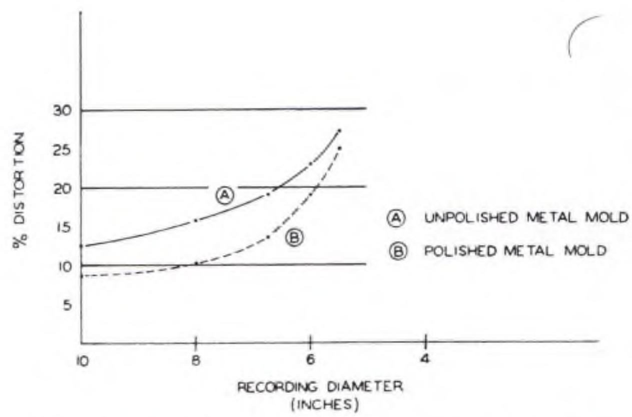

Fig. 9. Intermodulation distortion measurements, 400/4000 Hz, with 0.7 mil radius playback. **Curve A:** Unpolished metal mold. **Curve B:** Polished metal mold.

surements made in this way. The vertical nature of the cut tends to emphasize *differences* in the cutting and playback vertical angles. Even though major companies in the record and phonograph cartridge industries claim that both their products are designed around a 15° vertical cutting/playback angle, observed values tend to be on the high side. Also, it is amazing to observe 8° to 10° discrepancies between "nominal" 15° cutting and playback angles. There can be no question that a design center of 15° is too low a value to be practical in both cartridge and cutter design, and this is a problem which will have to be resolved sooner or later. For the present, however, it seems to be conveniently ignored by everyone concerned.

It was surprising to observe that the intermodulation distortion remained nearly constant from lacquer to mold to pressing. The progression of values from outer to inner diameters is shown in Curve A of Fig. 9. The lower values of distortion measured on the polished mold and a pressing made from it (Curve B of Fig. 9) are indeed interesting. Polishing of metal molds is standard in the record industry. It is a technique for removing "horns" from the metal part in order to facilitate the molding of vinyl pressings. Horns are tiny ridges of material at the groove edges thrown up by the stylus. They are transmitted through the first two replications and are present on the metal mold. At this stage they can be removed effectively by applying a fine abrasive to the metal part as it spins on a wheel. When done carefully, this technique results in a wearing away of the horns, which protrude above the disc surface, while barely affecting any portion of the groove below. It should be remembered, however, that the cut being measured here is *vertical*. At its narrowest point, the cut was approximately .0015 in. wide, and it is apparent that a certain amount of abrasion took place at these intervals over portions of the groove wall later to be contacted by the stylus. Since the first-order sidebands are the result of asymmetrical, or single-ended, aspects of a transfer system, one must look for a similar action at the widest, or deepest, part of the groove if one is to explain the reduction in sidebands.

There does appear to be an action in cutting which could cause the observed sideband reduction. It has been observed that, keeping all else equal, the amplitude of a high-frequency signal tends to lessen very slightly as the cut is made deeper; this appears to be due to the increased load on the stylus. Such an action would be symmetric with the amplitude reduction at the crest of the wave and could thus account for a reduction in sidebands resulting from asymmetry.

RECORD WARPAGE

There are three kinds of warpage which can afflict the vinyl pressing and degrade its performance:

"Dish" Warpage.—In this case, the record is shaped like a shallow bowl and, if played by itself, will perform well. However, if two such records are placed on a record-changer back-to-back, there is insufficient friction and slippage may result. Dish warpage appears to be due to minute differences in the press die and stamper thickness tolerances from outer to inner diameter. When such records are placed on a spindle directly after pressing, these minute dimensional errors will all add up in the same direction. The use of flat metal spacers at intervals in the spindle is a good remedy for the problem.

"Saddle" Warpage.—In this case the record is actually bowed along one diameter. The effect is almost always due to improper storage or to warped paper components used in record packaging. The effect is sometimes reversible, especially if the deformation occurs at some later point in the lifetime of the record. Saddle warpage can cause record slippage as well as a once-around "wow" in pitch.

"Pinch" Warpage.—This results from improper removal of the record from the press. If the record is still warm it can be easily twisted at the edge by the press operator, resulting in a permanent set at the edge. The cure is to insure more careful handling at this critical point or to remove the record from the press at a lower temperature.

Pinch warpage is probably the most annoying of the three forms. In extreme cases, it can cause a light-weight tone arm to leave the record altogether; when present to a moderate degree, it can produce an unpleasant once-around "wobble" in pitch. This will be discussed in more detail in a later section.

Figure 10 shows a frequency distribution of dish warpage for 100 samples. For this test, the records are placed concave-side-down on a gauge which allows a pin near the label edge to rise until it contacts the disc. The pin is attached to a dial indicator calibrated in mils. Dish warpage is easy to measure, and such data is useful for press floor control purposes. The two other varieties of warpage are best controlled on a "go—no go" basis.

RECORD CONCENTRICITY

Although a stamper may be accurately centered to begin with, wear and tear in actual use may result in its shifting to one side, depending on the nature of the tooling in the press. As a result, run-out measurements take the form of the distribution shown in Fig. 11. For this group of samples the upper allowable run-out limit is 15 mil, and the lot appears to be well controlled. It is interesting to note industry performance regarding record concentricity. NAB specifications call for run-out not to exceed 10 mil. This is the only NAB specification

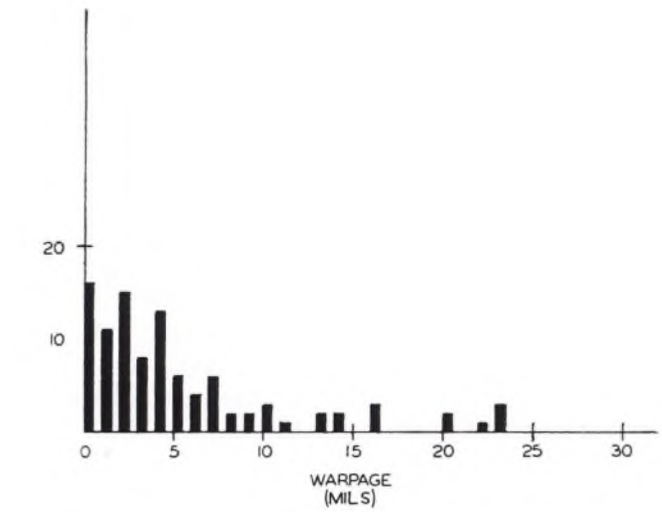

Fig. 10. Frequency distribution of record warpage (100 samples).

which the record industry, both here and in Europe, does not meet. A survey of major domestic and foreign labels indicates that, on classical product, control limits on run-out range from 15 to 20 mil.

TIME BASE ERRORS

The previous discussions of warpage and concentricity lead naturally into the subject of time-base errors in disc recordings. Of all the devices used in the recording transfer processes, none has quite the low flutter and wow of a professional disc cutting lathe. Its performance is better than that of the best tape machines operating at 15 ips. Normally, the program input to the cutting system is considered to be essentially free of time-base errors, and there is nothing inherent in the subsequent processing to vinyl disc which alters this. Only the gross mechanical aspects of warpage and off-center pressing affect the final performance in this area. The effect of record off-center varies with diameter. If a disc exhibits a run-out of 10 mil, the peak frequency deviation will be something slightly under $\pm .1\%$ at a diameter of 5 in. Probably because of the long period of the wow produced by this degree of off-center (about 1.9 sec), the effect on most kinds of music is negligible. With sustained organ or piano passages, a conditioned listener may be aware of the error at an inner diameter. Probably for this reason, the industry has elected to set its own standard, somewhat looser than the NAB one.

The effect of pinch-warpage is usually far greater than off-center effects in producing disturbing time-base errors. The pitch deviation results from the fore-and-aft movement of the stylus tip relative to its normal position. The tone arm itself tends to follow slow variations in the surface of the disc. If the arm is normally horizontal, there is a minimum of fore-and-aft motion and a minimum of wow. However, if there is a sudden disturbance, as would be caused by pinch warp, it cannot be followed by the relatively massive arm; instead, the stylus cantilever itself follows the disturbance. Since the stylus cantilever is nominally at an angle of 15° to the record (often greater), there can be a considerable fore-and-aft component to the motion of the stylus as it negotiates the rapid disturbance. In this regard, today's best highly compliant cartridges are more sensitive than earlier models.

RECORD DIMENSIONAL CONTROL

Only in rare cases will a record misperform because of some aspect of its thickness. Beyond specified limits, it might not trip properly on some record changers. The RIAA recommendation for center thickness is .075 in., + .010 − .015. These tolerances are much broader than those that can easily be maintained in a modern record plant, as shown in Fig. 12. This graph appears to show two modes in the distribution, but this is the result of the small sample size. Clearly, distribution fits well within RIAA recommendations and is a bit on the low side.

CONCLUSIONS AND RECOMMENDATIONS

The foregoing study outlines the performance characteristics and limits of phonograph records. Some of the more important observations are:

1. A well-processed record easily reflects the quality of the system input. This is a strong argument for noise reduction techniques in processing master tapes.

2. Below 1 kHz the benefits of noise reduction in mastering tend to be masked by noises attributed to mold

Fig. 11. Frequency distribution of concentric groove run-out (100 samples).

Fig. 12. Frequency distribution of record center thickness (50 samples).

grain and stamper grinding. Thus, the Mincom and the Dolby noise reduction systems would be equally effective at the present time.

3. The various diameter losses are the worst aspects of disc performance. The rapid deterioration of high-frequency response and increase in distortion below a recording diameter of 6.5 in. would suggest that record manufacturers avoid the tendency to "fill up a side," and instead use the most efficient kind of pitch and depth control in order to maintain a large ending diameter.

4. Whatever may be said about record wear, the best performance of records certainly results from the use of elliptical playback styli.

Some recommendations which would improve product performance are:

1. No discussion of record performance can be complete without at least a passing reference to the ticks, pops, and swishes which are a part of so many records. There are more than enough good records around to make it obvious that these defects are not inherent in the product but rather the result of improper control and decreased processing time. Improved performance in this area is probably the greatest challenge facing the industry at this time.

2. More work needs to be done by the industry in matching cutting and playback angles. A higher value than the nominal 15° is felt by many in the field to be better than 15°, but little is being done toward setting such a standard.

3. The correction of the effects of groove wall deformation by means similar to tracing correlation has been suggested. If the correction is made for a 0.7 mil radius playback stylus, then the degree of correction would appear to tax the electrical limits of any known lacquer transfer system, and the scheme should be approached cautiously. Standardization of elliptical playback styli would eliminate many such problems, but there is a serious question of record wear to be resolved.

4. Finally, were it not for the thorny problems of compatibility with existing players, the prospect of a disc with "built-in" noise reduction would certainly be a pleasant one.

ACKNOWLEDGEMENTS

The author wishes to thank the many people in the manufacturing and engineering sections of RCA Records who helped make this study possible. Special thanks go to Michel Pradervand of Record Engineering Section for his painstaking work in making all noise measurements.

Note: Mr. Eargle's biography appeared on Page 281 of the June, 1969 issue of the Journal.

A New Profile for LP Records*

WARREN REX ISOM

RCA Records, Indianapolis, Ind.

A new dimension has been found for the thickness of the music groove area of the long playing 12-inch stereo record. The 30-mil thickness instead of 50 mils provides turbulent mixing during the molding of the record for better groove formation and increases the flexibility of the record for its rotational engagement with the turntable. This 90-gram record development is reviewed in full.

INTRODUCTION: The profile of the cross-sectional area which is perpendicular to the surface of the record has been changed in order to enhance the moldability and flexibility of the record. The new profile results in a new dimension for the music groove area. This new profile is completely compatible with all record-playing instruments now in use throughout the world and is within the dimensional standards established by the Record Industry Association of America (RIAA).

Fig. 1 shows the new profile in comparison with the standard profile. Note that the thickness of the record at the bead around the circumference is unchanged. The typical thickness of the music area for the standard profile is 50 mils. The new dimension for this area for the new profile is 30 mils.

The transition angle between the lead-in ramp and the playing surface of the standard profile is 175°, leaving a 5° break between the surface of the lead-in area and the music groove area. In the new profile, this angle becomes 2° at the point indicated in the drawing as 175°. There is a secondary blend between the two surfaces in the music area shown in the drawing at the point marked 177°, representing a change of 3°. This point is well within the music groove area.

* Presented October 13, 1970, at the 39th Convention of the Audio Engineering Society, New York.

THE BREAK POINT IS BLENDED WITH A FILLET

These two new angular dimensions in reality define a fillet between the lead-in ramp and the playing surface of the record. There are two reasons for this fillet.

1) Without the fillet, the transition ramp from the top of the bead to the surface of the new profile record would extend into the music area. A 5° break in the

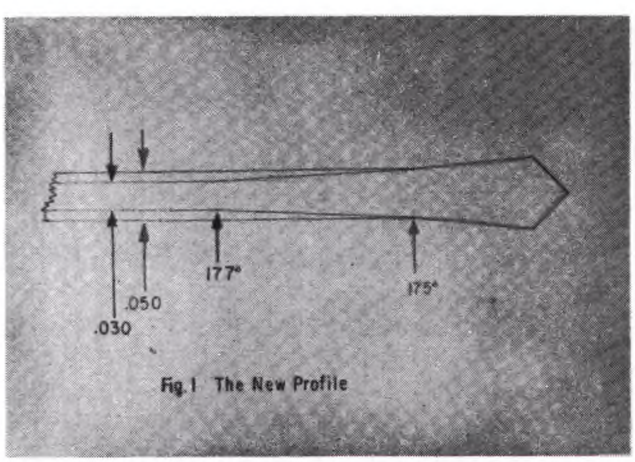

Fig. 1. New profile.

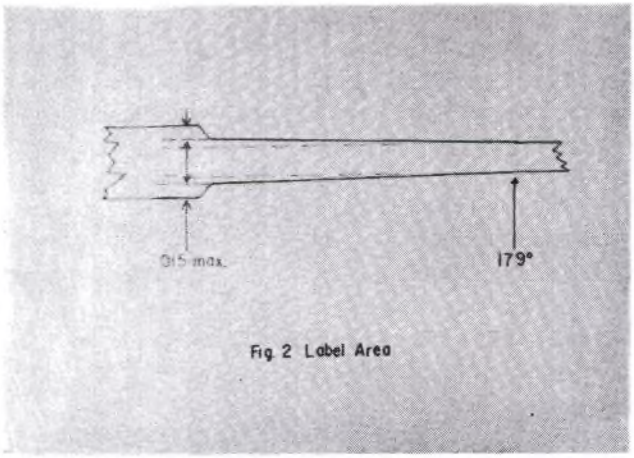

Fig. 2. Label area.

Fig. 3. Standard profile, layered pressing.

surface cannot be accomplished without effect on the sound. Blending this break point with a fillet reduces the change in angle in the music area to 3°. This change can be accomplished with no measureable effect on the sound. The 3° angle is well within the tolerance of the vertical angle adjustment of the stereo pickup. Measurements on a frequency test record made with the new profile gave, in the worst instances, differences of ½ dB between the two channels. In most instances when the measurement was made after the pickup vertical angle was adjusted, there were no differences between the channels.

2) The fillet provides the same clearance between the record surface and the pickup housing that has been established by the standard profile. For pickup cartridges that have housings less than 5/16-inch total width, the clearance between the housing and the lead-in groove ramp is equal to, or greater than, that previously enjoyed.

Fig. 2 details the difference in the label and lead-out groove area. There is a 1° ramp in the lead-out groove area to raise the surface to that dimension specified in the RIAA standard for records. This transitional ramp again maintains the same clearance between the pickup housing and the surface of the label area that has been in existence for many years.

135 — 90 = 33% OF IMPROVEMENT

The new profile record with a typical thickness of 30 mils gives a design objective record weight of 90 grams. This weight is exactly ⅓ less than the design objective weight of the standard profile. For many people, this difference in weight is the complete reason for making the change. This obvious answer, however, is completely opposite to that which happened. Experience, experimentation, and theory unanimously indicated that the moldability of a record is improved when its thickness has an optimum dimension. This is contrary to the thick-record tradition established in the shellac era as a hedge against breakage which identified thickness with quality.

The incentive to undertake a program to improve the modability of records has existed in the American record industry for several years. There has been progressive improvement in the vinyl compound itself, but outstripping this is the march of progress made by the electronic industry. This progress has made available to a great number of people highest fidelity playback equipment at very nominal prices, which on each play examines the record surface with the acuity of an electronic microscope and announces each minute fault with a shouted count that penetrates through the music to the annoyance of the listener.

THE PROOF IS IN THE MIXING

The improvement in moldability is dramatically shown by the comparison of Figs. 3 and 4. Each record was made from a layered preform consisting of five thin discs, each of a different color, with a total thickness

Fig. 4. New profile, homogeneous mix.

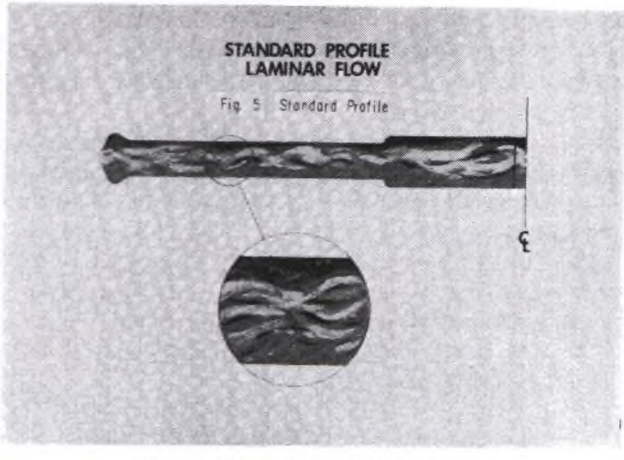

Fig. 5. Standard profile, laminar flow.

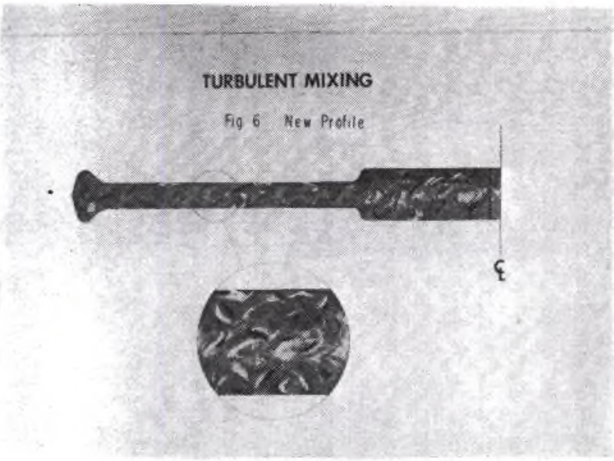

Fig. 6. New profile, turbulent mixing.

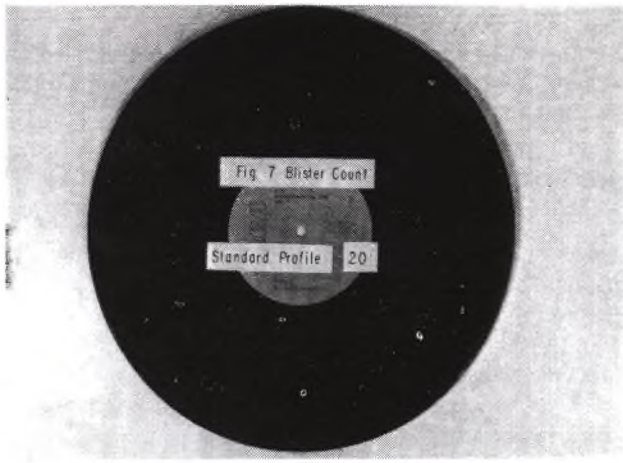

Fig. 7. Standard profile, blister count: 20.

and weight equal that of a normal preform. The standard profile record shows on the surface the layered nature of the preform. The new profile pressing surface reveals a homogeneous mix. The record is almost wholly black. This obviously shows that the compound was more completely mixed when pressed into the new profile record.

The examination of the improved moldability of the record was continued by viewing a cross-sectional area of each under the microscope. Fig. 5 shows the laminar characteristic of the cross-sectional area of the standard profile. Fig. 6 represents the dispersed array of the residual color in the new profile. The flow during molding of a standard profile was laminar. The flow during molding of the new profile is revealed as turbulent. A drawing is used to represent this to give nonsymmetrical enlargement of the thickness of the cross-sectional area. To this degree, the drawings are interpretive.

BOUNDARY LAYER INTERFERENCE IN RECORD MOLDING

This microscopic examination reveals that the reduced thickness of the music groove area provides for better molding of the record. The turbulent mixing is brought about, it is thought, by the interference between the boundary layer of the compound that adheres to the top stamper with that which adheres to the bottom stamper. This turbulence is greatest at the final critical instant when the record is given its permanent form. This turbulence provides for the release of the entrapped air and of the volatiles in the compound. This assures the elimination of the tiny blisters under the surface that tic and pop during the playing of the record.

An accelerated blister count test was run on ten records, each of the two profiles. The records were put in an oven at the softening temperature of the compound for several hours and then the blisters that developed were counted. Figs. 7 and 8 show the results. The typical count for the standard profile was twenty blisters per record side. The typical count for the new profile was no blisters. A total of only eight blisters was found on the twenty sides of the new profile.

The final turbulent mixing which gave the homogeneous, almost solid black record from the color layered preform means that the new profile record is free to a very great extent of molding flow striations which produce once around swishes during playback. The same flow characteristics fully fill the music grooves and provide a smooth surface for the pickup stylus. This is further enhanced by the higher pressure required at the end of the molding cycle for the new thickness. In addition to that, the record in the press has less thermal capacity and assumes the temperature of the press molds uniformly throughout the groove area. This eliminates localized internal strains in the record and affects favor-

Fig. 8. New profile, blister count: 0.

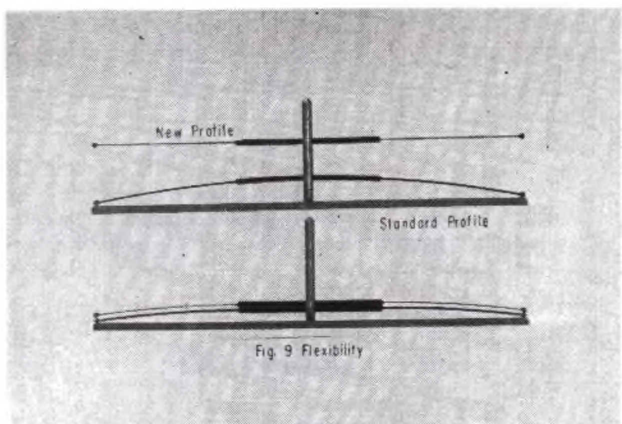

Fig. 9. Flexibility.

ably the surface conditions.

All of the preceding observations have been verified in the factories during the course of producing more than five million records with the new profile.

SURPRISE

An added advantage of the new profile came as a surprise. It was known that the new profile would have greater flexibility. The surprise was that this flexibility has two great advantagees.

1) When the new profile record is dropped on top of another record that is convex, the new profile record conforms and plays without slippage. This is tremendously advantageous.

2) The music groove area is not strong enough to warp the label area of the record when it is stored in a most adverse situation. In this worst case condition, the record plays without slippage because of the good contact in the label area.

Figs. 9 and 10 illustrate these two surprise advantages.

WHAT IS WRONG?

The new profile does have some negatives. In production, in the factories, these negatives are the same as those that accompany all changes. Substituting the new profile for the standard profile in production required cyclical changes to compensate for the difference in record heat capacity. It took a while to accomplish this. The new record must be handled differently in production. Press operators were quick to learn the technique of removing the record from the presses without pinch-warping them with their grip. Some problems may still turn up, but if they do, they will be solved and the solution will become part of the method of record making.

5 000 000 NEW PROFILE RECORDS

Five million records were distributed in the marketplace without notice to the customer. There have been a few hundred letters pointing out that the record is thin. Sometimes the phrase "paper thin" is used. Maybe a third of these respondents are sure that they have been robbed of vinyl to the enrichment of the manufacturer.

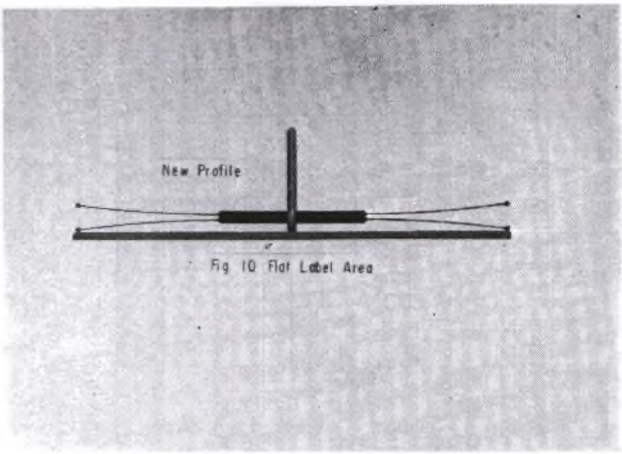

Fig. 10. Flat label area.

A relatively few have complained of genuine faults. Obviously, bad pinch warp does not go unnoticed. The reaction from the field to this new profile may be indicated by a quote from a reviewer, "In an apparent effort to save money on vinyl, this company (RCA) is making records almost as skinny as postcards. Consumer reaction so far is vaguely negative, but more important than getting cheated out of some plastic is the fact that these records seem to have very good surfaces."

The use of less material to manufacture a record of the highest quality promotes customer interest in two ways.

1) This may be a most important quality breakthrough. With the new compound, the very best compound can be economically used for all LP records without regard to the selling price.

2) At a time when costs are increasing on every hand, developments that avoid the full impact of these increases are of great importance to the customer. Increased manufacturing costs are somewhat neutralized by the smaller amount of compound used in the record. Rising shipping costs are largely offset by the less weight of the product.

This new profile for LP records is an advance in the art and science of manufacturing records.

THE AUTHOR

The professional career of Warren Rex Isom has encompassed the field of recording by work on wire, film, magnetic, kinescope, pre-detection space and disc recording. He has largely concentrated in its neglected aspect—the mechanics of recording and holds thirty-five United States patents in the field.

Mr. Isom was born in Mitchell, Indiana and holds degrees from Butler University, The George Washington University, and Harvard University. He is a member of Phi Kappa Phi, the Audio Engineering Society, and a Fellow of the Institute of Electrical and Electronic Engineers and the Society of Motion Picture and Television Engineers.

He has been connected with the engineering department of RCA since 1944 and Chief Engineer of RCA Records since 1966. Mr. Isom is technical advisor on sound recording matters to the U. S. National Committee of the International Electrotechnical Commission.

PROJECT NOTES / ENGINEERING BRIEFS

LONG-TERM DURABILITY OF PICKUP DIAMONDS AND RECORDS

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AND

FRANK H. HIRSCH

Zürich, Switzerland

The improvements achieved in the design of pickup cartridges during the past ten years are so far-reaching that it seemed worthwhile to check anew the durability of the diamond stylus and the record groove.

The two most significant improvements, notably, greater compliance and—much more important—reduction of dynamic mass, have made it possible to reduce the stylus force to minima that had formerly been thought impossible to achieve.

It appears that the wear of styli and record is even further reduced by the wet playback system, a procedure in which a liquid film is applied to the record surface during playback. This system is used more often in Europe than in the United States.

To clear the issue, long-term tests under controlled conditions were conducted, using both spherical and biradial diamonds, and the wear of the diamond stylus and the record groove has been compared by testing on both dry and wet playback systems.

The pickup used in the testing was a Shure M91 which is at the high end of the line of quality cartridges produced today. According to the manufacturer's specifications this pickup has a compliance of 25×10^{-6} cm/dyn and a dynamic mass of about 0.5 mg. These values allow for a stylus force between 0.75 and 1.5 grams. Consider-

Fig. 1. Modified Perpetuum-Ebner record changer used for long term tests.

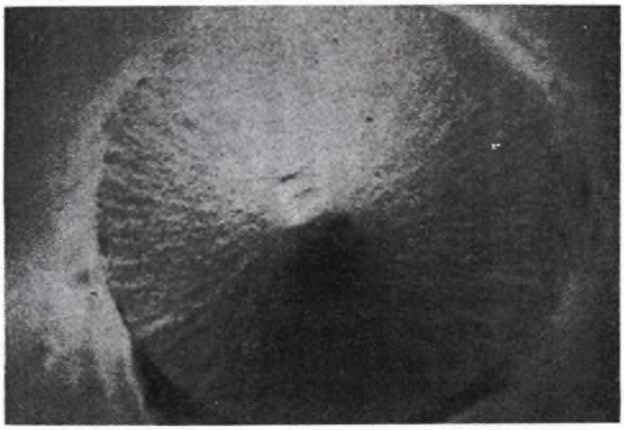


Fig. 2. Top view of spherical stylus after 1500 hours of play on wet record with stylus force of 2.5 grams. Notice splintered spot near contact points caused by one light touch to the rim of turntable.

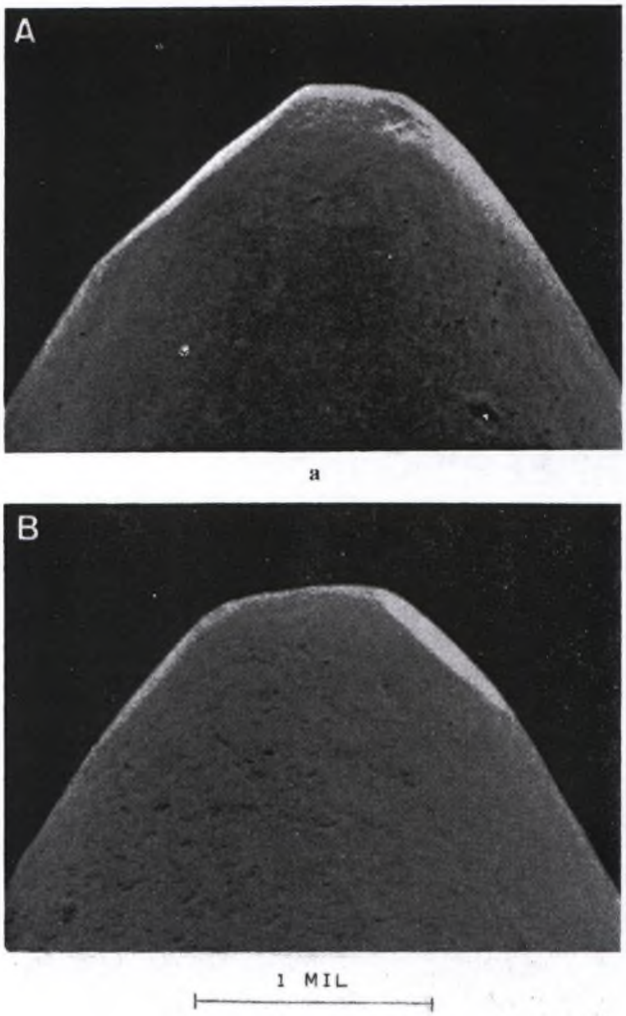


Fig. 3. Spherical styli after 1500 hours of play. Stylus force 2.5 grams. **a.** Profile of stylus used on dry record **b.** Profile of stylus used on wet record.

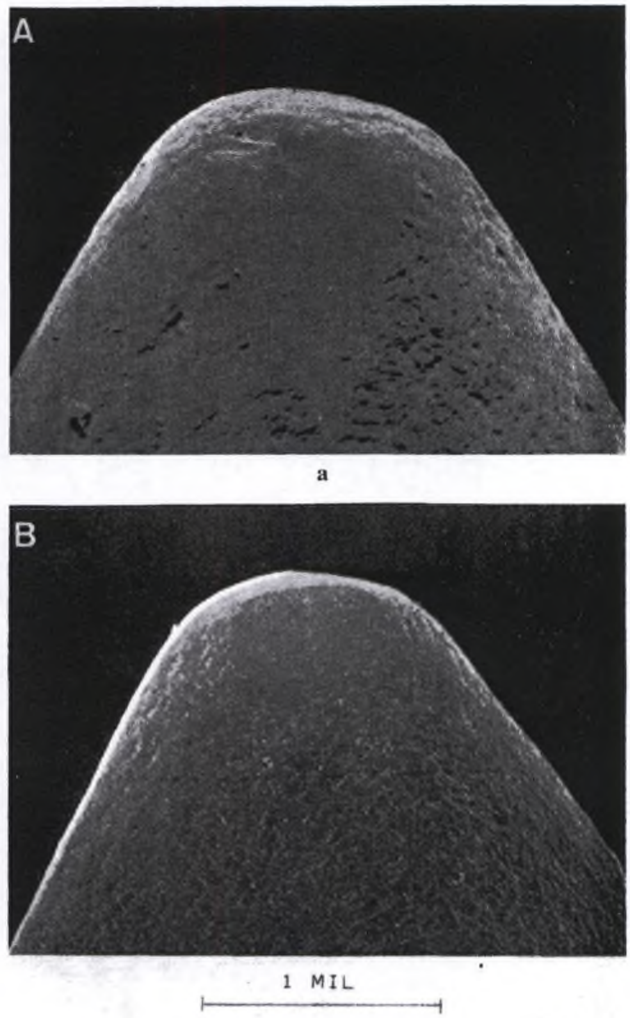


Fig. 4. Spherical styli after 1500 hours of play. Stylus force 2.5 grams. **a.** One of the contact points of stylus used on dry record. **b.** One of the contact points of stylus used on wet record.

ing the unavoidable vertical movement caused by warped records and the possible use of dynamically not well balanced tone arms, it is generally advisable to use stylus forces in the upper third of the range recommended by the manufacturer of the pickup. In order to determine whether or not an even greater stylus force would still result in an acceptable durability, the stylus force was deliberately chosen extremely high, the upper limit being dictated by the pickup compliance. The spherical stylus was tested at a setting of 2.5 grams and the biradial stylus at 2.0 grams.

The testing was done with two Perpetuum-Ebner 2020 record changers which were set up to play the same record continuously. This was accomplished by placing 20 small metal discs on the changer spindle. One of these metal discs is dropped from the spindle to the turntable each time the record under test is automatically played. The small diameter of the discs, less than that of the label, permits their use as a tally without interference with the repetitive play mode of the changer. The metal discs were numbered thus allowing for control of the number of times the records were played. If a malfunction occurred, it was possible to determine how many times the record had been played before the malfunction took place.

One of the changers used in the testing was standard, the other was set up for the wet playback system by means of a Lencoclean "L". A special device was created to guide the Lencoclean back to the starting position after each play. In Fig. 1 this device may be seen at the rear of the changer.

The liquid used to moisten the record surface was Lenco-Supertonic. Dust covers were used on both changers to prevent dust from influencing the tests. The styli in both pickups were cleaned each time after 20 records had been played, and at the same time the Lencoclean on the changer for the wet playback was refilled. The test being conducted with the standard changer was interrupted after 500, 1000 and 2000 plays to wash the record and thus keep dust accumulation at a minimum. No record cleaning was done on the record played wet.

The test records used were RCA LSC-3153 "Great Operatic Duets," the samples being taken from the same pressing. This record was selected because of its technical excellence and also because distortion—if present— becomes very audible in soprano and alto voice reproduction.

The magnitude of diamond and groove wear was judged in two ways: optically by taking pictures using a stereo-

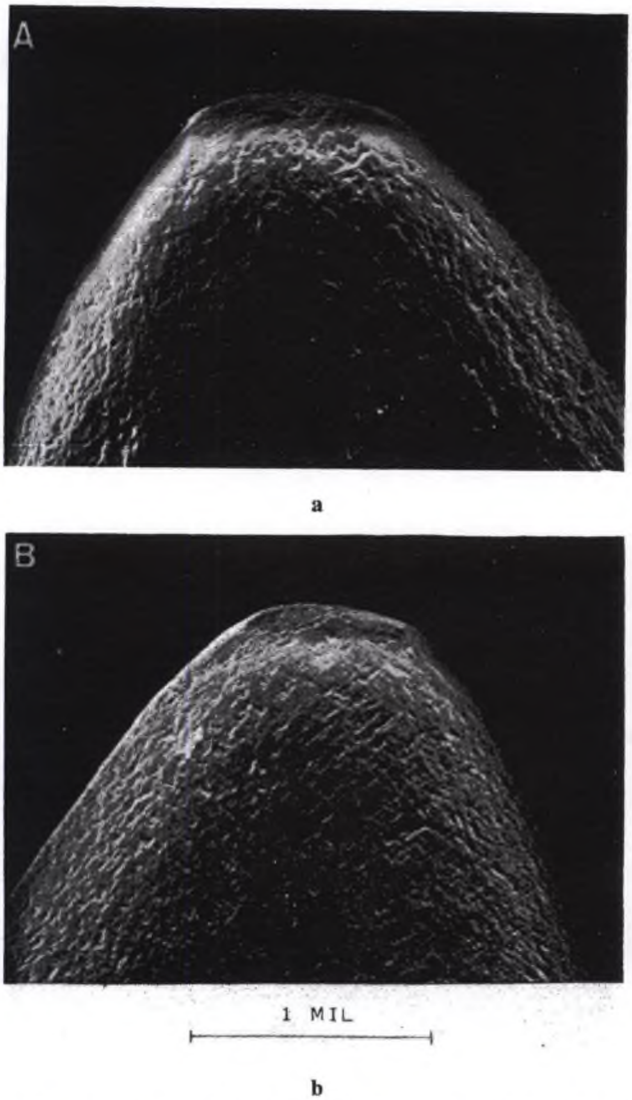


Fig. 5. Biradial styli after 1500 hours of play. Stylus force 2.0 grams. **a.** Profile of stylus used on dry record. **b.** Profile of stylus used on wet record.

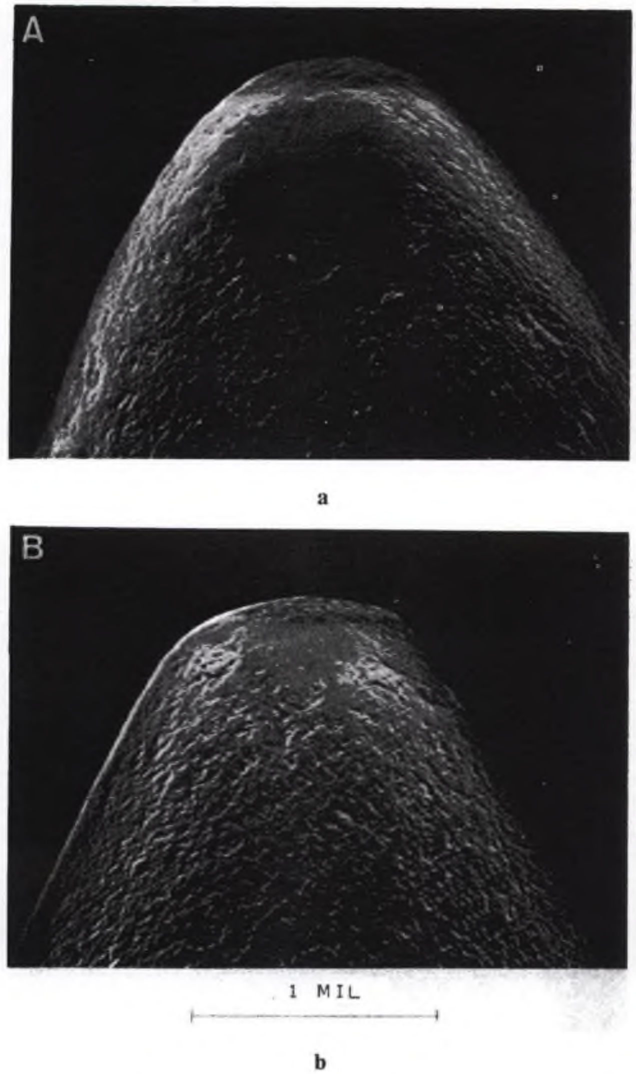


Fig. 6. Biradial styli after 1500 hours of play. Stylus force 2.0 grams. **a.** One of the contact points of stylus used on dry record. **b.** One of the contact points of stylus used on wet record.

scan microscope and acoustically by means of an *A/B* aural comparison. To implement this *A/B* comparison, tape recordings were made of the test records before the actual testing was begun. At various intervals during the testing, additional recordings were made which were spliced later and were used for comparison with the originals.

Very little degradation in audio quality was audible with the wet playback method at the end of 2000 plays. The standard dry playback method maintained a similar level of audio performance for approximately 1200 plays.

Figs. 2 through 7 show that even with an extremely high stylus force the durability of the diamond stylus in a modern high-quality pickup is very good. With stylus forces between 1.0 and 1.5 grams more than 1500 playbacks may be expected with the standard dry system and about 2500 plays with the wet system. Played with a modern high-quality pickup the record groove seems to be nearly indestructible (Fig. 8). Even after 2500 plays with extreme stylus force the record played wet still sounded good with a new stylus. On the record played dry under the same conditions a certain amount of distortion, a noticeable loss in the high-frequency range, and a lot more noise were audible.

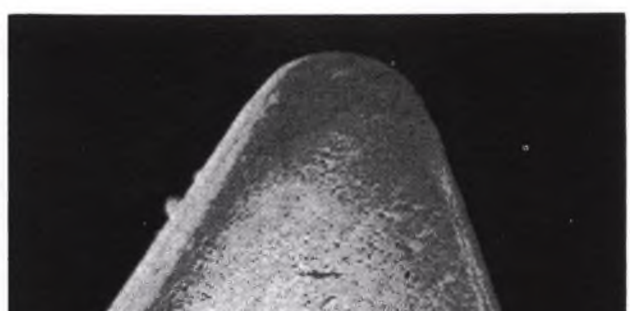
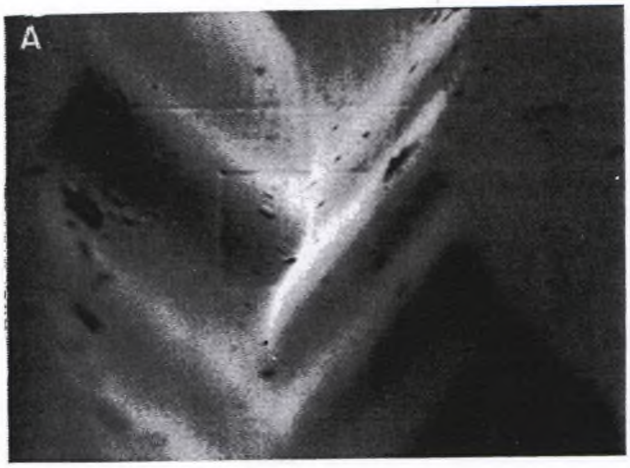


Fig. 7. One of the contact points of biradial stylus (Shure V 15-II) used in hi-fi installation of one of the authors for six years. Stylus force 1.5 grams. All records played wet.



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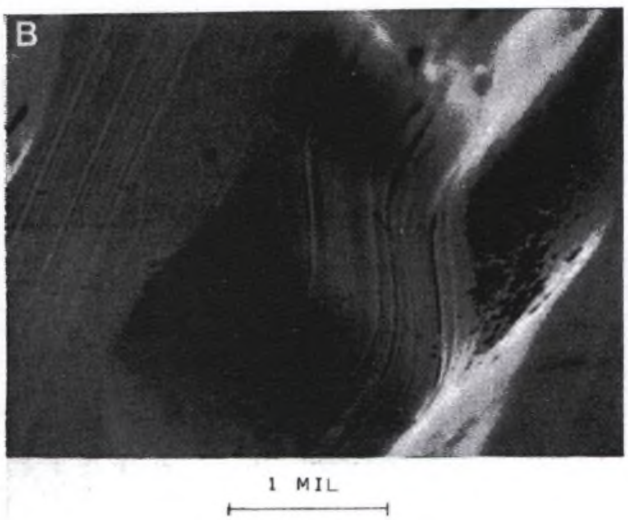

b

Fig. 8. Record grooves of test records after 1500 hours of play. **a.** Record groove played wet. **b.** Record groove played dry.

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ROLE OF POLYMER SCIENCE IN DEVELOPING MATERIALS FOR PHONOGRAPH DISCS*

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An introduction to the science of "flow and deformation" of vinyl chloride–vinyl acetate copolymers is given, along with the equipment used to study the rheological parameters. The molding of phonograph discs is related to the melt viscosity and melt elasticity. A correct balance of melt viscosity and elasticity appears to be a prerequisite for the successful molding of discs. It is shown that the molecular structure of polymers used in the phonograph record compounds has a significant effect on the melt flow and dynamic properties and, thus, the processing of the compounds. It is suggested that polymer rheology is a very useful study in the development of materials and to understand such problems as improper molding, warpage of discs, wear life of discs, and pressing cycle times.

INTRODUCTION: Rheology is the science studying the deformation and flow of materials. The development of melt flow data provides a positive means to the understanding and prediction of processing properties. In the following discussion the case of compression molding of phonograph discs (in stereo and CD-4 quad) will be examined where rheological techniques have been applied to the process and product design problems.

Recent literature [1]–[10] indicates that much effort has been spent in the analysis of resins. Though polyvinyl chloride and copolymers of vinyl chloride–vinyl acetate are extensively used, the application of melt flow data in compounding is not widely published. The compounding of vinyl compound formulations to a great extent is treated more as an "art" than a science. This is particularly true in the case of the phonograph molding industry where very little published work is known. To understand the basic problems and to reduce the inefficiency in the compounds, it is necessary to investigate rheological properties as related to process parameters.

Such relationships are

1) polymer properties,
2) compound additives and modifiers,
3) process variables (including shear and heat history).

The purpose of this note is to demonstrate the use of some of the known rheological parameters to scientifically develop compounds for disc molding processes. The results of these experimental investigations will show the flow measurement techniques used in understanding the materials and the processes used in order to develop an ideal compound.

EXPERIMENTAL

The equipment used in developing the rheological property measurements was

1) Instron capillary rheometer
2) Weissenberg rheogoniometer

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* Presented October 31, 1975, at the 52nd Convention of the Audio Engineering Society, New York.

3) mechanical spectrometer
4) Rheovibron.
These devices are shown in Figs. 1–4.


Fig. 1. Instron capillary rheometer.

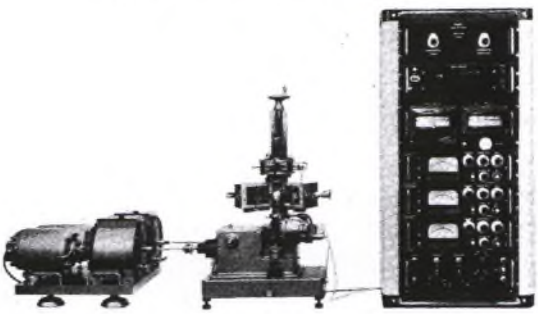
Fig. 2. Weissenberg Rheogoniometer.


Fig. 3. Mechanical spectrometer.

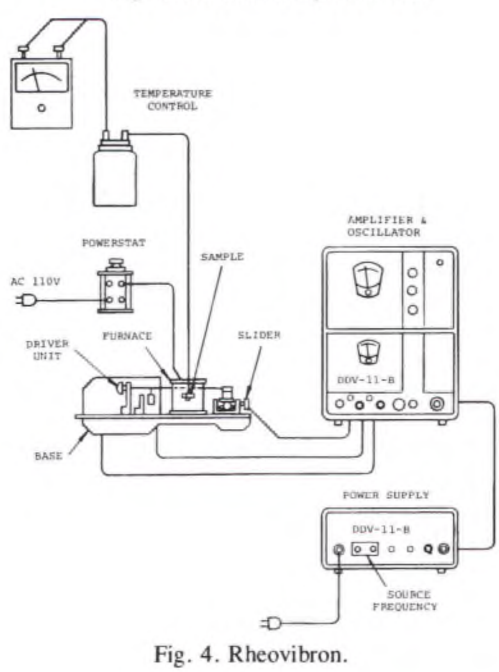

Fig. 4. Rheovibron.

The melt viscosities of the compounds were measured with the Instron and Weissenberg devices and the mechanical spectrometer over the range of temperatures which related to the nature of the compound and the fabrication processes. The Rabinowitsch and Bagley corrections were not applied. It was reported [2] that such corrections are small for polyvinyl chloride. The methods used for determining the melt viscosity and melt elasticity are described in [11]–[13].

The dynamic tests were made by using a Rheovibron. The techniques used are given in [14].

RESULTS AND DISCUSSION

Development of Compounds for Stereo and CD-4 Type Discs

At present, vinyl chloride acetate copolymers are the most widely used materials for phonograph record pressing. The function of the presence of acetate is to make the resin softer, and, therefore, acetate acts as an internal plasticizer. Different copolymer acetate levels are used by record companies according to their processing conditions. In the record industry it is common practice to use blends of two copolymers of different molecular weight and acetate or polyvinyl chloride homopolymer with acetate copolymers. It is known that such polymer combinations assist in the control of "unfill" (molding voids) though usually at some penalty to sound quality (signal-to-noise ratio). What was actually happening was not clearly understood.

The melt flow properties of a 15% acetate copolymer were measured by using a capillary rheometer at 150°C. The addition of 10% polyvinyl chloride homopolymer of K50 molecular weight in the 15% acetate copolymer increases the melt viscosity of the compound as shown in Fig. 5. The increase in melt viscosity can be explained due to the presence of a polyvinyl chloride homopolymer which is a stiffer resin than a vinyl chloride–vinyl acetate copolymer and hence increases the melt viscosity of the compound. The literature and the records industry have been claiming

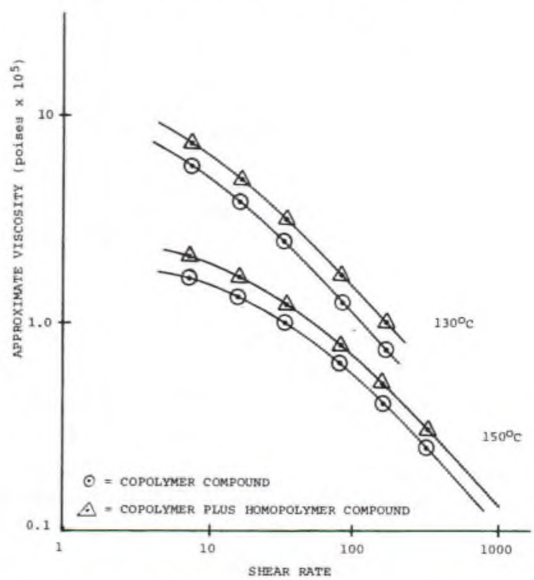

Fig. 5. Effect of the addition of 10% polyvinyl chloride homopolymer on the melt viscosity of vinyl chloride–vinyl acetate copolymer at 130 and 150°C.

that the addition of homopolymer makes the compound flow more easily, which is quite the opposite to our laboratory findings. Both the compounds containing copolymer and copolymer with 10% homopolymer were studied for melt elasticity using the Weissenburg Rheogoniometer at different temperatures. It can be seen from Fig. 6 that the presence of the homopolymer reduces the melt elasticity of the compound. Record pressings from these two compounds were made under standard molding conditions, and it was found that discs molded from the single copolymer system resulted in molding problems such as "unfills" which did sound during playing, whereas good molding of discs was observed with the compound having two resin systems (polyvinyl chloride/polyvinyl chloride–polyvinyl acetate copolymer).

Further work was carried out in order to find the best properties and requirements of a secondary resin in a compound to get best molding results. It is our finding that for best pressing and sound properties, a second resin should have a 4–6% lower acetate level and be higher in molecular weight than the primary resin. If these conditions are met, there should be a considerable difference in the melt flow properties between primary and secondary resins. Our laboratory results have proved that the higher the flow difference between the primary and secondary resins, the more effective the secondary resin is in lowering the melt elasticity of the primary resin, and thus the better the molding characteristics. Figs. 7 and 8 show the melt viscosity and elasticity of various secondary resins with a primary resin containing 15% acetate copolymer.

The rheological work to this point had adequately explained the compound properties sought for molding stereo records. This work had indicated that the successful molding of an audio disc requires a correct balance of melt viscosity and those rheological properties that are directly related to melt elasticity. The recognition of the elastic component was essential in the structuring of work to optimize a compound with respect to molding cycle time.

The introduction of the discrete four-channel record, however, required additional work which would cope with the phenomenon of wear. Wear in this case includes all those effects that would result in the progressive deterioration of the groove surface as read by a real-time audio spectrum analyzer. This includes the effects of abrasion, compressive yield, viscoelastic properties, and external effects of atmospheric debris.

An extensive viscoelastic study of resin combination systems along with modifiers and other additives yielded the result that very high melt viscosity compounds do not solve the problem of wear in this application. Figs. 9 and 10 indicate the melt viscosity versus shear rate and temperature for three specimen materials tested for quadraphonic use. It is to be noted that, in comparison with the successful material, materials on both sides of the viscosity scale do not meet wear requirements as indicated in our results.

The effect of modifiers and lubricants on the thermal stability of the compound is shown in Fig. 11. Table I illustrates the blending procedure for a CD-4 type compound. Each step was carried out in order to get consistent properties. The compound developed had similar melt flow properties as existing stereo compounds, but it is our belief that an adequate improvement was brought about by the use of certain modifiers and processing methods shown in Table I.

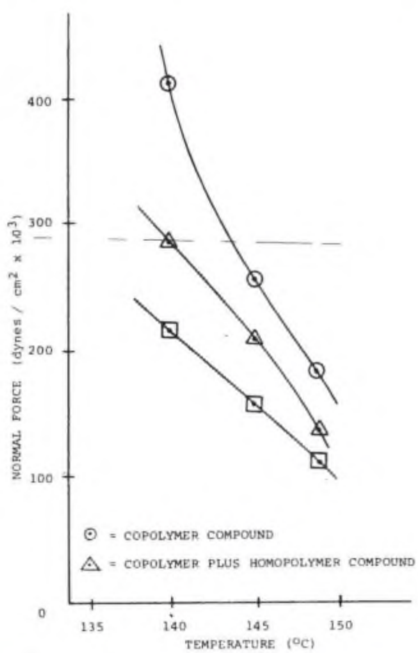

Fig. 6. Normal force at various temperatures (constant shear rate, 0.57 second^{-1}) measured on the Weissenberg Rheogoniometer.

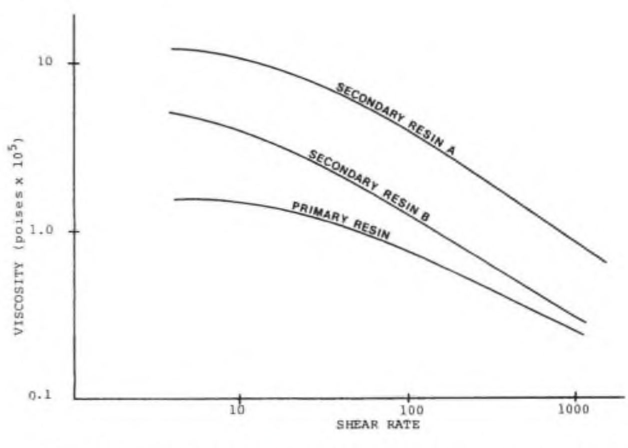

Fig. 7. Comparison of flow properties of two secondary resins *A* and *B* and primary resin at 150°C.

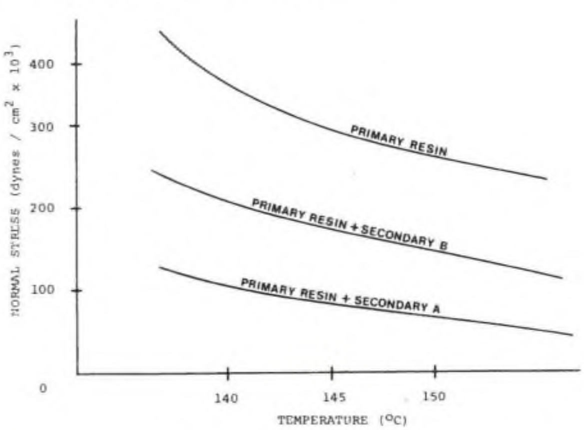

Fig. 8. Effect of type of secondary resin in reducing the melt elasticity of primary resin at 0.57 second^{-1} shear rate.

The author is proposing a rheological model as given in Fig. 12, showing the relationship between the rheological parameters and the processing conditions for compression molding of phonograph discs.

Study of Warpage Phenomenon in Phonograph Discs

The warpage phenomenon is due to the dimensional instability of the molded disc noted at any time from molding through any phase or time of storage. The warpage in the disc could be due to many factors such as

1) molding conditions,
2) orientation of the label paper,
3) dimensions of the molded disc,
4) polymers used in the compound,
5) storage and packaging conditions.

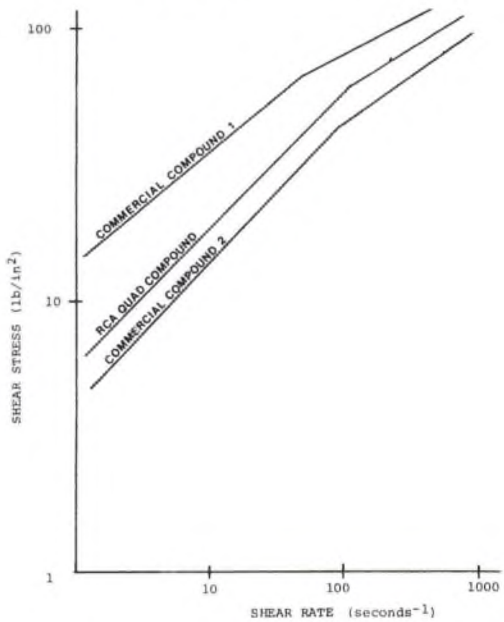

Fig. 9. Shear stress at different shear rates at 150°C for different quadraphonic compounds: commercial quadraphonic compounds 1 and 2 and RCA-developed quadraphonic compound.

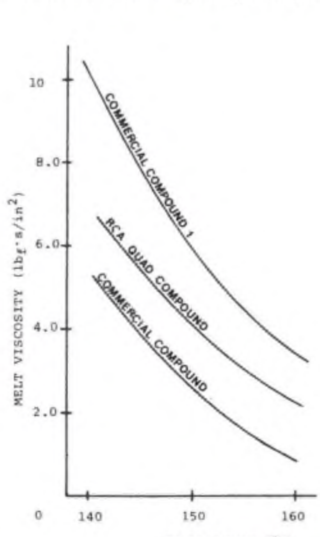

Fig. 10. Melt viscosity at various temperatures (constant shear rate 2.98 seconds^{-1}) on the Instron capillary rheometer for commercial quadraphonic compounds 1 and 2 and RCA-developed quadraphonic compound.

The properties of the resins used in the phonograph record compound contribute to the warpage phenomenon. One of the most important and indeed interesting problems of the polymer industry is the relationship of the molecular structure of the polymers and their processing and performance behavior. This work was undertaken to study the effect of the molecular structure of polyvinyl chloride–polyvinyl acetate copolymers on their viscoelastic properties in the hope to explain the warpage phenomenon due to materials. The copolymers evaluated in this study were commercially available products. Four different copolymer samples were

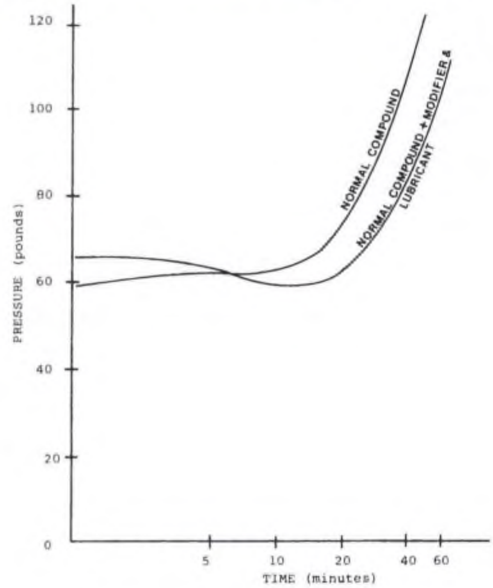

Fig. 11. Effect of modifiers and lubricants on the thermal stability of the compounds at 150°C using Instron capillary rheometer at a speed of 0.5 in/min (12 mm/min).

Table I. Blending steps for CD-4 compound.

1) Mixing of the resins
   Polymer resins (primary and secondary) and colorant are blended at high speed in an internal mixer to 110°F.
2) Addition of antistatic agent
   Mixing speed is reduced to low and antistatic agent added slowly. Mixing speed is changed to high and temperature of the mixer raised to 130°F.
3) Addition of stabilizers and modifiers
   Mixing speed is lowered and these ingredients are added. After the addition the mix termperature is raised to 150°F.
4) Addition of lubricants
   They are added and the mixing temperature is increased to 160°–200°F, depending upon the type of resin used. The mixed compound is cooled and fluxed in the extruder.

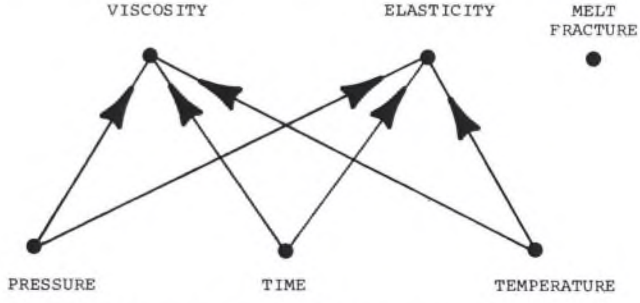

Fig. 12. Rheological model for compression molding.

evaluated with similar proportions of acetate and molecular weight.

The melt flow properties of these copolymers were measured by using the Instron capillary rheometer. All four samples have very close melt flow properties at 150°C, as seen in Fig. 13. These samples were then evaluated for dynamic properties using the Rheovibron. The samples were prepared under very tightly controlled conditions. The details of the equipment and mathematical calculations are given in [14]. The calculated complex modulus is plotted against temperature for these copolymers in Fig. 14. The results are summarized in Table II. It can be seen that the values of E^* for resins A and B are higher than for C and D. The glass transition temperature for A and B are also lower than for resins C and D. In order to relate the dynamic properties to the warpage phenomenon, four compounds were blended using A, B, C, and D resins separately, and records were pressed and stored under similar conditions. The rejects due to warpage were very high for compounds using resins A and B, whereas compounds with C and D gave very satisfactory results. At this point we suspected that the higher values of the dynamic modulus of elasticity of resins A and B could be due to chain branches produced by the presence of an acetate group in the copolymer. We have also noticed that as the percentage of acetate is increased in the copolymer, E^* is increased and T_g is decreased. The increased amounts of branches reduces the T_g of the polymer [15], which seemed to be the case in our studies (see Table II).

The next step taken was to see if we can observe any difference in melt viscosity for these copolymers due to different amounts of branches at very low shear rates. The mechanical spectrometer was employed to study the melt viscosity of these copolymers. The flow curves are shown in Fig. 15. The very low shear rates allowed us to reach into the Newtonian viscosity range. It can be seen that samples C and D gave lower Newtonian viscosities than samples A and B. It can also be seen that the Newtonian shear range for

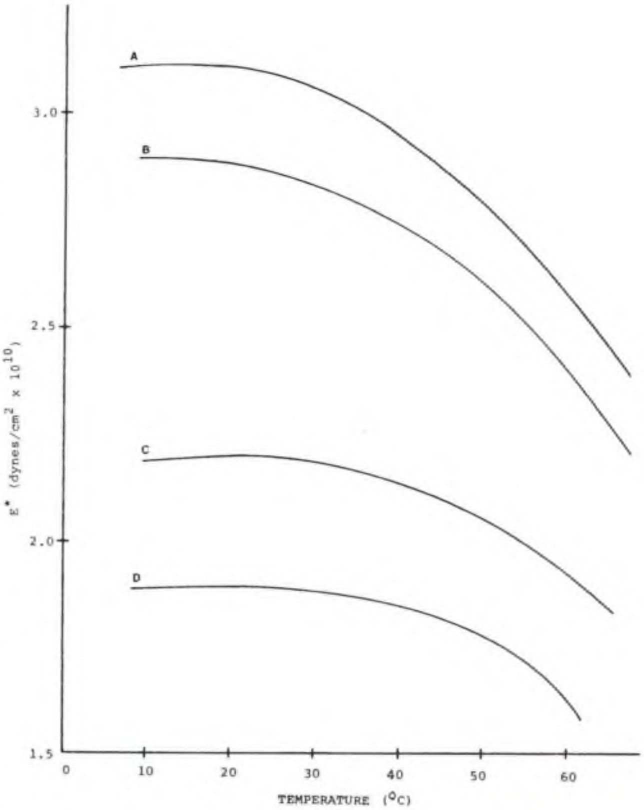

Fig. 14. Comparison of E^* at different temperatures for various copolymers using Rheovibron.

Table II.

Resin	E^* at 10°C	Glass Temperature T_g (°C)	Warpage
A	3.1×10^{10}	37	Bad
B	2.9×10^{10}	36	Bad
C	2.2×10^{10}	46.5	Very good
D	1.9×10^{10}	45	Very good

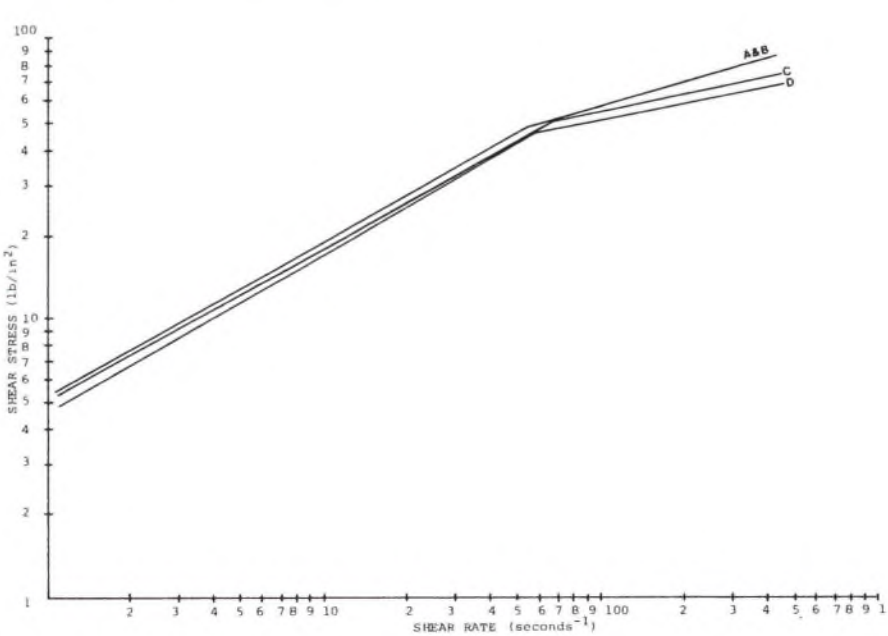

Fig. 13. Comparison of shear stress versus shear rate at 150°C for various copolymers of similar acetate levels and molecular weights.

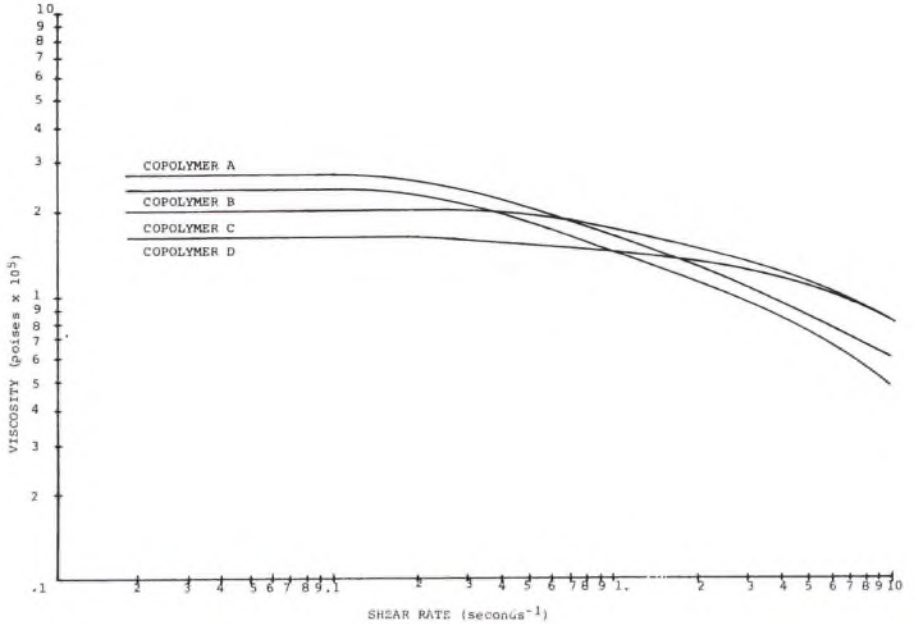

Fig. 15. Comparison of melt viscosity versus shear rate at 156°C for various copolymers using mechanical spectrometer.

C and *D* resins is greater than for *A* and *B*. These low shear viscosity data seemed to suggest that the chain branches for samples *A* and *B* are long enough to become engaged in interchain entanglements, that is, they are at least longer than the critical chain length, and the result is higher Newtonian viscosities. These data seemed to agree very well with Long *et al.* [16] and Kraus *et al.* [17], [18], but other workers such as Graessley [19] have shown that chain branches reduce the Newtonian viscosity over more linear structure.

SUMMARY

We have attempted to show that small changes in the polymer structure can have pronounced effects on the flow and dynamic properties of the compounds. These will in turn affect the processing and end product properties.

We have seen how modification of the flow and viscoelastic properties can be brought about by blending different resins (homopolymer or copolymer with different acetate and molecular weight), and we have seen how such modification can mean a difference between failure and success in achieving an acceptable molding of phonograph discs.

We would like to stress that the above-mentioned studies have explained in detail some of the problems this industry is encountering, and further work will improve the quality of the phonograph discs.

ACKNOWLEDGMENT

The author wishes to express his special thanks to J. C. Ruda for help in preparing this paper.

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Record Warps and System Playback Performance*

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There exists a great variation in the ability of current high-quality tone-arm–phonograph-cartridge combinations to successfully track warped records. In this paper the optimum design parameters of the tone-arm system are developed which are necessary to cope with record warps without sacrificing other important performance aspects. An extensive survey of warps found on commercially available phonograph records and an analysis of the dynamics of the playback system are included in this study. The analysis extends the concept of trackability into the subaudible, or warp, region. Choosing the optimum tone-arm and phonograph-cartridge parameters yields improved playback performance for the total range of signals found on phonograph records in the presence of warps.

INTRODUCTION: Trackability may be defined as "the ability of the stylus to remain in contact with the record groove at a specified tracking force across the frequency spectrum found on records." This definition has always been understood to include the frequency range of modulation recorded in the record groove. However, the phrase "frequency spectrum found on records" also includes the subaudible signals found on the record surface. It will be shown in this paper that above approximately 50 Hz, the trackability of the stylus is controlled primarily by the phonograph-cartridge design. Below 50 Hz it is the playback system composed of the tone-arm and cartridge design which determines trackability. The trackability in this region is largely affected by the frequency of the tone-arm resonance.

Tone-arm resonance is the resonance determined by the effective mass of the tone arm (calculated as that dynamically appearing at the stylus tip) and the dynamic compliance of the phonograph cartridge. An optimum design should position this resonance in a frequency region where it will least likely be excited by vibrations within the mechanical system. For many years, this meant simply positioning this resonance frequency below the audio spectrum, that is, the region below 20 Hz. With present top-quality arm and cartridge combinations, the range of possible resonance frequencies is well below the audible region. However, these resonances may be excited by other types of system vibrations. This paper concerns itself with one such system input, namely, record warpage and its effect on the playback performance of the system.

Record warpage problems occur whenever the tone arm begins to bounce or sway with respect to the record surface, thus rapidly changing the distance between the cartridge body and the record groove. This cartridge–record relative movement results in an oscillating or varying tracking force which can cause mistracking of the audio information if the tracking force becomes momentarily too low. In severe cases, groove jumping and/or bottoming of the cartridge on the record surface can result. In addition, the stylus tip oscillates in a forward and backward direction over the recorded material in the groove. This stylus action introduces undesirable frequency modulation of the recorded signals, commonly referred to as wow. The warp-induced stylus movement also creates a subaudible electrical signal and when amplified can result in distortion from extreme woofer movement or amplifier overload.

RECORD WARP INVESTIGATION

Description of the Test Setup

Before the effects of warp on playback performance could be investigated, it was necessary to establish the pertinent characteristics of the warps found on commer-

* Presented September 10, 1973, at the 46th Convention of the Audio Engineering Society, New York.

cially available records. To accomplish this, a method of measuring the record surface contours was developed. The experimental arrangement is shown in Fig. 1. An amplitude-sensitive transducer was rigidly mounted over the record surface. The transducer was placed approximately ½ inch (12.7 mm) from the edge of each record to be tested and adjusted vertically to accommodate the maximum deflection of the record. The pickup was terminated directly into an amplifier which was connected to the vertical input of the chart recorder.

The system was calibrated using known step input amplitudes and recording them with the chart recorder. Deflections up to 0.060 inch (1.5 mm) could be measured with an accuracy of ±0.001 inch (0.025 mm), which was considered adequate for the applications required later in this paper. The tendency of the tracking force of the transducer to distort the record contour was judged negligible, and no correction was necessary.

To assure accurate record contour measurements, it was important to eliminate the contour irregularities of the turntable. To assure this, the upper aluminum platter of the particular turntable was removed and the lower cast-iron platter was used to support the records under test. The slight residual irregularities of the platter were recorded and accounted for with a calibration curve. This curve indicated that the turntable platter contained a single, low-amplitude variation of less than 0.001 inch (0.025 mm) peak amplitude. The platter variation was not eliminated from the data, since it was found insignificant compared to the measured values for record warps. Also, the measured wavelength results in a playback frequency of approximately ½ Hz, which was below the range of major interest of this investigation.

Measurement Procedures

Sixty-seven records were measured. It should be noted that this was not a collection of badly warped records, but rather a randomly chosen collection from the Phono Department Lab files. Included were records from approximately 30 different manufacturers (foreign and domestic), which might be assumed to represent a typical record collection. Several records were included in the sample lot because they were known to contain "problem-causing" warps; however, no severely warped records were included in above sample. None were included which would not be acceptable commercially. (All records measured met the 1964 N.A.B. standards for record warp.)[1]

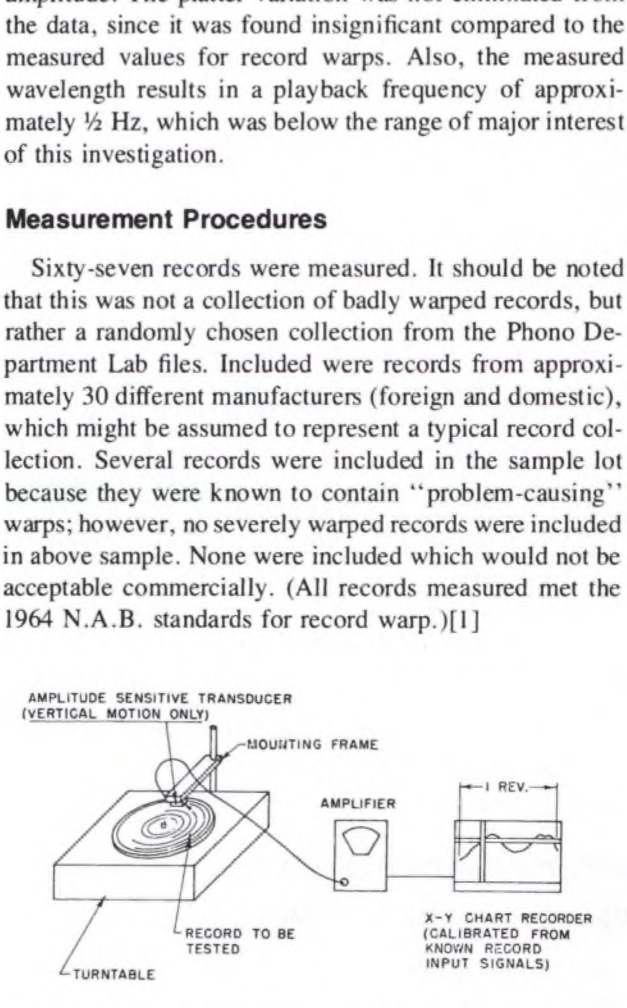

Fig. 1. Experimental setup for record contour measurement.

Both sides of some records were measured. It was observed that a low-amplitude, long-wavelength variation usually existed between the two sides of a record. This was primarily due to the fact that the record rests on different contact points when measuring the two sides and, thus, deforms differently in supporting its own weight. It was possible to detect thickness variations as well when measuring both sides of a record. Due to the deformation mentioned, only short wavelengths could be confirmed as being specifically thickness variations. No attempt was made to separate these types of irregularities from those considered record warps. The intent of this investigation was to obtain the spectrum of low-frequency surface irregularities tracked by the cartridge regardless of type or definition.

Measured record contours are shown in Fig. 2. As can be seen from the contours, the records were typically not flat, but each contained many peaks and dips of varying amplitude and wavelength. There were many identifiable kinds of warps, such as "dish," "saddle," and "pinch," described in past literature[2]. However, no attempt was made to classify them in this manner. It was decided to analyze each record contour in terms of the playback frequency and amplitude of each wave or set of waves occurring. This information was used directly in the theoretical analysis described later in this paper.

A procedure of examining the measured record contours was developed. The location of each measurement was determined by visual observation. (See examples in Fig. 2**a** and **b**.) Measurements were made in cases where at least one full cycle of a warp could be clearly observed. The amplitude and wavelength information was measured from the peaks in each set of waves. The amplitudes were measured from peak to peak and then divided in half. If the peaks varied in height, the average amplitude was chosen. If two waveforms were discernibly superimposed upon one another, each wavelength and amplitude was measured independently. The frequency information was calculated from the wavelength assuming a 33⅓ r/min playback speed.

A total of 210 warps were identified, although many more could have been measured. Only the higher amplitude signals were chosen for the longer wavelengths, that is, wavelengths greater than ¼ the record circumference. Low-amplitude, long-wavelength signals rarely cause

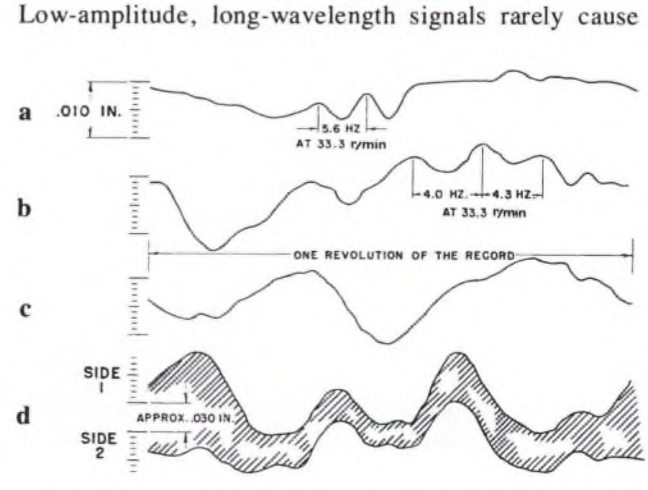

Fig. 2. Examples of actual surface contours found on commercially available records.

playback problems. Although these were the most frequently occurring type record warp, it will be shown that the frequency and amplitude range of the signals omitted were not of interest for this investigation.

The measurements of warp were plotted as points on a graph of amplitude versus playback frequency, assuming a 33⅓ r/min turntable speed (Fig. 3). Each point on this graph represents an identifiable warp condition on a record. The solid line represents the boundary that includes the maximum warp amplitudes found on these records, and therefore represents the maximum that may be expected at each frequency from any other collection of commercially available records.

The shape of the maximum-warp amplitude curve seems reasonable. The amplitude of the 1-Hz frequency would probably be the greatest, since it corresponds to the "saddle" type warp, that is, the record is bowed along its diameter. It is reasonable to expect that the longest wavelength warps also have the greatest amplitudes. The maximum amplitude is shown to decrease with increasing frequency, indicating a tendency for the amplitude to vary proportionately with wavelength.

From the maximum-amplitude curve, information as to the velocity and acceleration distributions can be derived. From the data obtained, most maximum warp velocities vary from 0.2 to 0.6 cm/s; correspondingly, most maximum warp accelerations vary from 10 to 16 cm/s^2.

A measure of the relative distribution of warp frequencies is shown in the upper left corner of Fig. 4. In general, this illustration shows that the most frequently occurring warps are the lower frequency type. For example, the graph indicates that of the measured warps, three times as many signals were observed at 4 Hz than were observed at 10 Hz. It was determined that approximately 70% of all the warps measured were below 5 Hz, and approximately 95% of all the warps measured were below 8 Hz.

Since there were a large number of warps on each record, an estimate of the percentage of records which contain higher frequency warps was also obtained. It was found that all records contained measurable signals in the range from 1 to 3 Hz. Approximately one of five records contained measurable signals between 5 and 6 Hz, and only one of twenty records had measureble signals from 9 to 10 Hz. Based on either of the statistical presentations, it can be observed that the quantity of warp signals which the system must properly "track" increases sharply below approximately 7 Hz.

To obtain the entire spectrum of low-frequency signals on records, it is necessary to include the range of intended signals recorded in the groove. This is approximated by assuming a 0.002-inch (0.05 mm) maximum cutting amplitude limit [3], a practical limitation imposed by the recording engineer to assure an adequate playing time. The composite curve of groove modulation and record warp is shown in Fig. 4 and includes the entire range of low-frequency signals found on records which the tone-arm system must "track" successfully. The warp envelope in Fig. 4 is the same as that shown in Fig. 2, but is represented on a velocity scale.

ANALYSIS OF THE SYSTEM DYNAMICS

System Representation and Tone-Arm Response Characteristics

The second step in this study was to analytically evaluate the dynamic performance of the tone-arm system. To accomplish this, the system was described in terms of the basic elements: the effective mass of the tone arm plus cartridge, the compliance of the cartridge, and the damping in the system. The general playback system may be represented as shown in Fig. 5. By summing the forces on the mass, the equation of motion for the tone arm can be described by the following differential equation:

$$M \frac{d^2}{dt^2} X_m + R \frac{d}{dt}(X_m - X_{in}) + \frac{1}{C}(x_0 + X_m - X_{in}) + P_m = 0 \quad (1)$$

where

M = effective mass of the tone arm referred to the stylus tip (includes both the arm and cartridge mass)

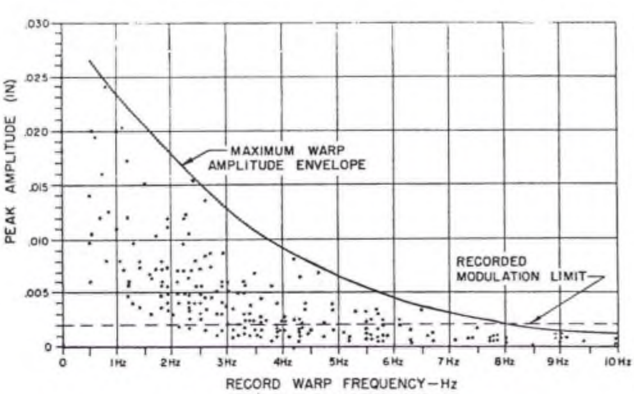

Fig. 3. Maximum warp envelope. Measurements from a random selection of 67 records, 210 measurements.

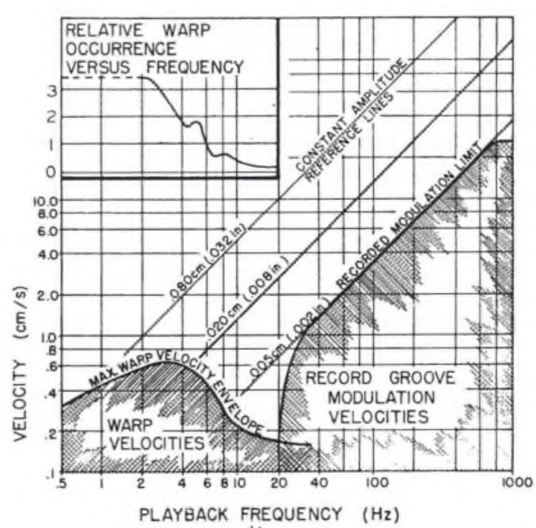

Fig. 4. Extended spectrum of low-frequency content on commercial records.

- C = dynamic compliance of the stylus measured near the resonance frequency of the stylus and tone arm (the dynamic compliance typically decreases slowly as the frequency increases; however, for the theoretical analysis in this paper, it will be assumed constant over the resonance frequency.
- R = damping present in the stylus assembly (frequency-dependent damping effects will not be considered in this paper)
- P_m = preset or static tracking force supplied by the tone arm
- P_s = instantaneous reaction force exerted by the record on the stylus tip
- X_m = mass displacement function
- X_{in} = input displacement function
- x_0 = static deflection due to the static tracking force (constant for any particular system once the tracking force has been set).

Note: The effect of the tracking force P_m is to cause the system to have a static or reference deflection x_0 about which the dynamic deflections take place. It may be neglected if it is assumed that the system deflection never approaches the undeflected position. This assumption will be checked later when the system trackability is examined. The simplified equation of motion becomes

$$M \frac{d^2}{dt^2} X_m + R \frac{d}{dt}(X_m - X_{in}) + \frac{1}{C}(X_m - X_{in}) = 0. \quad (2)$$

The transmissibility of any system is a function representing the gain of that system. It describes the ratio of the output deflection to the input deflection as a function of frequency, or

$$T = X_m/X_{in}. \quad (3)$$

This system will be analyzed assuming steady-state inputs of the form

$$X_{in} = x_{in}(\omega)\, e^{i\omega t}. \quad (4)$$

With this assumption, the resulting motion of the tone-arm mass can be described as

$$X_m = x_m(\omega)\, e^{i(\omega t + \theta)} \quad (5)$$

where

- $x_m(\omega)$ = amplitude of the mass motion
- $\phi(\omega)$ = phase of the mass motion with respect to the input signal.

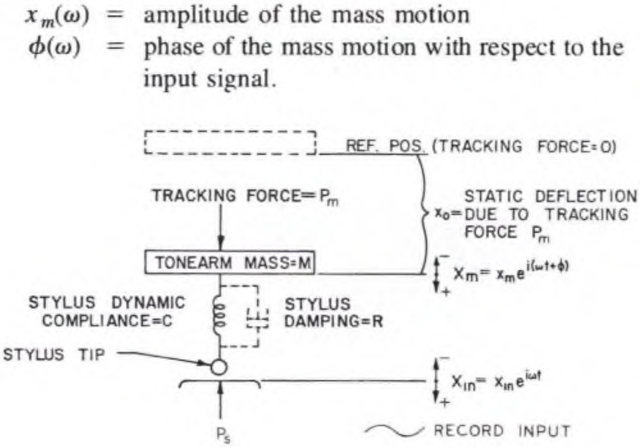

Fig. 5. Low-frequency playback system model.

Then the transmissibility of the system described with respect to the tone-arm motion can be shown to be

$$T = \{(1 + R^2C^2\omega^2)/[(1 - MC\omega^2)^2 + R^2C^2\omega^2]\}^{\frac{1}{2}}. \quad (6)$$

The maximum transmissibility with reference to angular frequency ω can be obtained by differentiating and checking for maximum. When this is done, it is found that the maximum T occurs at a frequency

$$\omega_d^2 = (1/MC)/[\tfrac{1}{2} + (\tfrac{1}{4} + R^2C/2M)^{\frac{1}{2}}]. \quad (7)$$

This is referred to as the damped natural resonance frequency for the system of Fig. 5. In vibration analysis, the term $(CR^2/M)^{\frac{1}{2}}$ is commonly referred to as the damping factor and given the symbol δ.

Using this notation, the last term in Eq. (7) becomes $\delta^2/2$. With the small damping factors found in most cartridge tone-arm systems (typically $0.2 < \delta < 0.4$) this term in Eq. (7) may be neglected. Then the frequency of maximum transmissibility becomes the undamped natural resonance frequency, or

$$\omega_r = (1/MC)^{\frac{1}{2}}. \quad (8)$$

Substituting this into Eq. (6) yields a function representing the maximum transmissibility in terms of the basic system parameters.

$$T_{max} = (1 + M/CR^2)^{\frac{1}{2}}. \quad (9)$$

In terms of the damping factor δ, Eq. (9) becomes

$$T_{max} = (1 + 1/\delta^2)^{\frac{1}{2}} \quad (10)$$

which describes the usual form for the maximum transmissibility at resonance for the system modeled in Fig. 5.

In general, this function shows that increasing the mass not only lowers the resonance frequency, but also increases the height of the resonance peak. This is the result which occurs when a phonograph cartridge is installed in a tone arm of increased effective mass. The function also indicates that increasing the compliance decreases the resonance frequency and decreases the height of the resonance peak. This generally does not occur in practice, since an increase in stylus compliance is accompanied by a reduction of the stylus damping. This phenomenon will be discussed later.

Stylus Deflection Function and Low-Frequency Trackability

The analysis up to this section has been with respect to the motion of the tone arm; however, the magnitude of the stylus deflection is the most significant measurement of the system performance in the low-frequency region. The stylus deflection, defined as $S(\omega)$, is a measure of the clearance between record and cartridge. This clearance relates to the instantaneous reaction force exerted by the record onto the stylus and, thus, is related to the system trackability.

To find the stylus deflection function $S(\omega)$ for low-frequency inputs, it is necessary to subtract the input signal X_{in} from the tone-arm mass motion X_m already derived.

We obtain

$$S(\omega) = X_m(\omega) - X_{in}(\omega) \quad (11)$$
$$= X_{in}(X_m/X_{in} - 1).$$

Substituting Eq. (3), we obtain

$$S(\omega) = X_{in}(T - 1). \quad (12)$$

Stylus motion is a means of defining the trackability in the low-frequency region since the amount of stylus deflection is a measure of the force existing between the stylus and record.

For the low-frequency inputs below approximately 400 Hz, the total amplitude available for stylus motion is determined by the preset tracking force and the stylus compliance, or

$$x_0 = P_m C. \quad (13)$$

In the extreme case, when the stylus deflects to $x_0 = 0$, the tracking force goes to zero, and the stylus loses contact with the record. Therefore, for proper playback with respect to the low-frequency signals, the stylus deflection function $S(\omega)$ can never exceed $|x_0|$ for the stylus to remain in contact with the groove, that is,

$$S(\omega) < |x_0|. \quad (14)$$

Trackability in frequency regions above 400 Hz has been discussed in previous papers [3], [4] and will only be indirectly covered in this paper. Eq. (12) and (13) can be substituted into the above equation, or

$$X_{in}(T - 1) < P_m C \quad (15)$$

$$X_{in} < P_m C / (T - 1). \quad (16)$$

Thus it is possible to solve for the range of record inputs which will cause a stylus deflection large enough to have the stylus leave contact with the record groove, that is,

$$X_{in} = P_m C / (T - 1). \quad (17)$$

This function represents the locus of maximum acceptable single-frequency inputs which would allow the system to remain in contact with the record, or the trackability function of the playback system in the low-frequency region. Of course, it is necessary for the playback system to have enough tracking force in reserve to ensure the tracking of all other signals on the record. Therefore, the allowable range of record inputs must be less than X_{in}, defined by the relation in Eq. (16).

Fig. 6 illustrates the general form of the trackability curve for the playback system near resonance. It has been drawn on a dB velocity scale to be consistent with the usual trackability scale.

Three distinct regions can be identified on the low-frequency trackability curve. The dip in the curve indicates the position of the tone-arm cartridge resonance. The resonance amplification of the input signal causes the sharp loss of trackability. This resonance effect is significant approximately an octave above and below the resonance frequency of the playback system.

The frequencies below the resonance region represent "the mass limited" region. Here the limitation is primarily the inertia of the tone-arm mass. With respect to the system trackability, mistracking will occur when the input signal acceleration, multiplied by the inertia, approaches a force F_a equal in magnitude to the tracking force. It is important to note that even if the stylus does not leave the groove, the presence of acceleration in this region momentarily causes a loss in the stylus force and, thus, lowers the trackability of the cartridge for all recorded signals for that instant. The curve approaches the asymptote defined by the equation

$$F_a = MA$$

where

A = record input signal acceleration

F_a = force of magnitude $|P_m|$.

The input signal acceleration equals the signal velocity V times the angular frequency ω. Then

$$F_a = MV\omega \quad (18)$$

or

$$V = F_a / M\omega. \quad (19)$$

Frequencies above the resonance region represent the "compliance limited" region. Here the tone arm does not move appreciably, and the groove motion is absorbed entirely by the stylus movement. This portion of the curve approaches the standard trackability curve defined for signals above 400 Hz. The ability to remain in contact with the groove is dependent on the force in the spring deflection, F_s, compared to the tracking force. As previously stated, this is defined by

$$F_s = x / C \quad (20)$$

and since

$$x = V / \omega$$

$$V = CF_s\omega. \quad (21)$$

The intersection of the two asymptote lines occurs at $\omega_r = (1/MC)^{\frac{1}{2}}$, the natural frequency of the system.

The stylus motion $S(\omega)$ [Eq. (11)] represents the actual stylus movement. When related to the electrical output of the cartridge, the function represents the response curve which would be obtained by recording the signal amplitude as a function of frequency for constant-amplitude inputs. Fig. 7 shows the theoretical representation of $S(\omega)$ as a function of frequency. Appendix I of this paper lists various records and techniques available to obtain tone-arm cartridge experimental response curves.

As Fig. 7 indicates, there exists a wide variation in the

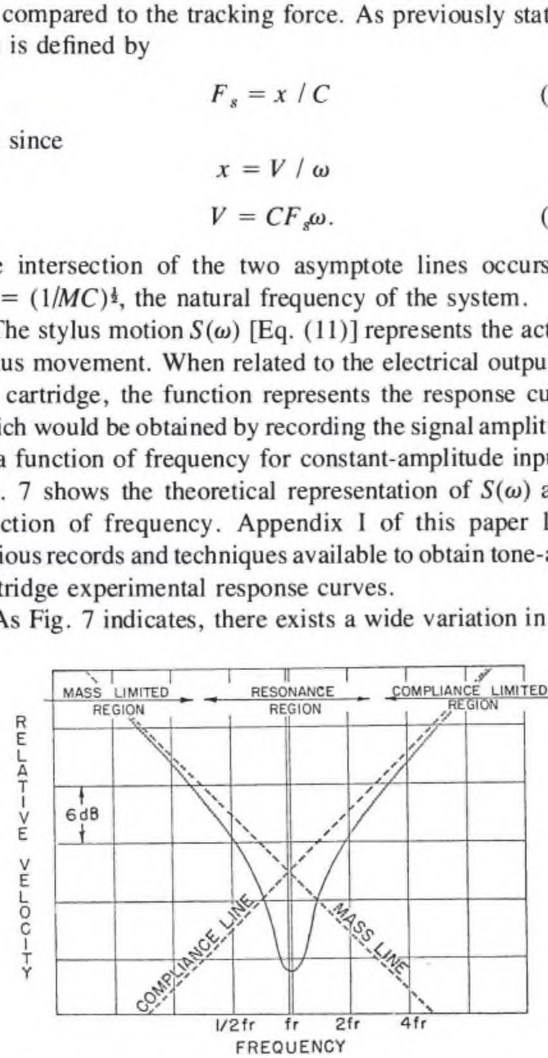

Fig. 6. Typical low-frequency trackability curve.

sensitivity of a particular system to record inputs. In the area of resonance, the input signal may be magnified several times due to the amplified tone-arm motion relative to the record. This is the region where signal "wow" and the chance for mistracking are most acute. In the region below resonance, the stylus movement is actually attenuated, due to the movement of the tone arm in phase with the input. Thus it is possible for two playback systems which are similar in other respects to vary by more than ten times in their relative sensitivity to particular subaudible inputs.

The final stage in this paper is to interpret the trackability curve just developed with respect to the input signal distribution derived earlier in this paper.

SYSTEM PARAMETERS AND THE RELATIONSHIP TO THE WARP PROBLEM

The trackability curve may be shown relative to the total signal distribution curve obtained in the warp investigation (Fig. 8). The vertical clearance between the trackability curve and the signal distribution curve at any frequency is an indication of the useful tracking force available to track the other modulations in the groove. The preferred placement of the trackability curve would be one which provides sufficiently large vertical clearance over all the frequencies shown. This preferred arrangement occurs when the dip in the trackability curve matches the dip in the signal distribution.

The degree of vertical clearance cannot be interpreted as an absolute performance criterion. However, for a comparison of two systems, the amount of increased vertical clearance of one system over another is a useful evaluation technique.

Fig. 8 illustrates a well-matched system for 1-gram tracking force using currently available equipment. The low-frequency trackability curve represents a system with a tone-arm mass of 13 grams (including the cartridge) and a dynamic compliance of 20×10^{-6} cm/dyn. The resonance frequency of the system is near 10 Hz, and the trackability values have ample margin over the entire signal envelope. For example, the maximum recorded signal level expected at 50 Hz is 1.5 cm/s assuming no other signals are present. This signal level would require 30% of the available tracking force, and the clearance between the curves indicates that 70% of the tracking force would be available to track other signals. Similarly, a maximum warp signal at 6 Hz would require an additional 30% of the tracking force. If both the warp and the 50 Hz signal are present, only 40% of the tracking force will be available to track other modulations in the groove.

The examples in Fig. 9 illustrate the result of compromises from this preferred curve. Each example shows the effect of a parameter change from the original system. The total record signal distribution and the system of Fig. 8 are presented for reference.

Fig. 9**a** illustrates a compliance change from the original. The dynamic compliance has been halved from 20×10^{-6} cm/dyn to 10×10^{-6} cm/dyn. This system has minimum reserve over the audible frequency content in the record groove and, thus, is unsuitable at 1-gram tracking force. If a loss in dynamic compliance at increasing frequency is taken into account, the actual performance would show less trackability reserve than the example, particularly at the frequencies above a few hundred hertz. The problem of peaked low-frequency response and acoustical feedback may also come into effect with resonance frequencies near 20 Hz.

Note: It has been observed that a change in stylus compliance is typically accompanied by an inversely proportional change in stylus damping. This implies a slight decrease in stylus resistance R as the compliance increases. For the general discussion in this paper, the damping factor δ was kept constant as a function of compliance.

Fig. 9**b** represents the system as in Fig. 8, except the tonearm mass has been doubled to 26 grams. This change slides the trackability limits closer and into the warp region. The reduced vertical clearance to the warp envelope curve indicates that the system will operate on many warped records at minimum tracking force, and the intersection region indicates that on some records the potential exists to lose the entire tracking force available.

It is interesting to note that as a cartridge is placed in heavier tone arms, not only does the resonance frequency decrease, but the damping factor δ is reduced, and thus the trackability at resonance is further reduced.

A survey was made on the effective arm mass of various low tracking force systems. (Appendix II details the mea-

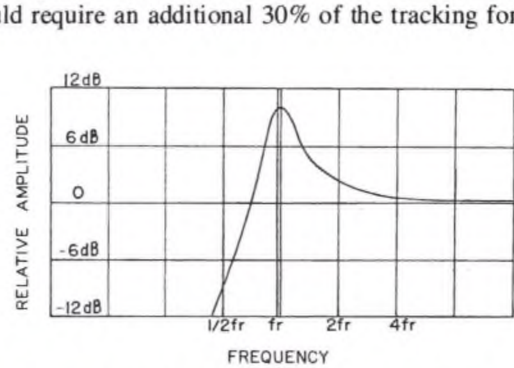

Fig. 7. Cartridge and tone-arm response characteristics. Stylus movement $= X_m - X_{in}$.

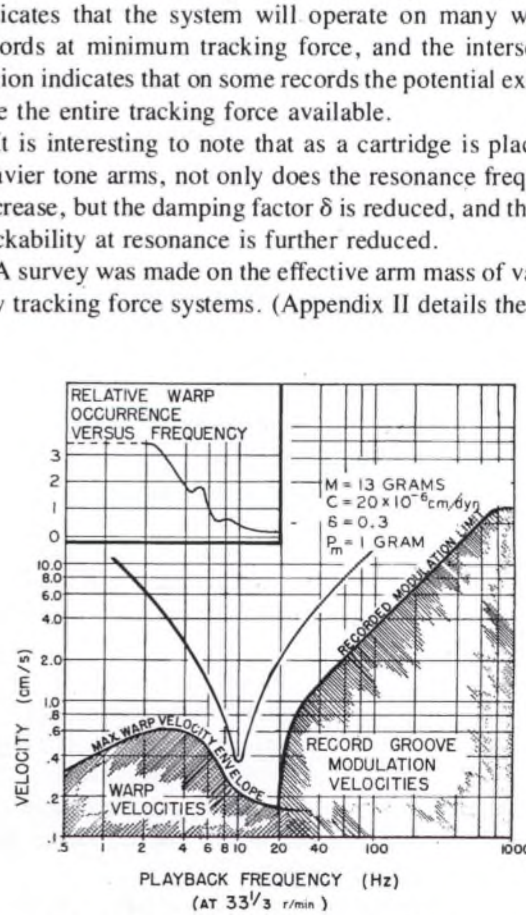

Fig. 8. Trackability curve relative to extended low-frequency signal spectrum.

surement techniques available to determine effective tone-arm mass.) The survey included manual tone arms and automatic changer arms which are otherwise claimed to be capable of tracking successfully at 1-gram force because of low friction, skating compensation, and other favorable qualities. The effective arm mass, including the cartridge, was found to range from a minimum of 13 grams to over 30 grams. The change in effective mass shown in Figure 9**b** nearly covers this range.

Fig. 9**c** illustrates a reduction in tracking force from 1 gram to ½ gram of the original system (Fig. 8). Here the entire trackability curve does not have sufficient clearance and clearly indicates difficulty at the resonance. This indicates a degree of difficulty in playing warped records at ½ gram tracking force, even with the excellent mass–compliance combination of Fig. 8.

The dynamic compliance of available phonograph cartridges specified for 1-gram tracking force varies from 15×10^{-6} cm/dyn to 60×10^{-6} cm/dyn. Fig. 9**d** illustrates the case where the compliance has been doubled to 40×10^{-6} cm/dyn. This has slightly reduced the trackability margin, and it has moved the resonance dip closer to the warp region. In the warp region the degree of difficulty any system will have must also be measured by the statistical distributions described earlier. Thus a system with a low-frequency trackability curve near or intersecting the warp envelope between 4 Hz and 5 Hz will have greater difficulties than a system with a trackability curve intersecting the warp envelope between 9 and 10 Hz.

However, the above example does not represent the extreme case which can occur. Fig. 9**e** represents the combination of the high dynamic compliance (Fig. 9**d**) and the high effective arm mass (Fig. 9**b**). Also, since the motive for high dynamic compliance is to provide low-frequency trackability at reduced tracking force, this curve was plotted at ½-gram tracking force. This system will exhibit severe playback difficulties on a large percentage of warped records.

Based on the range of effective mass and dynamic compliance values, the resonance frequency of low tracking force systems existing today can vary from 4 to 11 Hz. This range significantly overlaps the warp signal region. To minimize playback problems, a lower limit on the resonance frequency should be established at 7 Hz. Thus many cartridge and tone-arm combinations will not meet this limit. The examples in Fig. 9 have indicated the importance of matching the mass and compliance of the system. In addition, these examples have shown the need for keeping the tracking force compatible with the mass and compliance values.

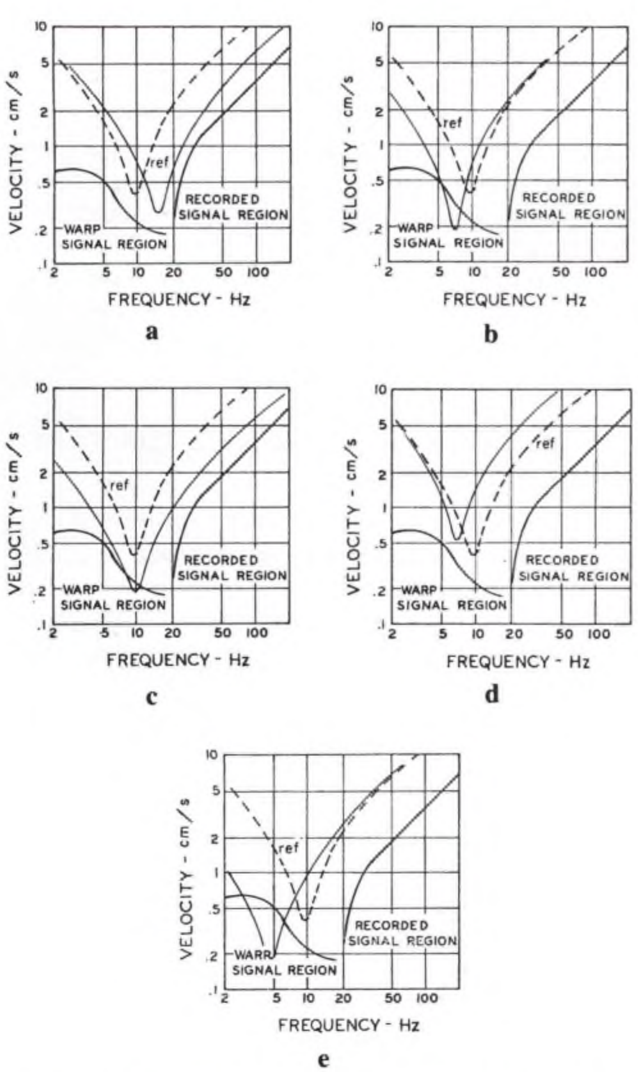

Fig. 9. Compromises from preferred curve of Fig. 8. **a.** $C = \frac{1}{2} C_{ref}$. **b.** $M = 2M_{ref}$. **c.** $P_m = \frac{1}{2} P_{m\ ref}$. **d.** $C = 2C_{ref}$. **e.** $P = \frac{1}{2}P_{ref}$; $M = 2M_{ref}$; $C = 2C_{ref}$.

ADVANTAGES AND LIMITATIONS OF THE LOW-FREQUENCY TRACKABILITY METHOD

Many new design approaches have been presented to allow favorable tone-arm response characteristics. Solutions such as splitting the arm mass by decoupling the counterweight or the cartridge mass have been suggested. Any of these approaches may be analyzed via the trackability method presented. One such example is shown in Fig. 10. Here the effects of arm damping are examined [5]. Fig. 10 indicates that there are advantages of adding a small amount of arm damping, but that higher damping values could cause difficulties with respect to the warp signals.

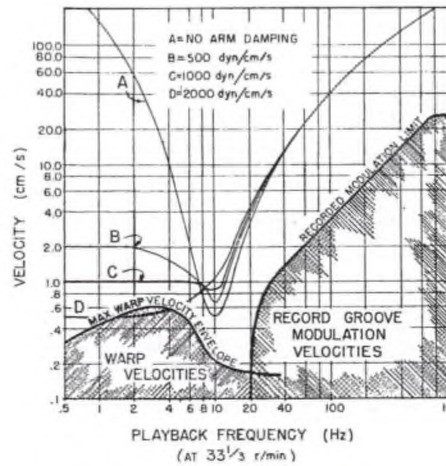

Fig. 10. Low-frequency trackability when viscous damping is applied to the tone arm.

With regard to the low-frequency trackability curve developed, certain assumptions should be stated. The resonance dip in the trackability curve is based on steady-state sinusoidal inputs. Although the possibility exists that enough pulses will occur at a particular frequency to cause the oscillations of the system to reach a maximum, this is not always the case. The data on the warp amplitude–frequency graph (Fig. 3) represents signal sine-wave forms where at least one cycle could be observed. Therefore, for any specific system on any particular record very little can be determined. It is the statistical data which add substance to this approach. The mass-controlled and compliance-controlled regions mentioned do not require the limitations of steady-state inputs and thus are not limited by these assumptions.

SUMMARY

Tone-arm resonance cannot be eliminated—it must be positioned in a frequency region where there exists a minimum number of low-amplitude inputs. This positioning is accomplished by the proper choice of arm mass and dynamic compliance. An improperly designed system can have as much as ten times the sensitivity to warp as a properly positioned system.

The signal spectrum of warps on records was obtained based on a survey of commercially available phonograph records. The record investigation has shown that a potential problem exists due to the amplitude and frequency of warps on records. It was also shown that the quantity of signals increase significantly for warp frequencies below 7 Hz. Although there is always a need for better quality control in the record-pressing process, the aim of this paper was to establish the requirements of the playback system necessary to accommodate existing record warps.

Through an analysis of the tone-arm cartridge resonance, the trackability concept has been extended to include the warp spectrum. The range of effective mass, dynamic compliance, and tracking force available to the consumer were considered in this analysis. For one gram tracking force systems, an imbalance appears to exist between the range of mass and compliance values. The extreme values of both the mass and compliance components are not compatible when playing warped records.

To assure proper system playback performance, it is recommended that a lower limit for the arm resonance frequency should be approximately 7 Hz. This agrees with the values stated in past papers by other authors [6]. This will assure that only a minimum number of low-amplitude signals "challenge" the system at resonance and that the quantity of warp signals between 3 and 5 Hz produce only negligible system oscillations. A 15-Hz upper limit should be established for the tone-arm resonance frequency based on the requirement of eliminating the resonance effects from the audio range. Also, the tracking force must be consistent with the mass and compliance values present in the system.

The trackability concept presented in this paper must be placed in perspective with the need for trackability over the entire recorded frequency spectrum. Above approximately 400 Hz the trackability is a function of other important cartridge parameters—stylus damping, stylus effective mass, and stylus resonance frequency. Only the trackability below 400 Hz has been defined in this paper. No one region is more important than any of the others. Sufficient trackability margin must be maintained over all the signal regions—subaudible, low, mid, and high frequencies—if any of the recorded information is to be properly played. Excessive trackability in any one region at the expense of another is obviously not an acceptable solution.

APPENDIX

I. OBTAINING THE ARM RESONANCE CURVE

There are many methods available for obtaining the experimental resonance curve. All the methods are similar in that they employ a drive mechanism to supply input frequency to the cartridge arm system and a recorder to measure the cartridge output versus frequency.

For this investigation a CBS STR120 record and GR chart recorder were used. The record has a constant amplitude signal from 10 to 500 Hz when played at 33⅓ r/min. Four signal directions are available—vertical, lateral, and the two 45° channels. By the use of a special drive pulley which allows an 11$\frac{1}{9}$ r/min speed on the Thorens TD124 turntable, the sweep frequency can be made as low as 3.3 Hz. An integrator is used to compensate for magnetic pickups, which are velocity sensitive. A calibration curve was determined to compensate for the low-frequency rolloff of the chart recorder and the high-frequency rolloff of the integration circuit.

Another test record which has been used successfully is the B&K pickup test record, QR 2008, which sweeps from 10 to 100 Hz at 33 r/min. Lateral, vertical, and 45° channel tests are available.

II. MEASUREMENT TECHNIQUES FOR TONE-ARM EFFECTIVE MASS

The method used to determine arm mass for this investigation was to resonate the tone arm with a calibrated spring. The diagram of the test setup is shown in Fig. 11. A loudspeaker was modified to supply the input. The paper core was removed and a spring attached to the spider. The spring compliance was measured statically and dynamically and found to have a constant spring rate as a function of frequency. This method is not limited by the frequency range of the input source and has a well-defined resonance peak due to the minimum damping present. When the spring is attached to the tone arm above the stylus tip position and the spring compliance and resonance frequency are known, the effective mass of the tone arm referred to the stylus tip can be calculated. To be accurately defined, the mass data should contain information as to the cartridge weight and tracking force set during the measurements. These parameters alter the mass distribution on the tone arm by movement of the counter balance weight and thus can affect the result. The standard adopted for this investigation was a 6-gram cartridge weight and a 1-gram tracking force.

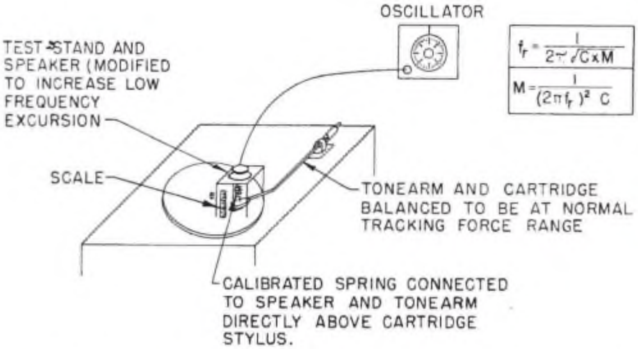

Fig. 11. Experimental setup for tone-arm effective mass measurement.

III. MEASUREMENT TECHNIQUE FOR THE DYNAMIC COMPLIANCE NEAR THE RESONANCE FREQUENCY

With the cartridge and arm mass determined as stated above, then the cartridge tone-arm response curve was measured and the dynamic compliance was calculated from the resonance frequency and tone-arm effective mass.

ACKNOWLEDGMENT

The authors wish to thank J. Kogen, R. Anderson, and B. Jakobs for their assistance and technical contributions and J. Scadron and A. Levin for their assistance in the preparation of this manuscript.

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high-density disk technology

Phonovid - A System For Recording Televison Pictures on Phonograph Records*

KENNETH E. FARR

Westinghouse Research Laboratories, Pittsburgh, Pennsylvania

The capabilities of the ordinary 33⅓ rpm phonograph record for high information packing density, ease of handling and cheapness of quantity reproduction have been utilized in the development of a still-picture television system called Phonovid. Up to 400 still pictures with accompanying audio can be recorded on a 12-in. record. The played-back signal, when processed through a scan converter, is a standard EIA television signal.

INTRODUCTION Slow-scan television is not a new art.[1,2] However, in recent years there has been increasing interest and activity in this field for applications ranging from educational uses to space probes. An excellent comprehensive article describing the Mariner space probe can be found in *Time* magazine.[3] When the available channel bandwidth is much too narrow for ordinary television use, and when low rates of picture presentation are acceptable, slow-scan television comes into its own. The Westinghouse Phonovid system takes advantage of these factors, utilizing the bandwidth capabilities of the ordinary twelve-inch 33⅓ rpm phonograph record to present still pictures whose quality is comparable to broadcast television at a rate of one picture per six seconds.

The system is described in its entirety below, but with emphasis on the phonograph recording and reproducing aspects, since the video techniques as such are considered to be outside the scope of this paper.

GENERAL SYSTEM DESCRIPTION

In a television system there is a fixed relationship between the number of lines per field, the number of fields per unit time, the resolution across a line, and the video bandwidth, for given values of aspect ratio and blanking time ratios. This relationship may be derived as follows, using the simplifying but realistic assumption that close-spaced picture elements representing the finest picture detail are adequately reproduced sinusoidally.

$$T_F = T_L N_F, \tag{1}$$

$$W = N_E / 2B_L T_L, \tag{2}$$

and

$$R_H = N_E / A, \tag{3}$$

from which

$$WT_F = AR_H N_F / 2B_L, \tag{4}$$

where A = aspect ratio (width/height of active picture area), B_L = line blanking factor (active time/total time), N_E = number of active picture elements per line, N_F = number of scanning lines per field, R_H = horizontal resolution in number of TV lines, T_F = time per field, T_L = time per line and W = bandwidth of video signal.

The left-hand member of Eq. (4) may be looked upon as a dimensionless constant which is a function of aspect ratio, horizontal resolution, scanning lines per field, and blanking width. It is thus seen that a direct trade-off is possible between bandwidth and picture presentation rate. In the present system, the aspect ratio and blanking factor were taken at 4/3 and 83 percent,

* Presented October 19, 1967 at the 33rd Convention of the Audio Engineering Society, New York.

respectively, which are the EIA standard values. Using 360 lines per field (arbitrarily chosen), and 208 TV lines horizontal resolution, for a 10 kHz video bandwidth the time per picture, from Eq. (4), is 6 seconds. These are the parameters used in the present system.

An additional practical constraint that was placed on the system parameters was to choose a line rate of 60 per second, locked to the power line. This minimized hum problems, and yielded the quite reasonable set of numbers presented above. For a more detailed discussion of television system standards and terminology the interested reader is referred to the literature.[4,5]

It is thus seen that the video bandwidth requirements fall well within the capabilities of present-day phonograph records. However, low-frequency components of the signal extend down to at least ⅙ Hz, and it is highly desirable to maintain dc transmission for proper reproduction of average brightness levels. Therefore, the video signal is modulated onto a carrier before recording. To make maximum utilization of the available bandwidth, some form of single or vestigial sideband signal should be used. Vestigial sideband FM was chosen for reasons of equipment simplicity as well as because uniform response for both the baseband and the carrier can be achieved at both the outer and inner diameters of the record by the use of amplitude limiting. In addition, the multiplexing of audio commentary on the record with a minimum of crosstalk is facilitated by the use of direct recording for the audio and an FM carrier for the video.

In playback, the audio and video portions of the signal are first separated by filtering. The audio signal is used directly, while the FM picture carrier signal is amplitude limited and then demodulated into a baseband slow-scan video signal. This video signal is processed through a scan converter which consists of two storage tubes and associated circuitry, including a fast-scan or EIA standard synchronizing generator. Two storage tubes are used so that continuous fast-scan output video can be obtained—the stored image is being read out of one tube for six seconds while a new image is being written into the other, after which the reading and writing functions are reversed. The video signal read out of the storage tube is an EIA standard signal. This video waveform is processed as in conventional television practice with regard to blanking, black-level clamping, sync pulse insertion and the like, and delivered out of the scan convertor for distribution over any standard closed-circuit television system. A block diagram of the recording system is given in Fig. 1 and of the playback system in Fig. 2.

THE CAMERA

The camera will not be described in detail, as it is a conventional vidicon camera operating at slow-scan rates.

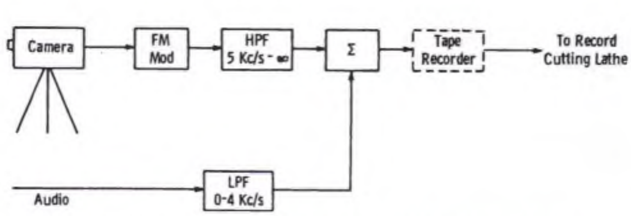

Fig. 1. Block diagram of the recording system.

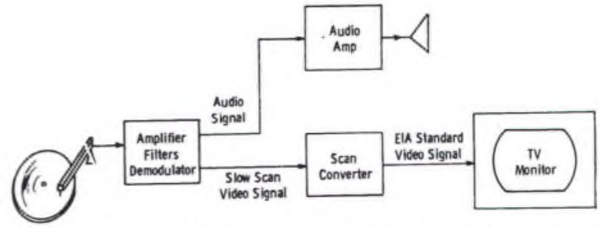

Fig. 2. Block diagram of the playback system.

A flying spot scanner could have been used equally well. Descriptions of slow-scan cameras can be found in the literature.[1,6]

Briefly, a vidicon tube is scanned at a line rate of 60 per second and a field rate of one field per six seconds. A shutter is included which exposes the target for approximately one-tenth of a second, thus effectively taking a snapshot. The image is stored in the target and destroyed in the process of read-out, thus readying the target for the next exposure. The vidicon beam current is chopped and the target output signal is thereby modulated onto a carrier, which is amplified and subsequently detected, blanked, clamped and delivered to the FM modulators. The camera unit includes the slow-scan synchronizing or timing generator, which provides a line scan drive of 60 per second, locked to the power line, and a six-second period field scan. A block diagram of the camera is shown in Fig. 3.

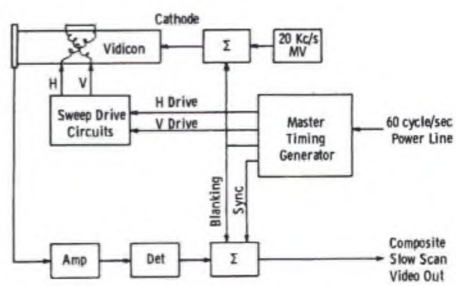

Fig. 3. Block diagram of the camera.

THE PHONOGRAPH RECORDING

A Neuman lathe was used to cut the recording. The recording characteristic had approximately constant velocity above 700 Hz and constant amplitude below this frequency, as shown in Fig. 4. For playback, an ADC-3 pickup cartridge is used in a Garrard Lab 80 turntable. This turntable was selected primarily for its cueing feature, which will be discussed later. The measured frequency response of the particular cartridge used in the experimental equipment is shown in Figs. 5 and 6. The response with a constant-velocity test record (CBS Type

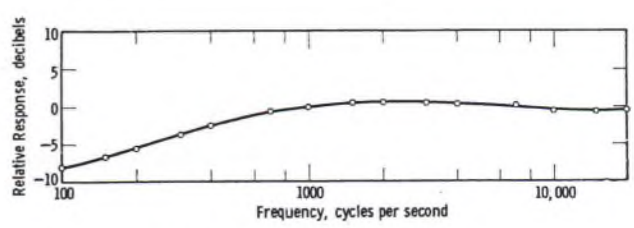

Fig. 4. Recording characteristic.

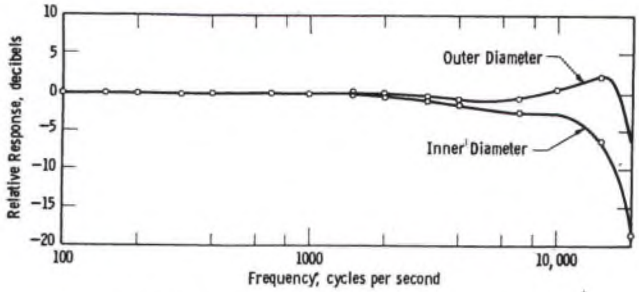

Fig. 5. Velocity response of the pickup cartridge.

STR-120) and flat preamplifier is shown in Fig. 5, while the overall response including the recording characteristic of Fig. 4 and the equalization as used in this equipment is shown in Fig. 6. Response for the outer and inner groove diameters are shown in each case. The useable bandwidth is determined by the noise level and hence by the depth of limiting of the FM carrier which is thus permitted. As can be seen from Fig. 6, this extends upwards to approximately 20 kHz. With the decisions made: 1) to use vestigial sideband FM for the picture information, 2) to allow a 10 kHz video baseband width and 3) to multiplex in the audio signal by direct recording, the frequency spectrum for the record and playback system becomes as shown in Fig. 7.

This spectrum permits, rather conservatively, a video baseband width somewhat in excess of 10 kHz, in which pictures of fairly good resolution can be presented at a six-second rate. Assuming a playing time of 20 minutes for the record, it is seen that 200 pictures can be recorded on each side, with accompanying audio commentary. The audio bandwidth is sufficient for voice reproduction—if better audio quality were desired the bandwidth division between audio and video could be modified by accepting lower video resolution, reduced picture rate, or both.

The video signal is modulated onto the FM carrier and passed through a highpass filter to insure that no lower sidebands extend down into the audio portion of the record spectrum. The audio signal is passed through a lowpass filter to prevent any audio frequencies from extending into the FM channel. These two signals are then added and applied to the recording lathe, through an intermediate tape recorder if desired, as was indicated in Fig. 1. The peak-to-peak amplitude of the audio component of the mixed signal was held to a maximum of 30% of the peak-to-peak amplitude of the FM carrier.

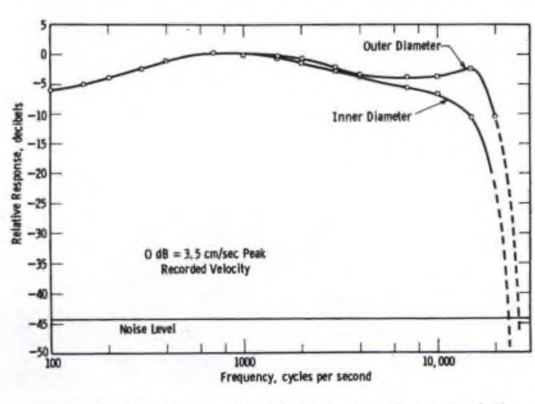

Fig. 6. Overall record/playback characteristic.

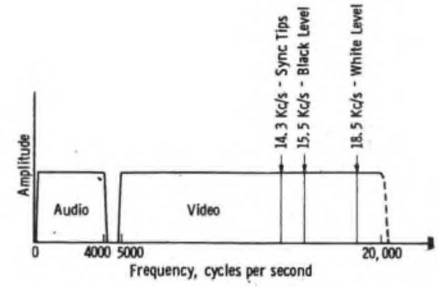

Fig. 7. Record frequency spectrum.

The maximum recorded velocity of the composite signal was 3.5 cm/sec. The ratio of the two signals and the recorded velocity were established empirically to be within the linear range of the system so as to prevent noticeable crosstalk between sound and picture with this particular equipment.

The playback system is essentially the inverse of the recording system, as shown in Fig. 8. After frequency separation, the audio signal is amplified and passed directly to the audio output terminal, while the FM picture carrier is first amplitude-limited and then demodulated to yield the slow-scan video signal for application to the scan converter. It should be noted that the system is reasonably tolerant of wow and flutter; inexpensive record changers have been used with good results. Figure 9 shows a typical picture from the monitor screen with the Garrard Lab 80 turntable. When the record was deliberately off-centered to produce approximately 3% wow, the resulting picture was as shown in Fig. 10.

SCAN CONVERSION

The scan conversion used in the presently described system is of the imaging type, i.e., a complete picture is laid down at the slow-scan rate and stored as a geometric replica of the image before the camera. The storage in this case is in the form of electrostatic changes on an insulated target of a storage tube. An electron beam scans the target at the slow-scan rate while an amount of charge is deposited which is proportional to the brightness of the scene at the point being scanned. By suitably switching the target and other storage tube potentials, a video signal which is proportional to the deposited charge can be obtained at the target electrode when the target is scanned by the electron beam. The read-out is nondestructive in a tube such as the Westinghouse experimental type WX30326 which is used in this equipment. The timing of the read-out scanning is independent of the timing of the writing scanning, thus permitting the conversion of the scanning standards from slow-scan to standard EIA format by switching the storage tube scan-

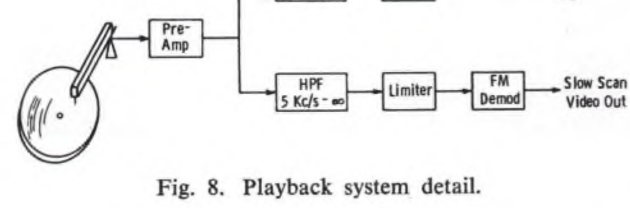

Fig. 8. Playback system detail.

Fig. 9. Normal picture from the monitor screen.

ning rates between reading and writing. Two storage tubes are used—the image is read out at 1/60 sec per field for approximately 6 sec, or 360 repetitive readouts, while the next picture is being written in the other storage tube. The image is then erased from the first tube by scanning the target at an elevated potential, after which the reading and writing operations are interchanged between the tubes. While the reading and writing operations are thus independent, one type of possible interaction exists. Since the incoming slow-scan picture contains a line structure with a different number of lines per field than the fast-scan output format, there exists the possibility of a beat between the line structures which can show up as a moire pattern. To avoid this, the slow-scan picture is scanned vertically, placing the line structures at right angles. The video waveform out of the storage tube is processed in a conventional manner like the signal from a camera tube, including blanking, clamping, pedestal and sync insertion.

The transfer of function from read to write is activated by the incoming slow-scan frame sync pulse. Since the storage tubes are capable of having their targets read out nondestructively for many minutes, it is possible to lift the pickup from the record at any time during the program and "freeze" the picture being viewed. A lecturer can ad lib during this period, then return the pickup to the record and continue the program. It is for this feature that a record changer such as the Garrard Lab 80 was chosen, since it incorporates a cueing mechanism that makes it possible to lift the needle from the record and automatically to return it to the same groove. A block diagram of the scan converter is shown in Fig. 11 and a photograph of the complete Phonovid equipment in Fig. 12.

Fig. 10. Picture from the monitor screen with the record off-center, showing the effect of record wow.

CONCLUSIONS

A method of recording television still pictures on a phonograph record has been described. This system has been shown to be technically feasible, and experimental apparatus has been built and demonstrated. Various trade-offs of audio and video bandwidth, resolution and frame time are possible; the parameters used in the apparatus described in this paper are felt to be a reasonable compromise.

The particular advantages of recording video still pictures on phonograph records are most evident in those applications where mass duplication of the program material is required. Phonograph records can be pressed in quantity quite cheaply. Large quantities of records containing the same program material might be required for educational institutions as visual education supplements to courses. Industrial training programs are another area of potential application. The system thus would seem to have some very practical applications as well as merely having a novelty value. To the best of the author's knowledge this is the first usage of phonograph recordings for other than voice or music reproduction. It should be emphasized, however, that the equipment described was an experimental project and has not been developed into a commercially available item.

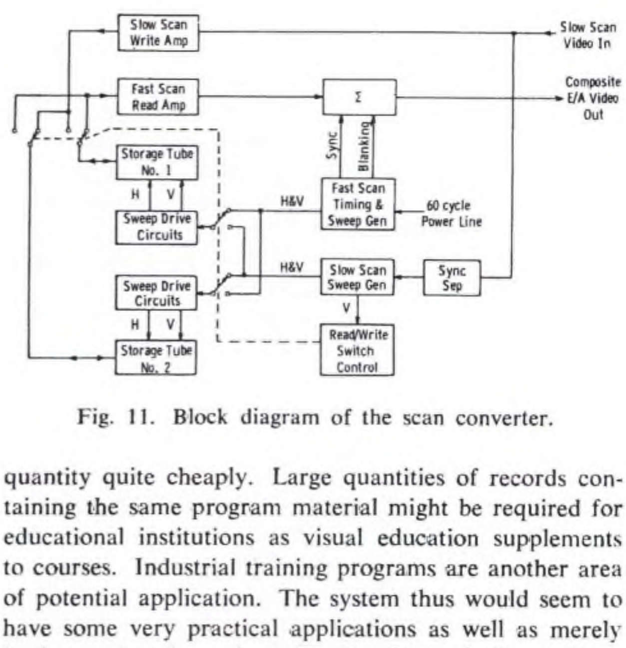

Fig. 11. Block diagram of the scan converter.

Fig. 12. Complete Phonovid playback equipment.

The system could be extended to provide full color pictures in several ways; the most attractive of these appears to be the use of a field sequential slow-scan system, with conversion to simultaneous color signals in the scan converter. It will be apparent that the phonograph recording and reproduction portion of the system is relatively simple and straightforward technically—the area presenting the greatest limitations from both the performance and cost standpoints is the scan converter.

ACKNOWLEDGEMENTS

Acknowledgement should be given to Dr. George C. Sziklai, formerly with the Westinghouse Research Laboratories, for providing the original suggestions for this system. The recording process was under the direction of Cedric R. Bastiaans of the Westinghouse Research Laboratories, and the design of the present embodiment of the experimental apparatus was under the direction of Copthorne MacDonald, formerly with Westinghouse.

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Mr. Farr is a member of the Institute of Electrical and Electronic Engineers, Tau Beta Pi and Pi Mu Epsilon.

RECORDING TELEVISION PICTURES ON PHONOGRAPH RECORDS

ALBERT ABRAMSON

Van Nuys, California

In his paper entitled "Phonovid—a System for Recording Television Pictures on Phonograph Records" (JAES **16**, Apr. 1968), K. E. Farr states that to his knowledge, "this is the first usage of phonograph records for other than voice or music reproduction." I wish to inform him that John Logie Baird invented a process called "Phonovision" in 1927. Baird was a prolific English inventor who experimented with color, stereoscopic, and long-distance television even though his equipment was limited to crude mechanical scanning discs and flying spot scanners. His "phonovision" recordings were made on regular phonograph discs using normal "gramophone" equipment. He recorded 30-line pictures at a rate of 12½ frames/sec. Incidentally, these were the first video recordings ever made and therefore, the art of television recording is some 41 years old. A complete description of Baird and his work showing photographs of his equipment is included in my book *Electronic Motion Pictures* (University of California Press, 1955).

Reply by T. S. Cole

My thanks to Mr. Abramson for the data regarding Baird's "Phonovision". I became aware of this after presenting my paper. My knowledge at the time was obviously incomplete.

High-Density Disc Recording Systems*

LEO M. LEVENS

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Continuing investigation in the field of high-density disc recording has resulted in a 4$\frac{1}{6}$ rpm record with a playing time of 20 hours. Some of the history and reasoning underlying this development is presented, description of the processes and elements of the system is given, and the potential of the system is assessed.

HISTORICAL BACKGROUND

The American Foundation for the Blind has been investigating communications media for blind persons since the 1920's. A need for a medium other than braille was apparent at the outset. A number of workers in the field felt that this need could be met by a suitable disc record system.

The initial research and development culminated in a system that was really advanced for its time. Most important of all, it did meet the requirements of a mass communications medium for blind people. Among its contributing characteristics were the following: In the first place, it was economical, so that a great number of persons could be served at low unit cost. Second, it met the aesthetic and semantic requirements for such a system, for with professional readers working under the direction of a competent observer, material could be presented accurately and yet maintain the interest of the listener. Third, the system was simple in design, rugged in use, and easily maintained for top performance in the field. The improvements which were made in the system came gradually, as the state of the recording and reproduction art developed further, and as better components gradually became more widely available.

The Talking Books produced in the early 1930's were 12 in. 33⅓ rpm records cut with a pitch of 150 lines per inch. A single record contained 30 minutes of reading matter. These were recorded in the studios directly on wax discs.

Support for the Talking Book program was obtained from the U.S. Government through the Library of Congress. The program grew continuously in size and scope. Now, more than 120,000 readers are being supplied with a wide variety of reading matter. Regional libraries throughout the United States circulate thousands of titles to their readers. They are currently increased by 500 titles annually. In addition, a number of periodicals are being recorded and circulated.

As the program grew in size, technical improvements were added by additional research and development. Record pitch was increased to 180, 200, 270, and finally 330 lines per inch. By 1957 the disc contained about an hour of material. Improvements in the state of the art were incorporated into the system. Some of these included the tape master, the lacquer master disc, the feedback cutter, the microgroove record, the hot cutter stylus and the 16⅔ rpm. system. The American Foundation for the Blind constructed a complete manufacturing facility, making it possible for the Library of Congress to order a book by merely specifying its title and number of copies. The plant obtained the copyright, hired the appropriate professional readers, and recorded the master tapes in its studios.

Mastering, electroforming and pressing facilities were developed to maintain the quality of the original recordings. Collating, packaging and shipping services were added. The disc system evolved to a point where a 1½ hour, 16⅔ rpm, 10 in. disc could be supplied to the Library of Congress at a very low cost. These were shipped to the regional libraries as complete books.

THE 8⅓ rpm PROGRAM

In 1956 experimental work on 8⅓ rpm was started, as this speed seemed to hold great promise for much

* Presented October 22, 1968 at the 35th Convention of the Audio Engineering Society, New York.

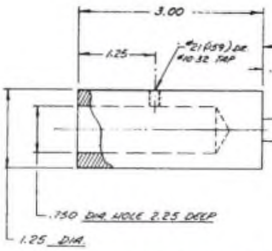

Fig. 1. 8⅓ rpm drive pulley for Scully lathe.

greater recording density and lower cost. In 1958 the American Foundation for the Blind produced 12 in., 8⅓ rpm records with 5 hours of reading material. The Library of Congress started a program of supply of three speed reproducers, 8⅓, 16⅔, and 33⅓ rpm. There are now enough reproducers available to meet the needs of most blind people. The Library of Congress took on the additional responsibility of supplying reading material to all physically handicapped people. Within the next year all blind and physically handicapped people will have three-speed machines of special design available. Periodicals are now produced and circulated on a regular basis on 8⅓ rpm, 3 hour discs.

The American Printing House for the Blind in Louisville, Ky., has carried on a similar program and provides services similar to those of the American Foundation for the Blind.

THE TALKING BOOK SYSTEM

Before going on to a description of the 4⅙ rpm record it might be well to review the Talking Book system. This is a communication system between the writer of the book and the blind or disabled listener. A copy of the book is read by the professional reader in the Talking Book studio under appropriate control; a master tape is produced; this tape is processed into a number of pressed disc copies. The present average is 750 for books, larger for periodicals. The discs, which contain braille information for blind readers, are collated into books, packed into mailing containers and shipped to more than 30 regional libraries. The libraries then circulate the books to the readers by free mail. The books are reproduced on playback equipment of special design supplied at no cost to the reader by the Library of Congress, and are returned to the libraries for recirculation by free mail. It should be noted that the reproducers and the U.S. mail are important elements of the system and have to be carefully considered in the system design.

THE 4⅙ rpm PROGRAM

In 1965, the engineering staff of the American Foundation for the Blind, in cooperation with the Library of Congress, proceeded to investigate a lower speed, 4⅙ rpm. It was felt that the added recording density would provide additional economies and would simplify packaging and shipping, and that the same high quality could be maintained. Initial tests were successful.

In 1966 a complete system, plant facility, disc and reproducer were developed and designed. Prototype discs and reproducers were manufactured and tested. The book was a single 12 in. disc with 10 hours of reading material, with a recording pitch of 420 lpi. This was the *O. Henry Prize Stories of 1965* read by Norman Rose, a popular professional reader. The full capabilities of the disc were not exploited in this recording. An increase of pitch to 700 lines per inch and a reduction in the terminal diameter gives the system a 20 hour capability.

The Disc Process

No special processes are involved in the production of the tape master except for editing the opening and closing announcements appropriately for the disc. A problem arises in mastering the tape, at this time 7½ ips, full track. Each side of the disc contains at least 5 hours of material. A 14 in. reel, the largest available, will hold a maximum of 2 hours. This necessitates using two tape reproducers and alternating the feed from one to the other as the recording progresses. The reproducers used are modified Ampex 3200 duplicator slaves. The disc is cut at twice normal speed, 8⅓ rpm, with the tape running at 15 ips. The tape equipment is equalized for flat output at double speed. Cutting is performed on a Scully Lathe modified for 8⅓ rpm.

System Design

The master was cut at double speed (8⅓ rpm). This necessitated the fabrication of a drive pulley for the Scully Lathe, shown in Fig. 1.

Frequency test records were cut at 11.5 in., 10.5 in., 9.5 in., 8.5 in., 7.5 in., 6.5 in., and 5.5 in. diameters. The discs were cut at a 1 cm/sec stylus velocity with constant amplitude below 1 kHz and constant velocity from 1 kHz to 10 kHz. The frequencies below 100 Hz were attenuated 5 dB at 50 Hz and 10 dB at 30 Hz. The cutting stylus used was a Cappscoop BB with a .00025 in. tip radius. The results are displayed in Fig. 2. The response curves agree in general with the relationship $D = (\sqrt{2}/r)(V/2\pi f)^2$ where D is the maximum allowable peak-to-peak groove displacement, r is the stylus tip radius, V is the linear velocity at the given diameter, and f is the frequency.

A study of the results pointed up the requirements for high-frequency diameter equalization. It was found desirable to maintain response to at least 3500 Hz. An audio system characteristic was designed to meet the require-

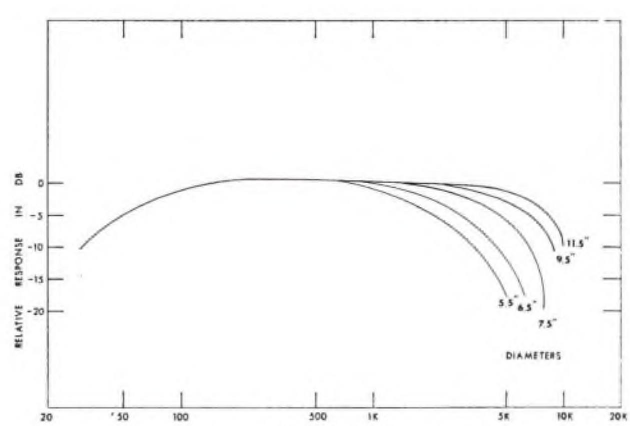

Fig. 2. Unequalized system response for the 4⅙ rpm system. Reproduced with .0005 in. tip radius from lacquer disc. Stylus velocity = 1 cm/sec. Cutting stylus: Cappscoop BB. Constant amplitude to 1 KHz, constant velocity 1 KHz to 10 KHz, reduced amplitude below 100 Hz.

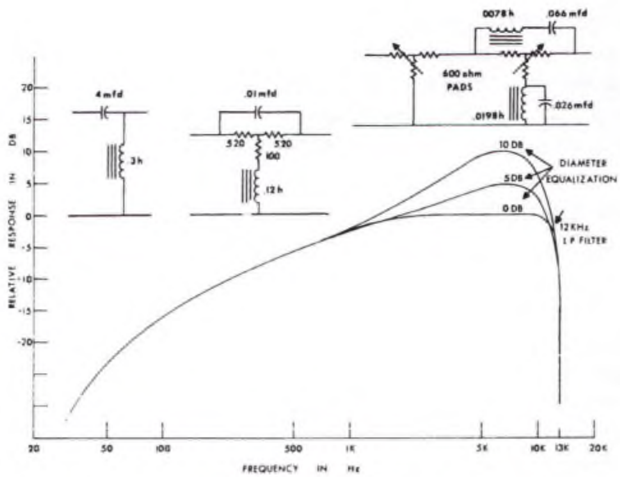

Fig. 3. Double-speed 4⅙ rpm system: audio characteristic.

ment. It was also desirable to reduce the low-frequency response to balance the loss of high frequencies. This is illustrated in Fig. 3. Notice that all frequencies are doubled to meet the requirements of the double-frequency cutting system. A sharp cutoff low-pass filter was provided to limit frequencies above 12 kHz (6 kHz at normal speed). This curve includes all elements from the tape reproduce head to the cutter feedback coil (Westrex 2B). Equalization at 7 kHz was increased 1 dB per inch of decreasing diameter to compensate for losses in the higher frequency area. The filters involved in controlling each section of the response curve are shown above the characteristic curve.

The equalization plus the better high-frequency response of the pressed vinyl record produced a highly intelligible, pleasing recording.

Electroforming and pressing processes were carried out in the normal manner.

THE DISC REPRODUCER

The principal requirement for the 4⅙ rpm reproducer is a motor-turntable combination capable of providing adequate flutter and wow performance at the extremely low rotational velocity, one revolution in more than 14 seconds. The pickup amplifier combination must be capable of low-distortion reproduction of low-velocity fine-groove recording. Stylus velocity is less than 1 cm/sec. Recording pitch runs from 420 lpi up to 700 lpi.

The turntable is a modified "Components" 16⅔ weighted unit with a loosely fitted center bearing. The unit is belt driven by a 600 rpm, 5 W synchronous clock-type motor with a 1/16 in. shaft used as a drive pulley.

A Pickering V-15/AM1 cartridge with a .0005 in. tip radius is mounted in a REK-O-KUT arm with a 1.5 g stylus tracking force. This cartridge feeds a silicon solid state 15 W amplifier.

Precautions are required to minimize ac coupling between drive motor and the pickup cartridge since the cartridge output level is extremely low at 1 cm/sec.

CONCLUSIONS

The 4⅙ rpm disc provides an excellent medium for high information density voice systems. It is most attractive because of its low cost: cost per hour of recorded material decreases in inverse proportion to increase in density. Wear, frequency response, distortion, and accessibility are adequate.

Since a complete book is contained on a single disc, collating, packing, shipping, and storage problems are reduced. It is possible to reproduce the record on a simple, rugged, inexpensive reproducer.

The use of the 4⅙ rpm disc system would make it possible to produce recorded books at the cost of paperbacks. This would even include the cost of a professional reader. With a complete book on a single disc, a library of 500 books would occupy a 5 ft bookshelf. The individual disc occupies a volume of about 10 cu. in., about half that of a paperback and less than a quarter that of a hard covered book. Should it prove possible to reproduce this record on a 10 mil vinyl sheet, a book would occupy about 1 cu. in. Use of vinyl sheet would further simplify shipping problems and reduce costs.

ACKNOWLEDGEMENTS

This research and development work was done under the enthusiastic supervision of John W. Breuel, director of Manufacturing and Sales for the American Foundation for the Blind, and with the excellent cooperation of Robert J. Engler, the Foundation's audio consultant.

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Leo M. Levens was born in New York City. He received the B.S. in electrical engineering from the College of the City of New York. Mr. Levens has taken graduate courses at Columbia and New York Universities and is a graduate of the U. S. Army Advanced Signal Officers School and the U. S. Army Command and General Staff College. Employed as Supervising Engineer, Federal Communications Commission; Director, Talking Book Division, American Foundation for the Blind, Inc. Presently Chief Engineer, American Foundation for the Blind, Inc., in New York City. Served in the U. S. Army Signal Corps as Radio Engineering Officer in electronic countermeasures, now Lt. Col., Signal Corps, U. S. Army Reserve.

A Mechanical Disc Recording and Reproducing System With High Storage Density and High Rate of Transmission*

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A record is described with a storage density 100 times greater than that of the usual stereo LP record. The new reproducing process enables the mechanical replay of signals within the MHz range. The system can be used for the transfer of television programs or multichannel sound recordings.

INTRODUCTION Many specialists throughout the world have been fascinated by the problem of overcoming the technical limitations of the phonograph record. In almost every decade there has been a considerable step forward, such as the advent of the long-playing record and, some ten years later, the advent of the stereo record. Every one of these steps increased the information density on the recording medium. The phonograph record, with its easy duplication process and quick accessibility of information, achieved a storage density which, up to now, has been unsurpassed by any other recording medium.

At the same time, the increased storage density also brought with it an increase in the transmission information. The long-playing record extended the frequency band to about 15 kHz. Cedric R. Bastiaan reported on the possibility of increasing the cutoff frequency during playback to the region of 100 kHz by using an extremely low dynamic mass so that another 2–3 octaves above the audio band could be utilized (JAES, 15, 389 (1967)) However, in addition to the unexplored possibilities he also very convincingly pointed out the limits of traditional phonograph reproduction, which are set by the minimum practical values of the dynamic mass.

It thus seemed as if, with this high technical achievement for the storage of sound, the ultimate limits for the mechanical tracing of signals had been reached. For this reason our initial interest was focused not on playback, but rather on the question whether the storage capacity of a disc pressed in plastic material had been fully utilized. This question was very quickly answered in the negative, for experiments in pressing showed that it was possible to produce from matrices a plastic surface, the roughness of which was less than 100 angstroms. Such a smooth surface could accept considerably smaller signals than those found so far on a phonograph record. We developed methods of cutting and pressing, which we termed "high-density storage techniques." These methods enabled us to store signals to such a high density, that each signal element occupied an area of less than 10 μm^2. A signal amplitude of only 0.5 μm was enough to achieve the required differential from the surface roughness. We reached the conclusion that, in the interest of the highest possible storage density, the information should be cut in the form of a modulated carrier. The introduction of frequency modulation proved to be of particular advantage.

HIGH-DENSITY STORAGE TECHNIQUES AND PRESSURE TRACING

Figure 1 shows, at the same magnification, grooves of a traditional record on the right, and the high-density

* Presented October 13, 1970, at the 39th Convention of the Audio Engineering Society, New York, by G. M. Nathan (British Decca).

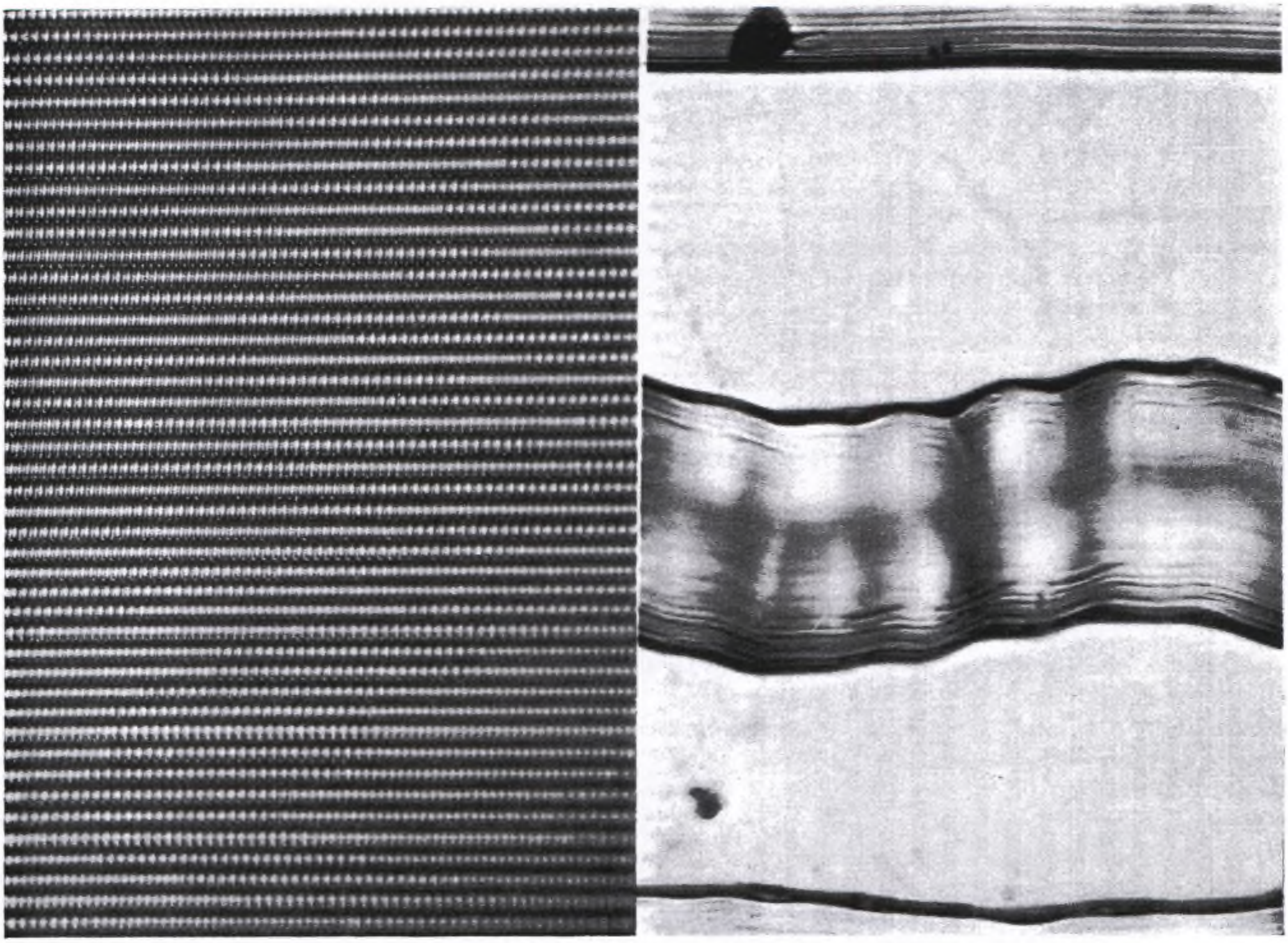

Fig. 1. Comparison of relative size of grooves in video disc (left) with conventional audio groove (right).

grooves cut with a frequency-modulated carrier on the left. The variation of the carrier wavelength with the signal can be seen clearly. This direct comparison shows convincingly the gain in storage density. The recording shown here has a groove density of 3300 lines/inch. The shortest wavelength is about 2 μm. Viewed from the present state of the art, an increase in the groove density should be possible and could be utilized. Taking into account the discrete amplitude steps which make up the analog recorded signal, the high-density storage record has an information density of 500,000 bits/mm^2. The channel capacity is according to Shannon:

$$C = B \log_2 (1+P/N) \quad \text{bit/s}.$$

For a bandwidth of $B = 3$ MHz and a signal-to-noise ratio of 40 dB, i.e., a power ratio of signal P to noise N of $P/N = 10{,}000$, one obtains

$$C = 4 \times 10^7 \quad \text{bit/s}.$$

The storage density D is calculated according to

$$D = C \times (\text{groove density})/(\text{groove speed}).$$

At 1500 revolutions per minute and a disc radius of, for example, 65 mm (2¼ inches), one obtains the value for D previously mentioned, i.e., 500,000 bit/mm^2. Compared with the traditional record, this means an increase in storage density of approximately two orders of magnitude.

Knowing the storage possibilities on disc, we had to inquire into the feasibility of tracing the stored information, and in this connection two questions arose:

1) Is it at all possible to play back such high information densities in a simple manner such as, for instance, by a mechanical tracing method as used with phonograph records?

2) Could the gain in information density be utilized not only to increase playing time, but also to increase the transmission of information?

If we confine ourselves to the conventional method of record tracing, both questions must be answered in the negative. As is well known, this method of tracing requires the dimensions of the contact area between tracing tip and disc to be smaller than half the area occupied by the smallest signal element. In view of the surface pressure occurring here, it seemed hopeless to find a practical and realizable solution using the conventional phonographic method, particularly if we recall the extensive theoretical and practical work on sound groove deformation.

The problem of utilizing the highly concentrated information to widen the reproduced frequency band is made even more difficult by the increased storage density. In this connection the problems of scanning loss function and cutoff frequency have been thoroughly investigated by many specialists. Elasticity of the groove walls and the finite inertia of the tracing tip appeared to present an insurmountable barrier to tracing much wider frequency bands by means of electromechanical transducers.

What Were the Starting Points for Overcoming these Limitations?

It is well known that for conventional tracing it is axiomatic that the almost stiff groove walls of a modulated groove passing the tracing tip cause movement of the tracing tip, and this movement is transferred to an electromechanical transducer. We have termed this principle movement tracing. It fails during the tracing of high frequencies which, due to the inertia of the tracing tip, give rise to forces acting on the groove walls, which are great enough to collapse the wavetrains of the recording completely. Under these conditions no movement of the tracing point takes place.

We can now ask whether the signal in the groove above the cutoff frequency has been erased by the tracing tip, i.e., has it disappeared in that region? Surely not, it is only present in another form. The action of the playing weight deforms the disc surface, and the wavetrains corresponding to the recording produce pressure differences at the point of tracing. A pressure transducer designed for this purpose must, therefore, be able to transform these pressure differences into electrical oscillations. A pressure transducer is understood to be a device with which these pressure variations can be measured closely. If we pursue this argument we can conclude that pressure tracing must be used above the cutoff frequency for the tracing tip, whereas movement tracing must be used below that cutoff frequency. During pressure tracing the principle is reversed; the tracing tip is not moved by the groove walls, but to a first approximation, the tracer remains stationary while the modulation in the groove is deformed.

How Can We Solve the Problems that Arise in Tracing Signal Elements of Such Minute Dimensions as are Recorded on the High-Density Storage Disc?

How, when using a pressure pickup, do we arrive at manageable tracking forces and surface pressures, where neither the disc material nor the tracing tip is destroyed? The solution is a special shape for the tracing tip which enables it to lie on a larger surface area. To achieve this goal it is important that the tracing diamond have an unsymmetrical shape in the direction of groove movement. On one side it is provided with a smooth curve whereas the other consists of a sharp edge. The smooth curve forms the front of the tracing tip. By this means the tracing tip can, like a skate, easily glide over the groove modulation without destroying it. During the tracing process, the tracing tip is subjected to a constant force which has superimposed on it an alternating component. This alternating component corresponds to the recorded signal, and is produced at the sharp edge on the back of the skate-like tracing tip. The compressed modulation hills are suddenly released as they glide under the tracing tip. This sudden release is detected by the pickup. This explains why we do not receive a mixture of information from the whole area of contact, but only the information that is traced by the edge of the tracing tip. The voltage developed by the pickup is directly proportional to the traced wavelength.

What Transmission of Information Can Be Achieved by Pressure Tracing?

Which new frequency band has become usable? Using materials available at present, pressure pickups can be built having a frequency range of 5–6 octaves. At present the limit is just short of 6 MHz. Thus for mechanical tracing, six further octaves have become usable. It is likely that this does not represent the absolute upper limit. Little is known about the mechanical properties of plastics subjected to the forces and high frequencies encountered here. Experimental results so far indicate that it is just these high tracing velocities and the resultant extreme brief mechanical contact which allow the loading of the disc surface to be far higher than would be permissible under static conditions. Resistance to wear of a high-density storage disc with a pressure pickup is greater than that of a normal phonograph record. Several materials tested stood up to many thousand playings.

What Are the Transfer Characteristics of this New System?

As mentioned before, it is advantageous if the recording is in the form of a frequency-modulated carrier. Here the recorded carrier amplitude remains constant, which permits effective utilization of the disc surface and constant conditions for the recording process. As opposed to direct recording on phonograph records, in this type of recording the nonlinear distortion does not depend on the recorded wavelength. It arises only in the processes of modulation and demodulation. Tracing distortion, pinch effect, and tracking distortion are terms which in this connection have lost their importance.

To a certain extent the useful dynamic range is dependent on wavelength and is limited by noise generated in the tracing process. The good surface quality of high-density storage foils, as ascertained with an electron microscope, is also evidenced by the values measured for noise during playback. For a carrier frequency of 3 MHz a signal-to-noise ratio of better than 65 dB was obtained.

The analyzer used in this measurement had a bandwidth of 4 kHz. From this result we conclude that the high-density storage foil will also be suitable for the storage and playback of wide frequency bands. A center frequency of 3–4 MHz was chosen for the storage of television signals. In this case the ratio of peak signal to effective noise is about 40 dB after demodulation. A further advantage of this system is that it is very insensitive to rumble originating from the playback drive mechanism and that low frequency stray fields induced in the pickup do not lead to hum. It is, therefore, possible to envisage very simple drive mechanisms, also for high-fidelity sound reproduction.

VIDEO RECORDING SYSTEMS

In developing the high-density storage record and pressure tracing, one of our aims was to create an inexpensive audiovisual storage medium and a corresponding reproducer which because of its low price would find wide distribution. The idea of storing video signals on a record is not new. The archives show that as far back as 1927 the British inventor Baird engraved video signals on a disc by means of a record cutting lathe, and reproduced them on a record player. Using the record rotating at 78 revolutions per minute, Baird had at the time at his disposal a bandwidth of only 5 kHz. He was therefore able to reproduce his picture with a horizontal resolution of only

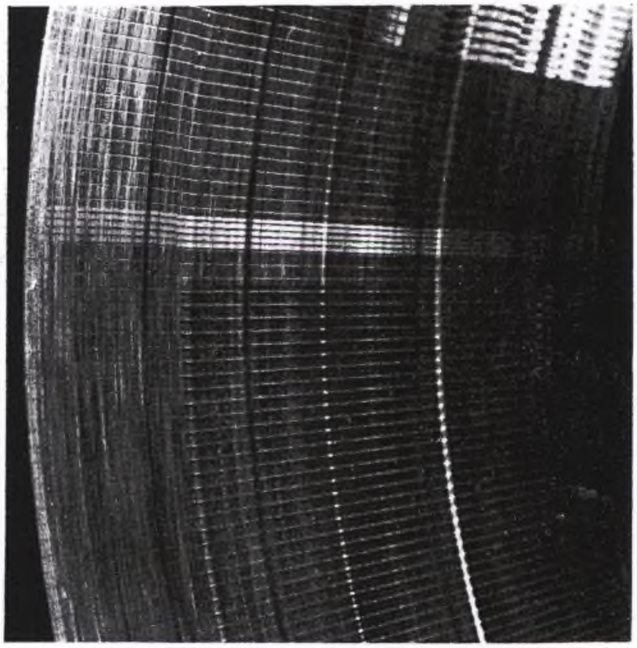

Fig. 2. Grove arrangement of video disc showing modulated grooves, synchronizing pulses, and blanking interval.

15 picture points at 30 lines with a frame rate of 12½ frames per second. Even if this picture does not bear comparison with today's television picture, its inventor had nevertheless stored video signals for the first time according to Edison's well established method. Thus, in 1927, the phonograph record became the *first* video storage medium.

The high information flow and its associated large bandwidth required by the television picture of today's standard appeared to give little hope for the specialist wishing to resurrect Baird's experiments.

The Phonovid process, developed in 1965 by Westinghouse, also used the phonograph record for the storage of video signals. This time the television pictures corresponded to today's techniques, but the inadequate storage density and information flow of long-playing record techniques permitted reproduction of only 200 still pictures, using an intermediate storage medium. The storing of moving television pictures according to today's standards on mechanically traceable discs was made possible by the high-density storage techniques reaching a storage density of 500,000 bit/mm^2, and the pressure tracing which allowed a reproduced information flow of more than 40 million bit/s. This technique makes it possible to record 3×10^{10} signal elements on a disc having a diameter of 30 cm (12 inches). If we use 30 million of these per second to produce the television picture—this corresponds to a video bandwidth of 2.5 MHz and a signal-to-noise ratio of 40 dB—we obtain, in theory, a program of 1000 seconds, i.e., just over a quarter of an hour. This fact made the phonograph record with its fast duplication process and information accessibility highly topical as a video storage medium.

Figure 2 shows part of the grooves of a video recording on a high-density storage foil. The speed of rotation during recording is chosen so that each rotation records one complete, i.e., two half-pictures. For this reason it is easy to recognize the individual line synchronizing pulses, as well as the frame blanking gap with the frame synchronizing pulse. The second half of the recording, for instance, no longer contains any picture modulation but only the synchronizing pattern and the black level value.

The size of a recorded television picture on the video disc is smaller than 2 mm^2. One would think that dust falling on the plastic would make interference-free reproduction impossible. If we wanted to achieve the magnification of television screen size to recorded picture size by photographic means, this would really be a very difficult problem. However, in mechanical tracing we have the advantage of the tracing tip always automatically cleaning the groove, i.e., the dust is swept out of the groove.

A plastic film was chosen as the recording medium for the video disc. The use of thin plastic has several advantages. To obtain the high speed of rotation of 1800 revolutions per minute required for television pictures, a film can be driven much more uniformly than a solid disc, i.e., without radial or vertical movement. However, the chief advantage of a video storage medium which is to be processed in large numbers is that the plastic is suitable for pressing processes which are exceptionally fast. A process at present being developed makes the ratio of playing time to duplicating time 1000:1.

Pickup and Stylus

The video disc is played by means of a pressure pickup the principal action of which was explained earlier[1]. The whole pickup is shown in Fig. 3. The skate-shaped sapphire or diamond tip is rigidly held to a piezoceramic element which serves as the actual transducer. The electric signal is taken from the electrodes situated at the sides of the transducer. The pickup is dimensioned so that no mechanical resonances occur within the transmission band. The tracking force of the video disc pickup is about 0.2 gram. Since the pickup experiences no movement within the groove, this tracking force is fully adequate for the groove to guide the stylus.

Video Disc Reproducer

Figure 4 shows the main structure of the playback equipment. There is a certain resemblance to record players, but it differs from these in two important respects:

1) The pickup has a positive feed. It is fastened to a slide which moves one groove width along the radius for each turn of the disc. This can be achieved by means of a cord and pulley arrangement, as shown in the figure. The groove itself provides fine guidance of the stylus, which is mounted on a resilient material for this purpose.

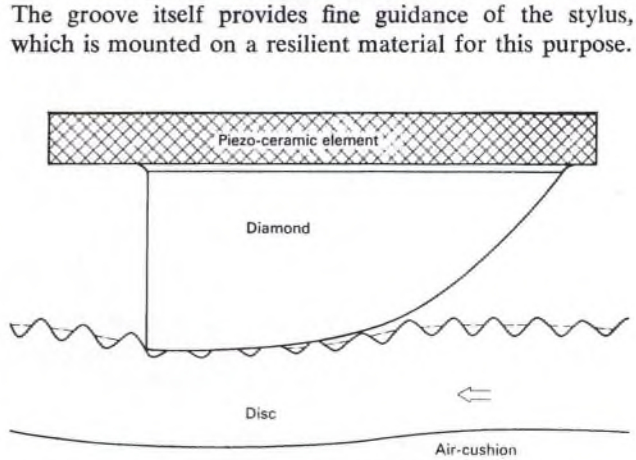

Fig. 3. Elastic deformation of video disc during pressure pickup scanning.

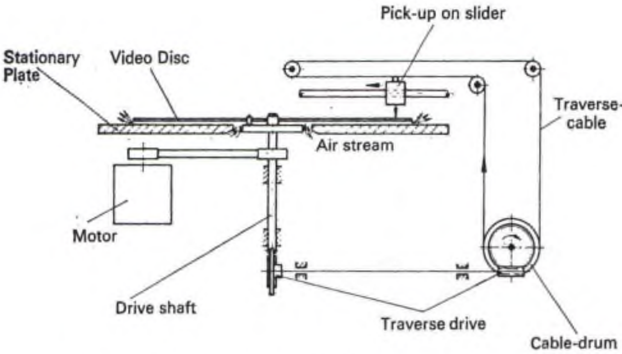

Fig. 4. Sectional view of video disc reproducer.

In this way small radial movements of the disc can be accommodated.

2) As opposed to phonograph record players, the playback equipment does not possess a turntable that rotates with the record. The video discs are driven by means of a central drive and rotate above a stationary table. At the rotation of 1800 revolutions per minute at present in use, a thin air cushion is formed between the disc and the table. Since video discs consist of plastic film, the speed of rotation makes them accept the contour of the table. Their passage through the air cushion stabilizes them so that up and down movement in the record area is less than 50 μm. Incidentally, this principle is also used with magnetic film storers.

In all other audio-visual systems it is usual to employ a separate track for sound recording. In the video disc, both the picture and sound have been accommodated in one track. The sound requires no additional storage area. The separation of picture and sound signal is achieved purely electronically. Such a process is of value if the picture track is recorded continuously, as in the video disc. For a discontinuous picture track, as used in magnetic video recording, interference on the sound, due to constant interruption of the video signal, could be avoided only by devoting considerable effort to the equipment side.

Two sizes are envisaged for introducing the video disc to the market. A 21-cm (8-inch) disc with a playing time of 5 minutes, and a 30-cm (12-inch) disc with a playing time of 12 minutes. We have noticed that many programs would fit into these playing times. For the playback of longer programs an automatic changer could be envisaged. A two-hour program would require a record pile only a few millimeters high.

Since the video disc grooves are very narrow, i.e., less than 8 μm (0.3 mil) wide, and the tracking force is only 0.2 gram, it might be concluded that playing a video disc is a very difficult matter. The reverse is the case. The playback equipment, when in use, can be exposed to considerable shocks without markedly affecting sound or picture reproduction. It is even possible to play back with the equipment mounted vertically.

The robustness of the records is readily shown, by simply disengaging the pickup transport. Since the stylus is resiliently suspended, it will be carried along by the spiral groove for a little distance, and will then jump back to its original position, right across the grooves. This means that part of the program is constantly repeated. This effect can be utilized in educational programs. By constant repetition a particular movement can be studied very closely. Our investigations have shown that neither the stylus nor the record suffer any damage during this unusual treatment. With some record materials, a drop in quality only occurs after more than 1000 repetitions.

SUMMARY

By means of the high-density storage disc a storage medium has been developed which follows in the tradition of Edison's cylinder and Berliner's phonograph record but which, in combination with a special mechanical pickup, serves for the storage not only of sound, but also of television pictures. Reproduction is by means of an uncomplicated robust player. We believe that these new properties have provided the mechanical storage medium, which in its new form has attained one of the highest storage densities, with a good chance to maintain its position alongside magnetic and optical storage systems.

THE AUTHORS

Gerhard Dickopp (AEG-Telefunken) was born at Knapsack, Germany, in 1933. At Aachen Technical University he studied general electrical engineering. In 1960 he joined the Rogowski Institute in Aachen as a scientific assistant. In his dissertation he dealt with the theory of networks with variable parameters.

In 1967 Dr. Dickopp started work in the basic research department of the AEG-Telefunken Record Player and Tape Recorder Division in Berlin. There, as an associate of Eduard Schüller, he devoted his attention to the basic problems of video disc development. In 1970 he succeeded Schüller as manager of the basic research department for record players and tape recorders. Dr. Dickopp belongs to the team of inventors of the video disc by dealing principally with the theory of scanning the video disc.

Hans-Joachim Klemp (TELDEC) was born in Berlin in 1921 and studied electrical and mechanical engineering there. From 1947 to 1950 he was employed as a development engineer for sound engineering with the Tobis film company and from 1951 to 1954 he managed the development section for magnetic sound recording with the Defa film company, devoting particular attention to drive mechanisms and amplifiers.

In 1951 he became the closest associate of Horst Redlich with the Teldec record company. His work covered all phases of recording technology. He played a significant role in the preliminary work for the stereophonic LP record and the Royal Sound stereo method. Between 1965 and 1970 Hans-Joachim Klemp accomplished important creative work in the development of the video disc in cooperation with Horst Redlich and

DICKOPP, KLEMP, REDLICH AND SCHÜLLER.

the AEG-Telefunken engineers Eduard Schüller and Dr. Dickopp, in the Teldec development laboratory in Berlin.

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Horst Redlich (TELDEC) was born in Berlin in 1921 and studied precision mechanical engineering there. In 1947 he joined AEG as a development engineer and worked mainly in the field of magnetic sound recording for studio purposes. From 1950 to 1951 he managed the development work at the UFA film company which resulted in the first magnetic sound track films (35 mm).

In 1951 he became chief engineer of the Teldec record company. Under Redlich's management Teldec produced the first stereophonic LP records in Europe during 1955. In the course of further years he developed the improved-method Royal Sound stereo. Since 1959 Horst Redlich has been technical director of Teldec, and during recent years has devoted his attention to the development of dense-storage technology.

Eduard Schüller (AEG-Telefunken) was born in Liegnitz, Germany in 1904. He studied electrical engineering at Berlin Technical University and worked then at the Heinrich Hertz Institute. As early as the 1930's, he was devoting attention to the problem of magnetic sound recording at the instigation of the free-lance inventor Fritz Pfläumer. In 1933 he joined AEG as a basic engineer and two years later, following his development of the magnetic tape head, he produced the first tape recorder (later called the Magnetophon). During the following years Schüller established production shops for domestic tape recorders for AEG.

During 1953 he completed the development of his diagonal recording method which is now being used all over the world for magnetic video recording. As ex-manager of the basic research department of the AEG-Telefunken Record Player and Tape Recorder Division, Eduard Schüller played a prominent part in the development of the video disc in collaboration with his colleagues Redlich, Klemp, and Dr. Dickopp.

A Long-Play Digital Audio Disk System*

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The development of video disk systems has made long-play digital (PCM) audio disk systems possible. The bandwidth required for the two channels of digital audio signals is less than that of video signals and, combined with a reduction of the revolution, makes the longer playing time possible.

For a further improvement in packing density, a kind of run-length-limited code is adopted. The code is known as 3PM (three-position modulation), and the packing density can attain 150% of MFM coding for the same minimum wavelength to be recorded. This is achieved at the expense of the decreasing jitter margin, which is relatively easy to solve in optical disk systems.

A playing time of 2½ hours is compiled on one side of an optical disk with a diameter of 303 mm. The sampling rate is 44.056 kHz and each of the two channels is coded by 16-bit linear quantization. The revolution is 450 r/min. Code errors are analyzed for each revolution of 1800, 900, and 450 r/min on the plane of bit error rate and bit error correlation coefficient. An effective error-correcting scheme called "cross interleave" has been developed which makes it possible to decode using various types of decoder from a simple erasure type to the complex "crossword" type while maintaining full compatibility.

0 INTRODUCTION

A long-play digital audio disk system was developed after three years of studies to increase packing density. Table 1 shows our steps of development [1]–[3], which indicate that the disk size has remained the same, while the playing time has increased by five times, without decreasing the minimum wavelength to be recorded.

This advance is mainly achieved by changing the coding method from NRZ-FM (video format) to MFM and then to 3PM [4]. Accordingly, revolution was reduced from 1800 r/min to 900 r/min to the present 450 r/min.

The quantization was changed from 13 bit nonlinear to 16 bit linear so that the quality of the master recording will be maintained in the final pressed disk to be played in the home. Another great advance is found in the error-correcting schemes. The characteristics of code errors are different depending on the revolution rate of the disks. These are analyzed, and an effective correcting method is described in this paper.

* Presented at the 62nd Convention of the Audio Engineering Society, Brussels, Belgium, 1979 March 13–16; revised 1979 September 5.

1 DIGITAL AUDIO DISK SYSTEMS

Fig. 1 shows the total digital audio disk system. The master recorder is the PCM-1600 VTR-based two-channel 16-bit system where all processing is carried out digitally until the signal reaches the digital-to-analog converters in the player. For this reason the sampling rate of the system is set at 44.056 kHz in correspondence with the PCM-1600.

The process from cutting to the final production of disks does not differ from that of video disk systems. The player (Fig. 2) consists of a signal pickup part (disk drive, tracking and focus servo, and He-Ne laser tube and its optics) and a signal-processing part (PLL, frame and bit synchronization, demodulation of 3PM [4], time-base correction, error correction, and D/A). The former resembles that of a video disk player, but as the revolution rate is 450 r/min, higher stability is required. The latter is described in the following sections.

2 CODE-ERROR ANALYSIS

The possible causes of code errors in optical digital audio disk systems are:

1) Defects in pits on disks

Table 1. Steps for development of a long-play digital audio disk system.

Step	1	2	3
Date completed	1976 Sept.	1977 Sept.	1978 Sept.
Coding	NRZ-FM (analog video format)	MFM	3PM
Playing time [min]	30	60	150
Revolution [r/min]	1800	900	450
Track pitch [μm]	2.5	1.7	1.3
Minimum wavelength [μm]	1.1	3.4	2.4
Diameter of disk [mm]	303	303	303
Sampling rate [kHz]	44.1	44.1	44.056
Number of channels	2	2	2
Quantization [bit]	13 nonlinear	13 nonlinear	16 linear
Error-correcting schemes	Combination of random and burst error correction		Cross interleave system
References		[1], [2]	[3]

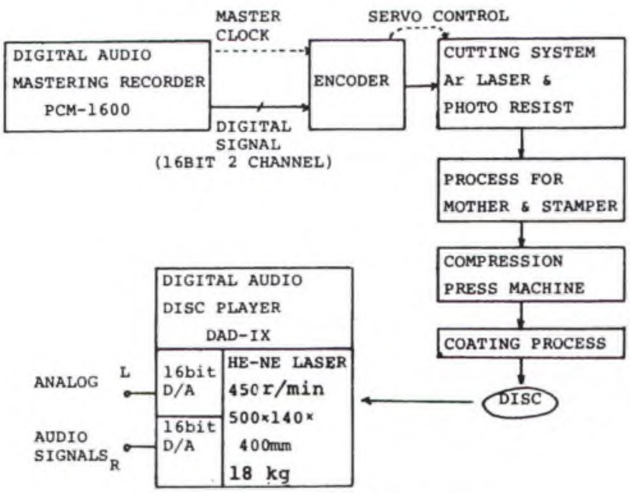


Fig. 1. Digital audio disk systems.

2) Defects in metallic coating on disks
3) Bubbles, irregular refraction or other defects in disks
4) Dust, scratches, fingerprints on disks
5) Defocusing
6) Mistracking
7) Noise
8) Jitter
9) Fluctuation of RF levels
10) Intersymbol interference.

If the revolution rate of disks is reduced and the packing density increased, the length of the code errors caused by 1)–4) will be increased, but the bit error rate will remain the same. But reducing the revolution rate might make the flutter factor worse and thus code errors 5) and 6) might increase. Besides that, the higher the packing density, the more severe the code errors generally become caused by 7) and 8).

The design of error-correcting schemes largely depends on the characteristics of the code errors, and thus the selection of the revolution rate should be made very carefully. In this paper the analysis of code errors is carried out based on a simple statistical model called the Gilbert model [7], [8], as shown in Fig. 3. B and G express the states of error and no error, respectively, and the letters T and t are the transition probabilities. The characteristics of error are expressed by two parameters, the bit error rate γ and the bit error correlation coefficient ζ:

$$\gamma = \frac{T}{T+t} \tag{1}$$

$$\zeta = 1 - T - t\ . \tag{2}$$

Fig. 4 shows an example of the difference in the error characteristics due to the revolution of the disks on the $\gamma-\zeta$ plane. The calculation is carried out by comparing the block error rate before and after error correction for the following three cases: 1) 1-bit error, 2) 2-bit random error, and 3) 15-bit burst error. The length of the block is between 60 and 75 bits.

The block error rate in blocks of n bits is expressed as follows:

$$E_{\mathrm{BLK}} = 1 - \frac{T}{T+t}\,(1-T)^{n-1}. \tag{3}$$

The probability of 1-bit error E_1, 2-bit random error E_2, and that of burst error within 15 bits E_{15B} in the n-bit block is expressed as follows:

$$E_1 = \frac{Tt}{T+t}\,(1-T)^{n-3}\{2(1-T)+(n-2)t\} \tag{4}$$

$$\begin{aligned} E_2 = \frac{Tt}{T+t}\,(1-T)^{n-5}[&(1-T)^2\{2(1-t)+T\} \\ &+ (n-3)(1-T)t\{2T+(1-t)\} \\ &+ \frac{1}{2}\,t^2T(n-3)(n-4)] \end{aligned} \tag{5}$$

$$\begin{aligned} E_{15B} = \sum_{i=1}^{15} \frac{Tt}{T+t}\,&(1-t)^{i-1}(1-T)^{n-i-1} \\ &\times \{2+(n-i-1)t\}. \end{aligned} \tag{6}$$

The data shown in Fig. 4 are to some extent not relevant because the disks are produced at different times under normal room conditions. Usually the quality of disks is dependent on production lots rather than on the revolution or the error-correcting schemes.

If precise data are necessary, a single disk specially made to include various revolutions in various different positions must be prepared.

Nevertheless a general trend of errors is obviously shown in Fig. 4, where we find that the value of the bit-error correlation coefficient decreases as the revolution is reduced, and that the bit-error rate does not significantly depend on the revolution.

Fig. 2. Digital audio disk and player DAD-1X.

Specifications

General	
Playing time	2½ hours (maximum)
Revolution	450 r/min
Number of channels	2 channels, digital
Sampling frequency	44.056 kHz
Quantization	16-bit linear/channel
Dropout compensation	Newly developed error-correcting code
Dynamic range	Better than 95 dB
Total harmonic distortion	Better than 0.03%
Frequency response	2 Hz – 20 kHz ± 0.25 dB
Player System	
Signal detections	Optical reflection type with a He-Ne laser
Wow and flutter	Undetectable
Dimensions	520(W) × 140(H) × 400 mm(D)
Weight	18 kg
Power consumption	75 W
Disk	
Dimensions	303 mm diameter
Thickness	1.1 mm
Material	Polyvinyl chloride, coated with reflection material

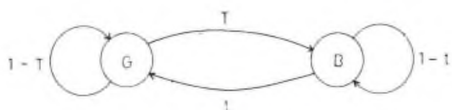


Fig. 3. Gilbert model.

3 ERROR-CORRECTING SCHEMES

In this system an effective error-correcting scheme called "cross interleave correction" is adopted. This method basically consists of delayed interleave and at least two sets of delayed parity words [9]. Decoding is based on the pointer-erasure method with which the pointer of the erroneous word is obtained by CRCC.

For further correcting ability "crossword decoding" is possible while full compatibility with the pointer-erasure decoding is maintained.

3.1 Encoder

Fig. 5 shows the code format and encoder. One frame consists of an 18-bit synchronization word and three subblocks. Each subblock includes two information words (L_i, R_i), two parity words (P_i, Q_i), and CRCC.

The encoder consists of simple delay memories, exclusive – or gate, and CRCC encoder. The parities P_i, Q_i are composed by the following equations:

$$P_i = L_i \oplus R_i \tag{7}$$

$$Q_i = L_i \oplus R_{i-3} \oplus P_{i-8} \tag{8}$$

where $\oplus$ means modulo-2 summation (exclusive – or).

3.2 The Simplest Basic Decoder

Fig. 6 shows the simplest decoder, which only utilizes the parity words P_i. (The parity words Q_i are ignored.) Each subblock is checked by CRCC decoder, and an error pointer of 1 bit is fed to the parity P decoder after passing through the same delay as the main information words.

Within the P decoder the syndrome word

$$Sp_i = L'_i \oplus R'_i \oplus P'_i \tag{9}$$

is calculated, where the primes indicate the received word which might include errors. If $Sp_i = 0$, that is, all the error pointer bits indicate no error, the words L'_i and R'_i are considered not to include any code errors. If $Sp_i \neq 0$, and the error pointer bit of L'_i indicates error (R'_i and P'_i are no error), for instance, L'_i is supposed to be erroneous, and since the error pattern appears in Sp_i, the error correction is carried out by the following formula:

$$L_i = L'_i \oplus Sp_i. \tag{10}$$

Error correction is possible if the erroneous words for one syndrome are less than 2. Related subblocks are separated by 16 subblocks from each other, and therefore a burst error within 16 subblocks (1290 – 1308 bits) can be corrected by this simplest decoder. A guard space of 32 subblocks is necessary for correction.

3.3 Multiple Decoders

Fig. 7 shows a double decoder which computes two

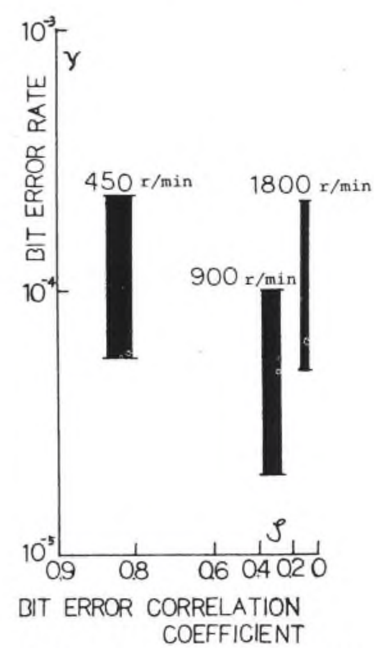


Fig. 4. Code error analysis on $\gamma - \zeta$ plane.

syndromes, Sp_i and Sq_i.

$$Sq_i = L_i \oplus R_{i-3} \oplus P_{i-8} \oplus Q_i \ . \qquad (11)$$

Using each syndrome Sp_i and Sq_i, one error word can be corrected by the method described above. Supposing that both L'_i and R'_i are erroneous, correction by the basic decoder as shown in Fig. 6 cannot be done. However, by using a double decoder (Fig. 7) both words can be corrected by using Sq_i:

$$Sq_1 = L^*_1 \oplus R'_{-2} \oplus P'_{-7} \oplus Q'_1 \qquad (12)$$

$$Sq_4 = L'_4 \oplus R^*_1 \oplus P'_{-4} \oplus Q'_4 \qquad (13)$$

where the asterisks indicate erroneous word.

Supposing that L_1, R_1, and R_{-2} are erroneous, with two word errors included both in Sp_1 and Sq_1, error correction is nevertheless possible by using the double decoder as shown in the procedure below.

Step 1: Q Decoder

$$Sq_1 = L^*_1 \oplus R^*_{-2} \oplus P_{-7} \oplus Q'_1 \qquad (14)$$

$$Sq_4 = L'_4 \oplus R^*_1 \oplus P_{-4} \oplus Q'_4 \ . \qquad (15)$$

R^*_1 is corrected by Sq_4, but L^*_1 and R^*_{-2} are still erroneous.

Step 2: P Decoder

$$Sp_1 = L^*_1 \oplus R'_1 \oplus P'_1 \qquad (16)$$

$$Sp_{-2} = L'_{-2} \oplus R^*_{-2} \oplus P'_{-2} \ . \qquad (17)$$

Since R^*_1 is corrected in the preceding step, L^*_1 is corrected by Sp_1 and R^*_{-2} by Sp_{-2}.

Fig. 8 shows the quadruple decoder. Supposing that L'_1, R'_1, L'_{-2}, R'_{-2}, and Q'_{-2} are erroneous, quadruple correction in four steps is necessary.

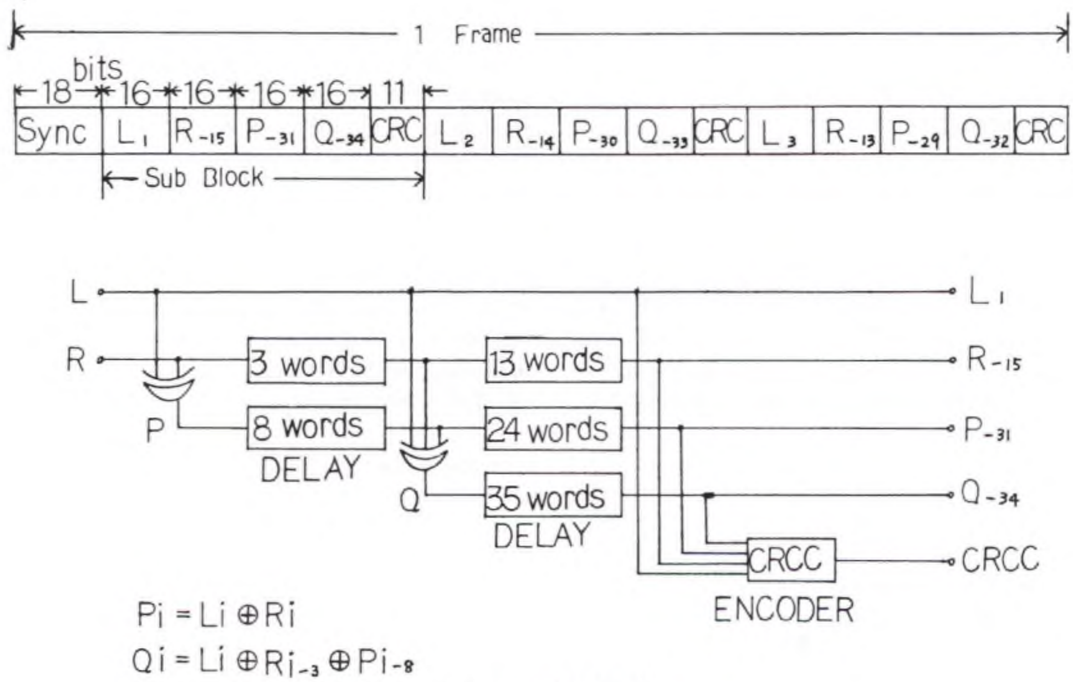


Fig. 5. Format and encoder.

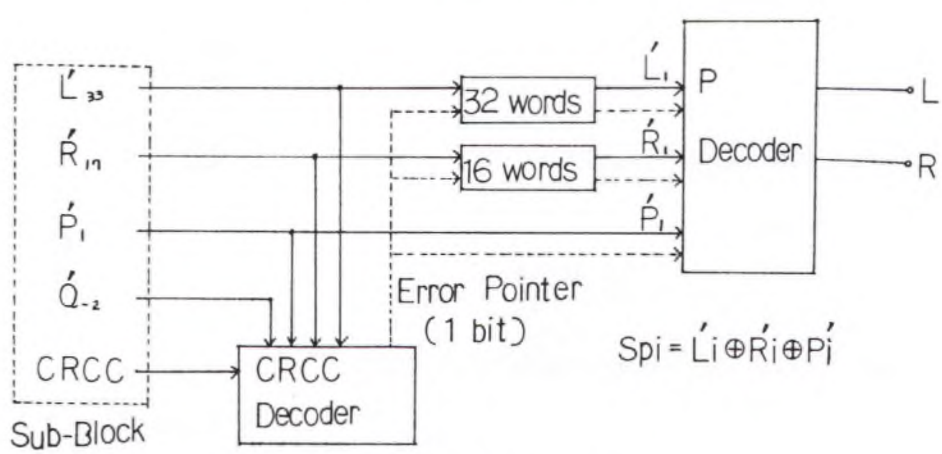
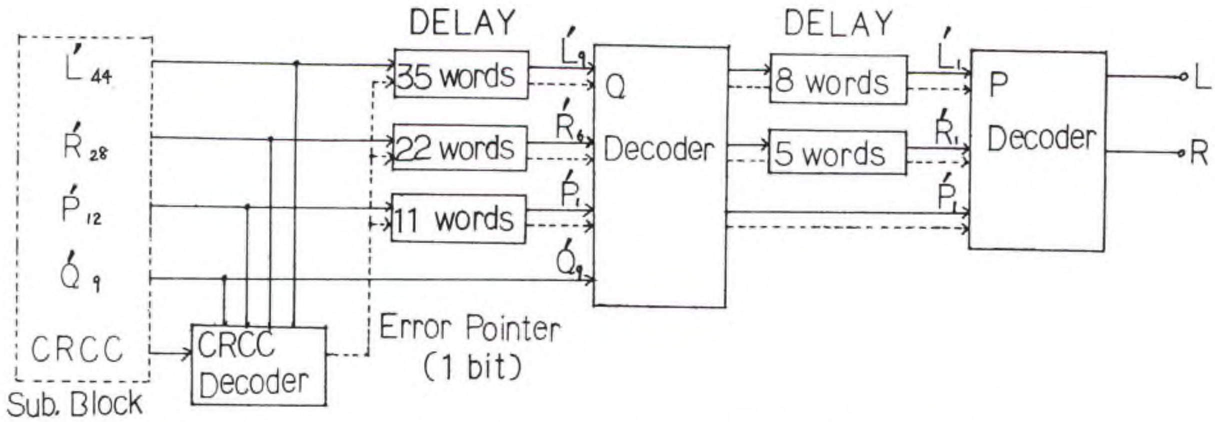


Fig. 6. Basic decoder (ignoring parity Q).



$$Sq_i = L_i \oplus R_{i-3} \oplus P_{i-8} \oplus Q_i, \quad Sp_i = L_i \oplus R_i \oplus P_i$$

Fig. 7. Double decoder.

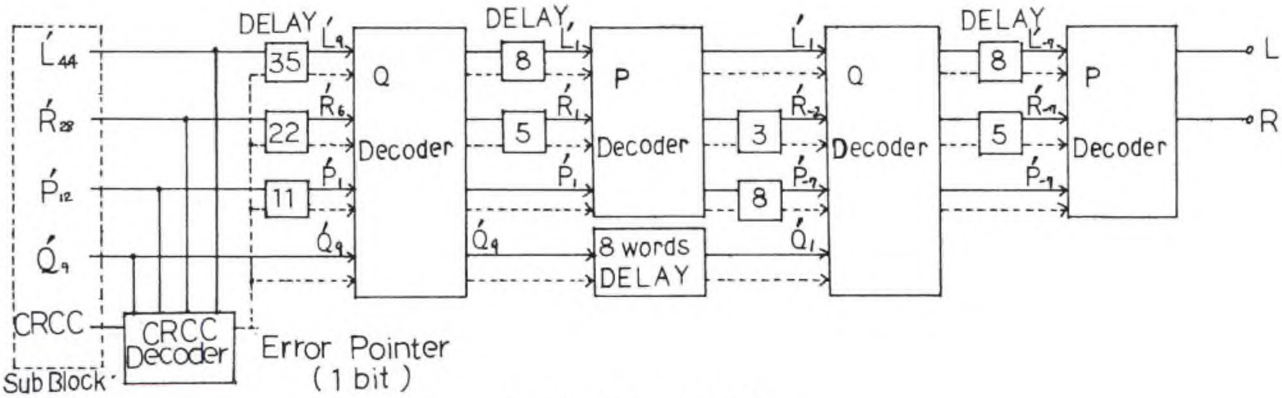


Fig. 8. Quadruple decoder.

Step 1: Q Decoder

$$Sq_{-2} = L^*_{-2} \oplus R'_{-5} \oplus P'_{-10} \oplus Q^*_{-2} \tag{18}$$

$$Sq_1 = L^*_1 \oplus R^*_{-2} \oplus P'_{-7} \oplus Q'_1 \tag{19}$$

$$Sq_4 = L'_4 \oplus R^*_1 \oplus P'_{-4} \oplus Q'_4 \,. \tag{20}$$

R^*_1 is corrected by Sq_4, but others cannot be corrected.

Step 2: P Decoder

$$Sp_1 = L^*_1 \oplus R'_1 \oplus P'_1 \tag{21}$$

$$Sp_{-2} = L^*_{-2} \oplus R^*_{-2} \oplus P'_{-2} \,. \tag{22}$$

Since R'_1 is corrected in the preceding step, L^*_1 can be corrected by Sp_1.

Step 3: Q Decoder

$$Sq_1 = L'_1 \oplus R^*_{-2} \oplus P'_{-7} \oplus Q_1 \tag{23}$$

$$Sq_{-2} = L^*_{-2} \oplus R'_{-5} \oplus P'_{-10} \oplus Q^*_{-2} \,. \tag{24}$$

R^*_{-2} is corrected by Sq_1.

Step 4: P Decoder

$$Sp_{-2} = L^*_{-2} \oplus R'_{-2} \oplus P'_{-2} \,. \tag{25}$$

L^*_{-2} is corrected by Sp_{-2}, and error correction is completed.

If only a double decoder (Fig. 7) is used for the above example, L_{-2} and R_{-2} remain uncorrected. This is one of the features of the cross interleave system. With each increase in the number of decoding steps, better correctability is obtained.

There is of course some probability of uncorrectable errors, however many steps of decoding are taken. One of these examples is shown below:

$$Sq_1 = L^*_1 \oplus R'_{-2} \oplus P'_{-7} \oplus Q^*_1 \tag{26}$$

$$Sq_4 = L'_4 \oplus R^*_1 \oplus P'_{-4} \oplus Q^*_4 \tag{27}$$

$$Sp_1 = L^*_1 \oplus R^*_1 \oplus P_1 \,. \tag{28}$$

In this case L^*_1, R^*_1, Q^*_1, and Q^*_4 are erroneous. All related syndromes, namely, Eqs. (26)–(28), include two erroneous words.

3.4 Crossword Decoding [10]–[13]

If an information word relates to two or more syndromes, crossword decoding is possible by comparing those syndrome patterns. In the above format three types of syndromes (Sp_i, Sq_i, and residuals of CRCC) can be utilized.

In the case shown by Eqs. (26)–(28), related residuals of CRCC, Sc_i, are as follows:

$$Sc_1 = \mathrm{Res}(L'_1, R'_{-15}, P'_{-31}, Q'_{-34}, C'_1) \tag{29}$$

$$Sc_{17} = \mathrm{Res}(L'_{17}, R'_1, P'_{-15}, Q'_{-18}, C'_{17}) \tag{30}$$

$$Sc_{36} = \mathrm{Res}(L'_{36}, R'_{20}, P'_{4}, Q'_{1}, C'_{36}) \quad (31)$$

$$Sc_{39} = \mathrm{Res}(L'_{39}, R'_{23}, P'_{7}, Q'_{4}, C'_{39}) \quad (32)$$

where Res ( ) means residual of CRCC, and C'_i is the received CRCC word.

In the case described, which is not correctable by the erasure method, Eqs. (26)–(32) are not zero. If the words L^*_1, R^*_1, Q^*_1, and Q^*_4 are really erroneous, they cannot be corrected by any means. But the error position in Eqs. (29)–(32) cannot be indicated and the following error patterns are also considered as uncorrectable:

i) $L^*_1, R^*_1, R^*_{20}, C^*_{39}$

ii) $L^*_1, L^*_{17}, P^*_4, Q^*_4$

If the word error is considered as random, real uncorrectable errors make up 1% of the errors not correctable by the above decoders, and 99% will be corrected by the crossword decoder. The principle of crossword decoding is to find the error location by comparing syndromes. For instance, in case i) the equation

$$Sp_1 = Sq_1 \oplus Sq_4 \quad (33)$$

will be satisfied, and in case ii)

$$Sq_1 = Sp_1 \, . \quad (34)$$

There is another method of crossword decoding, that is, to put the parity words into the CRCC decoder and, after appropriate shifting, compare with Eqs. (29)–(32) [13]. If the error is within an 11-bit burst, the error location in Eqs. (29)–(32) can be found by this method.

4 3PM CODE

Among the various classes of "run length limited code" 3PM (three-position modulation) [4] is selected because of its relatively simple hardware and fairly good efficiency. The principle of 3PM, shown in Table 2, is to convert the original data word of 3 bits into a 6-bit word, in which 1 means transition, and any 1's are always separated by two 0's. In other words, the minimum duration between transitions is $1.5L$, that is, 50% larger than that of MFM, where L is the length of the original data bit cell.

At the junction of the words another consideration is necessary. If word 8 comes after word 1, the transition pattern is

0 0 0 0 1 0   1 0 0 1 0 0
No. 1   No. 8

and there is a 1 0 1 pattern which violates the above law. In this case the 1 0 1 pattern is changed to a 0 1 0 pattern as follows:

0 0 0 0 **0 1**   **0** 0 0 1 0 0
No. 1   No. 8

In order to permit this nonlinear junction, the last transition in Table 2 is always 0, and coding always proceeds while the preceding and following words are watched carefully.

Table 2. 3PM

Number	Original Data Word			Transition Positions
1	0	0	0	0 0 0 0 1 0
2	0	0	1	0 0 0 1 0 0
3	0	1	0	0 1 0 0 0 0
4	0	1	1	0 1 0 0 1 0
5	1	0	0	0 0 1 0 0 0
6	1	0	1	1 0 0 0 0 0
7	1	1	0	1 0 0 0 1 0
8	1	1	1	1 0 0 1 0 0
	$\mid \leftarrow L \rightarrow \mid$			$\mid \leftarrow L \rightarrow \mid$

Table 3 shows a comparison between 3PM and MFM. The packing density of 3PM can be 50% higher than that of MFM, but the jitter margin is 50% worse, and the maximum transition is 100% longer. Therefore bit synchronization and the mechanisms of the player should be designed carefully.

Table 3. Comparison between 3PM and MFM.

		MFM	3PM
Duration between transition	Minimum	$L(1/2\ \lambda_{min})$	$1.5L(1/2\ \lambda_{min})$
	Maximum	$2L(\lambda_{min})$	$6L(2\ \lambda_{min})$
Jitter margin		$0.5L(1/4\ \lambda_{min})$	$0.5L(1/6\ \lambda_{min})$

5 CONCLUSION

A digital audio disk system realizing a playing time of 2½ hours on one side of a 303-mm disk is described. The redundancy of the error-correcting schemes is 64.2% including synchronizing bits, and the transmitting bit rate is 3.568536 Mb/s for two channels of 16-bit signals. But on account of 3PM coding, the maximum recording frequency is 1.189512 MHz, and the minimum wavelength at the inside of the disks (120 mm diameter) is 2.4 μm for a 450 r/min revolution. These values are decided tentatively for experimental systems. For the systems to be commercialized the redundancy and the minimum wavelength must be reduced, and probably can be with highly controlled processes. Longer playing time or smaller sized disks will also be possible.

Another great problem for commercialization is compatibility between digital audio and video disk players. The system described in this paper has that compatibility. However, consequently an extremely long playing time is achieved, which might be unnecessary from a software point of view. The final standard should be established by mutual studies of various software and hardware makers.

6 ACKNOWLEDGMENT

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A New PCM Audio Disk Pickup Employing a Laser Diode*

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An improved semiconductor laser pickup for an optical PCM audio disk player is described. In order to raise the servo gain, a reduction in the size and weight of the optical system was tried through the use of a newly designed lens and a revised optical path. A unique simplified two-dimensional servo actuator without any spring for suspension was developed.

0 INTRODUCTION

Simplification and miniaturization are very important factors for the optical system of the laser PCM audio disk player to put the player into mass production and mass utilization. The conventional He–Ne laser pickup uses a rather complex optical system, comprising many optical parts such as mirrors, grating, galvanometer, etc., which are aligned in the optical path.

We have proposed [1] a new semiconductor laser pickup to overcome these disadvantages. The whole optical system, which was assembled in a tube of 38-mm length and 10-g weight, although substantially smaller and lighter than the conventional systems, was not yet quite satisfactory. The semiconductor laser weighing only 0.2 g suggested that an optical system of about 2 g could be developed. Besides, we expected that this lightweight optical system might permit the introduction of a new actuator structure without spring suspension.

Here we introduce an improved optical pickup for the PCM audio disk player.

1 DESIGN OF PICKUP

The targets of the development were as follows:

1) The optical system should be less than 20 mm in length and 2 g in weight.

2) The two-dimensional servo actuator should be simple structured, without any spring suspension.

3) The servo actuator should be separated from the optical system and easily mounted by screws.

Item 3) expects that the optical system and the actuator can be modified separately.

* Presented at the 64th Convention of the Audio Engineering Society, New York, 1979, Nov. 2–5.

2 RESULT

2.1 Optical System

Fig. 1 shows a sectional view of the structure of the developed pickup. Its external appearance is shown in Fig.

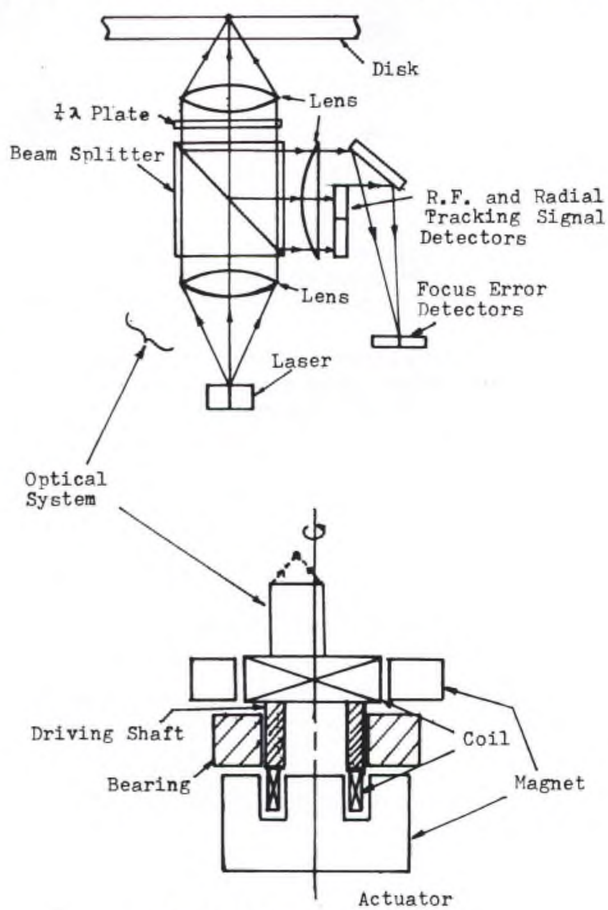

Fig. 1. Illustration of pickup.

2. The radiated beam from the semiconductor laser is collimated by the collimating lens of the numerical aperture (NA = 0.25) and focused on the disk by the objective lens (NA = 0.5). A part of the reflected beam, separated from the incident path by a beam splitter, is passed to the split photodetector placed at the far-field position in the reflected beam path, and generates both a radial tracking-error signal and a radio frequency (RF) signal. The rest of the reflected beam is focused on the other split photodetector to generate the focusing-error signal.

Ultra-lightweight lenses were specially developed for this purpose. Both the collimating and the focusing lenses are 4 mm in diameter, 3 mm in length, and 0.1 g in weight. The astigmatism was not corrected to make the lenses as light as possible.

The dimensions of the developed optical system are 15 by 7.5 by 7.5 mm³ and 1.5 g in weight. The lightweight lens and the direct mounting of the semiconductor laser have made possible the lightweight optical system.

2.2 Actuator

As shown in Fig. 1, the two-dimensional actuator has its driving shaft, with one end connected to the optical system, held by a bearing with a small friction coefficient, and allowing translation and rotation. The rotation of the driving shaft gives a radial tracking function since, as shown in Fig. 1, the optical system is mounted on the actuator with an offset from the rotational center of the shaft. Both the translational and the rotational movements are generated by respective moving coils, to which the error signals are fed.

We have thus realized a remarkably simplified actuator employing the two-dimensional freedom of the movements. Fig. 3 shows the gain versus frequency characteristics of this actuator.

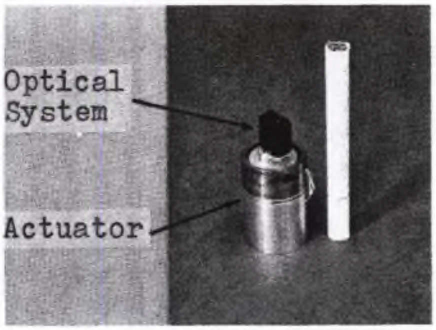

Fig. 2. External appearance of pickup.

2.3 Servo System

It is not difficult for this actuator to attain a servo gain as high as 60 dB. However, the following improvement was needed to increase the radial tracking performance:

1) Improvement of the surface smoothness of the driving shaft,

2) Selection of a less frictional bearing material (Teflon showed the best result).

As a result, a radial tracking accuracy within $\pm 0.1\ \mu m$ was attained. A typical reproduced RF signal is shown in Fig. 4.

3 DISK PLAYER

Fig. 5 shows a block diagram of the total PCM audio disk

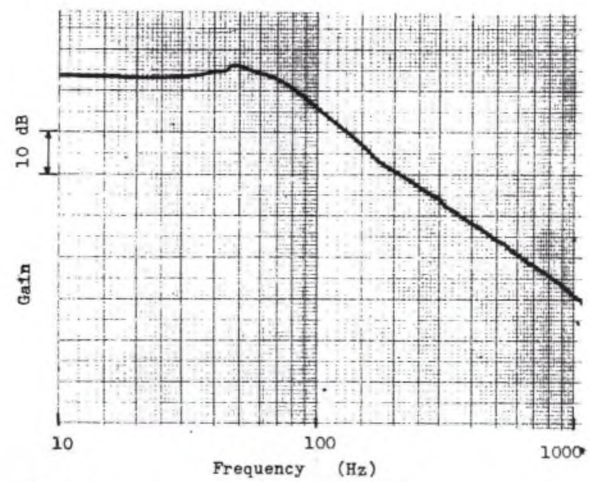

Fig. 3. Gain versus frequency characteristics of the focusing actuator.

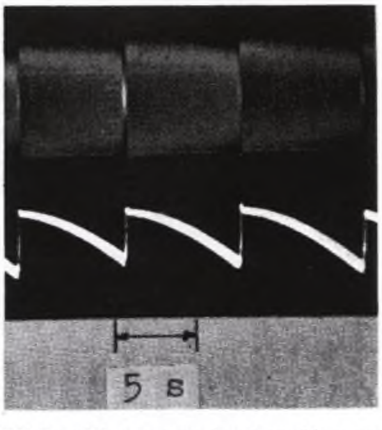

Fig. 4. RF signal (upper trace) and tracking-error signal.

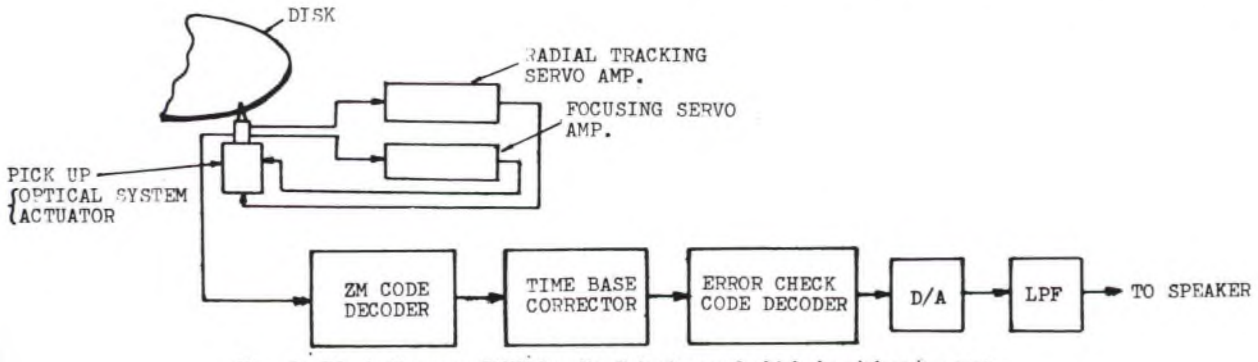

Fig. 5. Block diagram of PCM audio disk player of which the pickup is a part.

player system using the newly developed pickup. The system is composed of player, PCM processor, and retrieval part (Fig. 6). We also tried to design a vertically revolving disk player system which is vertically thinned in contrast to the conventionally horizontally revolving disk player.

Fig. 6. Outer look of player.

4 CONCLUSION

The newly developed pickup which employs a semiconductor laser satisfies all the requirements for PCM audio disk reproduction. It was designed so that the optical system could be easily detached to facilitate a separate modification. The newly developed two-dimensional servo actuator has produced superior dynamic responses owing to its springless structure, and it has shown a promise for a cost reduction of the pickup for mass production.

5 ACKNOWLEDGMENT

The authors would like to thank all members of the project for their contributions.

6 REFERENCES

[1] K. Okada, W. Susaki, H. Kondo, K. Kime, T. Miyazawa, and T. Sato, "A Compact PCM Audio Disk Pickup Employing Semiconductor Laser," presented at the 61st Convention of the Audio Engineering Society, New York, 1978 November 3–8, preprint no. 1409.

THE AUTHORS

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T. Kubo

T. Sato

Kazuo Okada was born in Ehime Prefecture, Japan, in 1937. He was graduated from the University of Osaka City, Japan, in 1962.

From 1962 to 1972 he worked at the Central Research Laboratory of Mitsubishi Electric Corp., concentrating on laser optics, nonlinear optics and automatic meter reading.

Since 1973 he has been with Products Development Laboratory of Mitsubishi Electric where his activities primarily involve optical pickup systems.

In 1975 he received a Ph.D. degree from the University of Osaka City.

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•

Takahiro Kubo was born in 1935 in Kobe, Japan. He received a B.S. degree in 1961, and Ph.D. in 1974, both from Osaka University in Japan.

In 1961, he joined the Central Research Laboratory, Mitsubishi Electric Corp., Amagasaki, Japan, where he was engaged in the research and development of the optical pumping magnetometer and the research of holography. He is now a leader of optical disk group at the Products Development Laboratory of Mitsubishi Electric Corp.

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•

Wataru Susaki was born in Ehime Prefecture, Japan, in 1938. He received a B.S. degree from Kyoto University, Kyoto, Japan, in 1961, and a Ph.D. degree from the Tokyo Institute of Technology, in Tokyo, in 1975. He joined the Mitsubishi Electric Corporation in 1961, where he was engaged in the development of high-frequency GaAs microwave diodes at the Central Research Laboratory between 1962 and 1967. From 1968 to 1974, he worked in the development of light-emitting devices at the Kitaitami Works. Since 1975, he has been leading the Semiconductor Laser Group of the Semiconductor Laboratory, and working in the research and development of laser diode and related devices.

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•

Takaharu Sato was born in Nagasaki, Japan, in 1933. He received a B.S. degree in communication engineering from the University of Tohoku, Miyagi, in 1957.

In the same year, he joined Mitsubishi Electric Corp., Japan, starting his career with the design and development of color television at its Communication Equipment Works in Hyogo. From 1971, he worked on the development of video tape recorders, a PCM audio tape recorder and PCM laser sound disk at the Products Development Laboratory. He is now engaged in the design and development of hi-fi audio equipment as a manager of engineering development at the Koriyama Works.

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The MD (Minidisk) System: A Contribution to the Digital Audio Disk Standard*

KLAUS WELLAND

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AND

HORST REDLICH

TELDEC, Berlin, Germany

Translation by Stephen F. Temmer

As a contribution to the discussion on standardization, a digital audio disk is described in detail. It combines the advantages of modern digital techniques with the simplicity of further developments in conventional disk manufacturing techniques. The disk is contained, and remains during replay, in a protective cassette, which facilitates handling and is completely closed when not attached to the player.

0 INTRODUCTION

It is surely not an accident that the possibilities of using large-scale integration (LSI) in the record industry, and possibly creating a significant change thereby, is being discussed during the LSI decade. But it certainly *is* a coincidence that this comes virtually on the 100th anniversary of the phonograph record. The development which our record technology has experienced during that time has led to enormous improvements as far as fidelity, playing time, price reduction, and distribution are concerned, without really altering the original analog methods of recording and playback. But in spite of these admirable advances, we appear to have reached the upper limit of quality attainable in this technology. Due to the handling of records, the impulse type of ticks and pops cannot entirely be avoided. The signal-to-noise ratio, especially for the low frequencies (rumble), is unsatisfactory. The multiple tape copies necessary during production of the program produce modulation noise as well as cumulative time base and compression effects which adversely affect the transparency of the recording (Fig. 1). That is why direct-to-disk recordings, which circumvent in their production the tape copies of the first and second generations, enjoy increasing popularity. Because these direct recordings cannot make use of the usual artistic editing and because of their manufacturing limitations on quantity, their production costs remain high, and their use is not possible for complex works such as symphonies and complete operas.

1 DIGITAL TECHNOLOGY PROVIDES ANSWERS

From the communications field we know of methods that virtually eliminate disturbing influences by converting analog signals into digital language and later, after transmission or storage, reconvert these into the original analog signals. This is done by measuring the instant amplitudes of sound waves at very short intervals and by assigning to each of these values a prearranged code. These data in the form of pulses are transmitted or stored. Any errors which may appear are corrected, and the data are finally returned to

* Presented at the 64th Convention of the Audio Engineering Society, New York, 1979 November 2–5.

their original analog form. This is done through so-called pulse code modulation (PCM).

Very-high-quality sound signals require a great amount of data per unit of time, that is, the transmission bandwidth must be made greater than for normal analog technology, and with it the storage density on the disk record as well. This results, of necessity, in a digital record which is totally incompatible with traditional phonograph records.

This incompatibility is more than compensated for by the unusual advantages of such a disk:

1) Much greater signal-to-noise ratio, great dynamic range
2) Elimination of pulse-type disturbances through error correction
3) Practically no crosstalk
4) Negligible distortion
5) Time-base errors are corrected
6) Virtually no quality reduction when several generations are created during production.

The technical prerequisites for making this technology available to entertainment electronics are known today:

1) Codes that almost fully correct for errors are available.
2) The enormous strides, which large-scale integration (LSI) has made in recent times, will soon provide us with digital-to-analog converters of very high precision.
3) Records featuring high storage capacity and playback methods with extremely wide bandwidth are available from video disk technology.

2 THE PROBLEM OF STANDARDIZATION

As is the case with the phonograph record throughout the world today, it would be desirable to evolve a worldwide standard for the PCM disk before such a record is marketed. Such a standard must assure decades of excellent quality. It must take into consideration both the total electronic processing to be used as well as the high-density storage system (disk/playback unit). There is a certain interdependence between these two.

When choosing a code, the following must be considered:

1) High, but not exaggerated, demands on quality and fidelity
2) Compatibility with the code employed in the studio
3) Possible compatibility with codes of other future transmission channels (for example, PCM satellite transmission)
4) Simple but adequate error-correction system, keeping equipment costs reasonable.

The high-density storage system (disk/playback unit) should fulfill the following demands:

1) Simple and serviceable replication method for the disk
2) Compact in size, small diameter
3) Double-faced disk
4) Playing time per side at least 45 min
5) Search capability for individual selections
6) No excessive demands on accuracy
7) Simple handling of both record and playback unit
8) Secure protection against damage to record
9) Long life expectancy for both disk and player.

In view of these problems of world standardization, numerous national and international standards committees have been organized, with members including virtually every major record and high-fidelity manufacturer. So far these efforts have borne some fruit. The original demand for compatibility between a video disk and such an audio record is no longer in the foreground. However, agreement must yet be achieved for the code as far as quantization, sampling rate, and bit rate are concerned, and for the player, the type of high-density storage, as well as the disk diameter.

It is important to remember that the entertainment electronics field has two groups of high-density storage systems available (Table 1); those systems that play back without record contact, and those in which the pickup touches the disk. The second group may be divided into those with piezoelectric or electrostatic pickups riding in a groove and electrostatic ones without a groove.

It is conceivable that the systems that have a groove structure can be made compatible. It is possible to use a pressure transducer (piezoelectric) for both PVC pressed disks and those made of a conductive material intended for electrostatic playback. The storage densities achieved are comparable.

3 TELEFUNKEN/TELDEC—A SUGGESTED PCM AUDIO DISK

Telefunken and Teldec suggest the following standard for a digital audio disk based on prior work in the video disk field and the preceding descriptive analysis.

1) For the high-density storage system:

a) A conventionally pressed PVC record with trapezoidally shaped, vertically recorded information grooves on both sides, protected against damage by a cassette. The record is to remain inside the cassette during playback.

b) Playback using a piezoelectric pressure transducer. This type of pickup has recently been greatly improved as far as its groove tracking behavior and service life are concerned. The latest of these transducers can play extremely short wavelengths down to 0.5 μm.

2) For the encoding parameters:

a) *Quantization:* 14-bit linear. The signal-to-noise ratio achievable with 14 bits (86 dB) is fully sufficient for a program dynamic range optimized for normal living rooms. The much greater cost of a 16-bit system appears justified only for recording use with its much greater overload reserve demand.

b) *Sampling rate:* 48 kHz. This frequency is compatible with the rate suggested by the Technical Commission of the Federal Association of the German Phonograph Industry and the RIEE, and expands on the suggestion of the CMTT for digital transmission via cable, microwave links, and satellites. Simple transcoding is possible.

c) *Transmission code:* biphase. The selected piezoelectric pressure transducing system with its high resolution down to the smallest wavelengths permits a biphase code.

Even though this requires twice the bandwidth when compared with the Miller code, its demodulation is accurate and simple.

3) Additional technical data for the disk:

a) Playing time	2 × 60 min
b) Number of transmission channels	4
c) Addressing	Automatic search
d) Diameter	135 mm (5.3 in)
e) Groove spacing	1.66 μm $\triangleq$ 0.065 mil $\triangleq$ 600 lines/mm $\triangleq$ 15 240 lines/in
f) Playback groove velocity	Constant: 1.89 m/s $\triangleq$ 6.2 ft/s $\triangleq$ 4.23 mi/h
g) Revolutions per minute	Between 278 and 695
h) Smallest wavelength	0.61 μm $\triangleq$ 0.024 mil
i) Transmission rate	3.072 Mbit/s
j) Storage density	$\approx$ 1000 kbit/mm^2 = 645.16 Mbit/in^2

4) The playback unit concept for these disks permits the following uses:

a) User selectable stereo or binaural reproduction.

b) Separately recorded "ambiance" to permit the user free choice of mixing, depending on whether loudspeakers are used in connection with room acoustics or binaural headphone listening.

c) Separate recording of vocal or solo instrument to permit a choice of balance during playback or to allow the user to add his own solo part (so-called add-a-part records).

d) Spoken recording of album liner notes.

e) Using the automatic addressing feature, 2 hours of stereo per side for extra-long works such as oratorios, operas, and concertos.

f) Quadraphonic sound with absolute channel separation.

4 RECORDING AND MANUFACTURING THE DISK

It was the purpose of our research to bring the tape-to-disk transfer, plating, and pressing processes for this high-density record on the one hand, and the playback unit technology on the other, into agreement to permit use of today's phonograph record technology while fulfilling the increased mechanical tolerance demands which a digital disk poses.

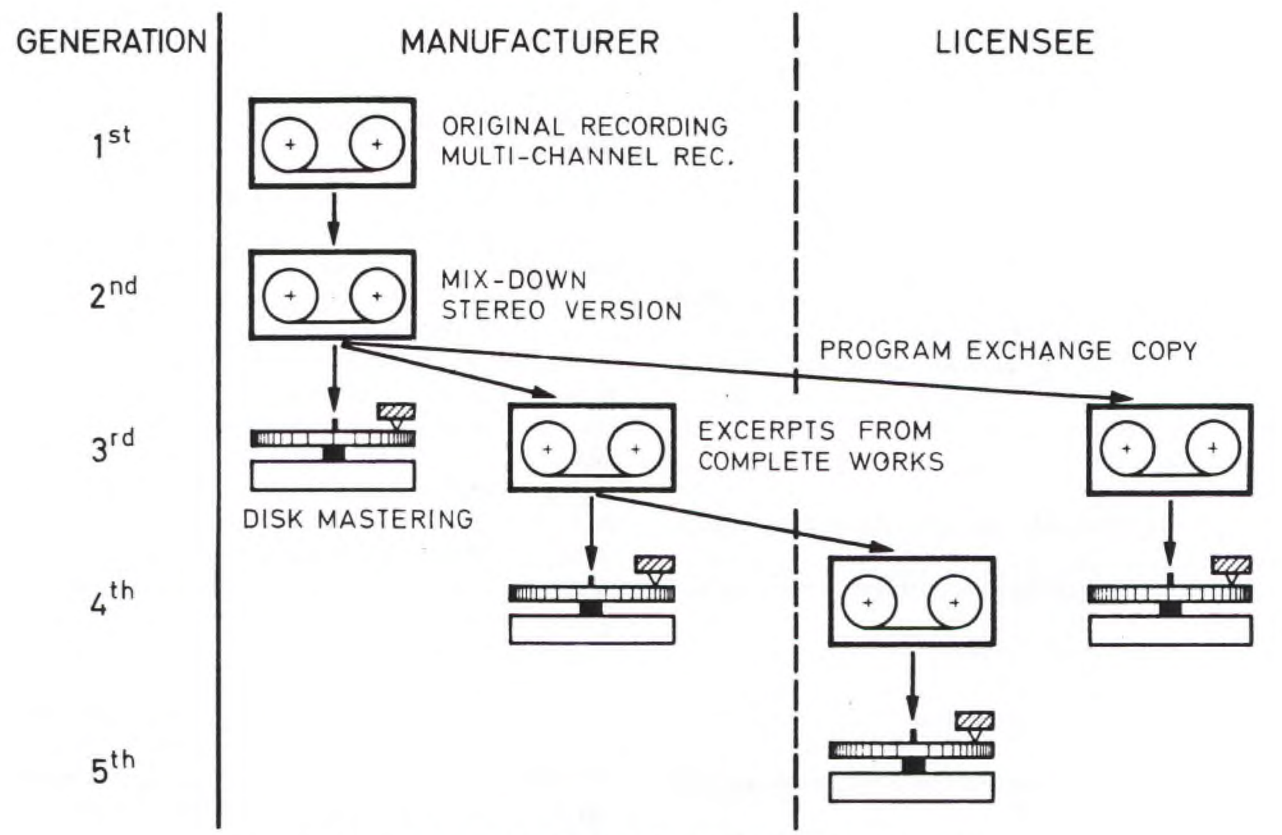

Fig. 1. Tape generation flow chart from the studio to the disk master.

Table 1. High-density disk systems.

	Playback Systems with Mechanical Pickup Contact			Playback Systems without Mechanical Pickup Contact	
Information track	Groove	Groove	Grooveless	Grooveless	Grooveless
Record compound	PVC	PVC + carbon	PVC + carbon	PVC	PVC, metallized surface
Pickup system	Piezoelectric	Electrostatic	Electrostatic	Photoelectric	Photoelectric

The recording of the digital information from tape is done mechanically in real time onto a metal blank. This original already possesses the centering and profile of the finished record, obviating the need for additional work on the stamper which might adversely affect tolerances. The direct cutting of the mother has a distinct advantage over optical systems by eliminating seven additional mechanical steps which may increase the chance of raising the number of faults significantly (Table 2 and Fig. 2).

This new PCM disk uses practically the same methods and materials as are used today in the manufacture of the traditional record. Additional investments are not required. The comparatively low number of faults which such a disk has is one of the reasons why this system, with a data redundancy of only 30% for error correction, suffices in spite of the tremendous storage density.

Table 2. Manufacturing process of high-density disk system.

		Piezoelectric System	Optical System
Matrix Process	1	—	Photoresist coating
	2	—	Exposure under clean-room conditions
	3	—	Developing
	4	—	Silvering
	5	—	Metal master
	6	Direct cutting into metal surface	Electroplated mother
	7	Stamper	Stamper
Record Manufacturing	8	Pressing	Pressing
	9	—	Metallizing
	10	—	Sealing

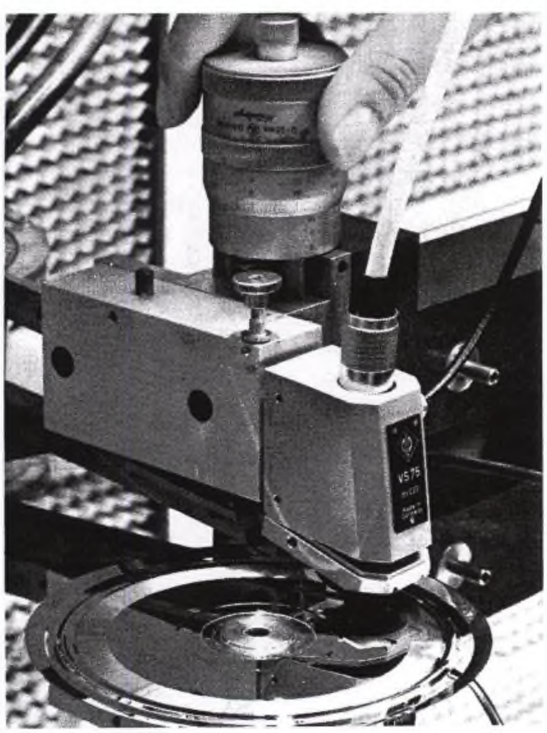

Fig. 2. Direct cutting of the metal mother.

5 THE MD (MINIDISK) SYSTEM—A CASSETTE RECORD

All high-density systems using record contact playback use surface-recorded information. The disk surface protection as well as the simplified structural disk/player concepts are combined in the MD (minidisk) system.

The disk is protected against damage by a fully enclosed cassette, which is only partially opened for playback within the player itself. Playback is from below the disk. Within the centering area, recognizable from its pressed conical shape, the disk contains a ferromagnetic material. During playback a magnet pulls the record into the centering ring and holds it there. The playback from below and the centering system, aside from precise centering, have the advantage that in the record press it is no longer necessary to have top and bottom stampers in accurate alignment with one another. The disk is, of course, free to move within the cassette (Fig. 3).

The cassette replaces the traditional albums and permits a compact collection when compared with traditional long-playing records of identical playing time (Fig. 4).

6 THE PCM DISK WILL EVENTUALLY GAIN THE UPPER HAND

The results of our research and development show that

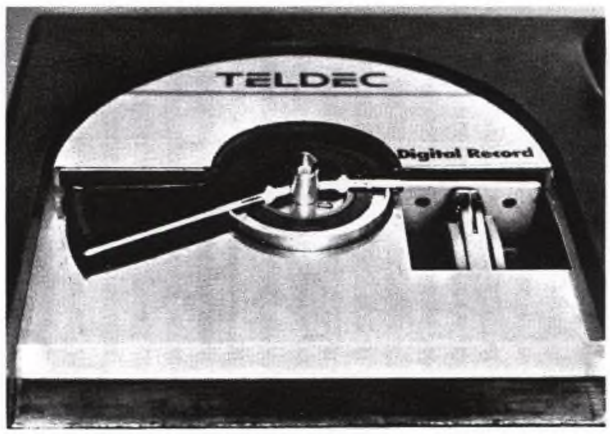

Fig. 3. Cross section of a possible MD player showing the pickup underneath the disk at right.

Fig. 4. The MD cassette surrounded by the four standard long-playing disks whose total music time it accommodates.

today's record and all of its parameters may be significantly improved upon using available and near future sound storage technologies involving high-density PCM systems. Besides the handling ease of this disk, the MD system playback unit concept permits problem-free operation, automatic selection search, and a very compact size as well.

In view of the incompatibility with the worldwide conventional records and playback units, it is a foregone conclusion, provided we get a worldwide standard for the PCM disk, that we will see a gradual changeover to the new disk.

THE AUTHORS

Klaus Welland was born in Witten, Germany, in 1928. He studied communication engineering at the Technical University in Darmstadt until 1954. He then worked as assistant at the Technical University in Berlin until 1960. He graduated after completing a thesis on "Electronic reversal of color negatives."

Since 1960 he has been working for Telefunken, first in the TV-Development-Department. He became director of R & D of Telefunken Consumerelectronics in 1969. He has been a member of the Executive Board of Telefunken GmbH since 1975.

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Horst Redlich was born in Berlin, Germany, in 1921, where he studied precision mechanical engineering. In 1947 he joined AEG as a development engineer and worked mainly in the field of magnetic sound recording for studio purposes. From 1950 to 1951 he managed the development work at the UFA film company which resulted in the first magnetic sound track films (35 mm).

In 1951 Redlich became chief engineer of the TELDEC (Telefunken-Decca) record company. Under his management TELDEC produced the first stereophonic LP records in Europe in 1955. In the course of the ensuing years he developed the improved Royal Sound stereo method. Since 1959 he has been technical director of TELDEC, and during recent years has devoted his time to the development of high-density-storage technology, which led to the first public presentation of the TED Video Disc in 1970. He was honored for this work with the Bundesverdienstkreuz (Federal Republic Medal of Honor (first class) in 1972. He is a fellow of the AES and the recipient of its Silver Medal Award (1977).

standards and invention

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AES Standard Playback Curve

BASED ON THE PREMISE that the proper approach to the problem of equalizing disc recordings and transcriptions is to standardize on a *playback* curve and to let the recording engineers make their records however they see fit, knowing that they must sound properly balanced when played on this standard reproducing characteristic, the Audio Engineering Society announces the adoption of such a curve. This announcement follows action of the Board of Governors approving the report of the Society's Standards Committee consisting of: Gordon Edwards, chairman; S. E. Sorensen, vice chairman; James Bayless, Harry Bryant, and Russell Hanson, members of the Western Division; and Theodore Lindenberg, N. C. Pickering, A. A. Pulley, and Ralph Schlegel, members of the Eastern Division. Robert Liesenberg served as alternate to Mr. Sorensen.

The standard curve, shown in *Fig.* 1, is represented by the values in Table 1.

The decision to specify a standard playback response characteristic instead of a recording characteristic was deliberate on the part of the Standards Committee. This course was chosen because of the impossible task of achieving a universal recorded characteristic compatible with all individual recording conditions and systems.

Reference to the tabulation will indicate that all points on the curve are related to 1000 cps. This reference point has been used as a standard for many years, making it evident that the maintenance and calibration of equipment would be expedited by retention of this frequency as a reference point. Furthermore, the slope of the curve at this point is sufficiently flat so that an error of 10 per cent in frequency will produce a deficiency of not more than 0.5 db.

TABLE 1

Frequency	db	Frequency	db
30	+22.5	1500	− 1.5
40	+20	2000	− 2.2
50	+18	2500	− 3
70	+15	3000	− 4
100	+12	4000	− 5.5
150	+ 8.5	5000	− 6.7
200	+ 6.5	6000	− 8
300	+ 4.5	7000	− 9
400	+ 3	8000	−10
500	+ 2	9000	−11
800	+ 0.5	10000	−12
		12000	−13.5
1000 (ref.)	±0	15000	−15.5

Permissible tolerance ±2 db

The majority of engineers active in the recording field have felt for some time that the degree of high-frequency emphasis prescribed by the NAB transcription characteristic is excessive. The trend in modern microphones and amplifiers to a wider frequency range, approaching 15,000 cps, and the use of acoustically brighter studios have made this problem much more difficult. With this extended range, the acceleration of the reproducing stylus becomes a limiting factor. Consequently, it was deemed necessary to restrict the degree of high-frequency rise used in recording. This was accomplished by making the reproducing characteristic roll off only 12 db at 10,000 cps—approximately 3 db below the NAB specification—and continuing the response out to 15,000 cps. By doing this, the high-frequency situation has been alleviated somewhat. Since microphone and studio characteristics must be considered by the recording engineer, it is required that the sum of the electrical rise in the recording equipment and the acoustical rise in the microphone must not exceed the values shown by the reciprocal of the reproducing characteristic, unless it is intended to make the high end overbrilliant.

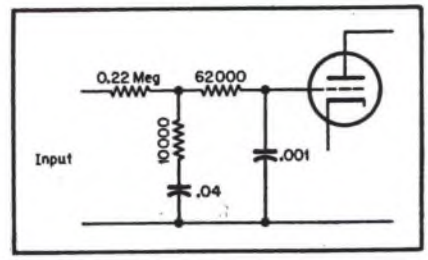


Fig. 2. High impedance network to provide standard playback curve in grid circuit of amplifier stage.

The low-frequency characteristic was chosen to fall somewhere in the middle of the numerous low-frequency curves now in use. It is felt that the turnover frequency is low enough to keep rumble down to reasonable levels, and high enough to avoid excessive amplitude and intermodulation at low frequencies. It will be noted that no "shelving" of the characteristic at low frequencies is recommended. Again, if the recording engineer desires for some reason to have a "bassy" sound, he can easily

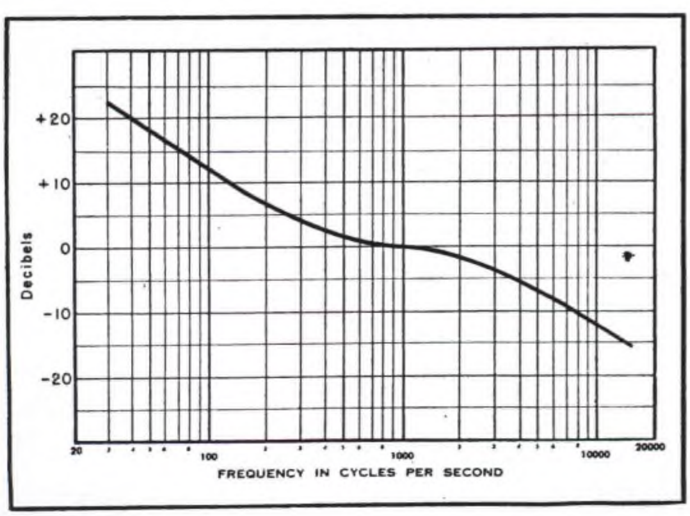


Fig. 1. Newly adopted standard playback curve.

accomplish this by making his recording characteristic tip up at the low end; conversely, he can "thin out" the sound by the opposite procedure.

The shaping of this curve can be duplicated on a flat playback system with two sections of RC equalization, as shown in *Fig. 2* which is one possible arrangement for use in an amplifier circuit. Both of the straight portions of the curve are slopes of 6 db per octave. The intersections of these slopes with the reference axis occur at 400 cps and at 2500 cps. At these points the response is 3 db away from the reference level. Within a tolerance of ±2 db it will be seen that all turnovers between 325 and 500 cps will fall in the area covered.

The adopted response curve (within its tolerances) is sufficiently parallel to the NAB response curve so that no problem will be encountered in the reproduction of NAB recording.

It is to be expected that the characteristic at the low-frequency end will stop rising at the 6 db/octave rate at some frequency determined by the range of the reproducing equipment. It is felt that first-class wide-range equipment will continue to 30 cps within the specified tolerance and then flatten off as rapidly as possible. Where equipment has a higher low-frequency cutoff, it is recommended that the reproducing characteristic follow the curve to its lower limit and then drop off as rapidly as possible.

On the high-frequency end, it is recommended that the reproducing characteristic be followed to the desired upper frequency cutoff, above which point the response should drop off smoothly and rapidly. In wide-range equipment it is expected that the playback characteristic will follow the curve to 15,000 cps within the tolerance specified, and then drop off rapidly above this point.

Typical Equalizing Networks

The equalizers of Fig. 3 are shown in order to facilitate the construction of these networks for use in professional installations. The Playback De-Emphasis Network is designed to give the proper roll-off characteristic in circuits of the impedances shown. If used with existing equalizers in playback circuits, the high-frequency response should be set on "flat" to obtain the proper curve.

The Recording Pre-Emphasis Network is designed for insertion in circuits of the indicated impedances ahead of the main recording amplifier. It is presumed that modifications will be made in the cutter network to obtain the desired low-frequency response. For information on the methods of adjusting these circuits, it is suggested that the engineer make inquiry from the cutter manufacturer.

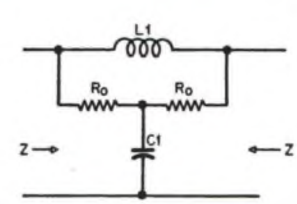
While most installations will already have some form of low-frequency equalizer for reproduction of existing types of records and transcriptions, it is possible that an entirely separate network will be required. The Playback Low-Frequency Boost Equalizer is designed to give a turnover frequency of 400 cps, with a total insertion loss (at 1000 cps) of 20 db. The half-loss point is 125 cps, and this equalizer will result in a slight decrease in response over the projected curve below about 70 cps. However, it falls within the limits down to 45 cps, and the decrease in response below that frequency may be an aid in reducing rumble.

All of these networks are designed to have constant impedance characteristics, and since they are symmetrical they may be used without regard to input or output connections. All networks shown are unbalanced, and usual transposition methods can be used to convert them to balanced networks if such are required in any particular installation.

Conclusion

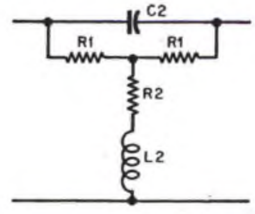
The new standard playback curve, if accepted by the Recording Industry, can achieve at long last a common platform for the reproduction of all recordings regardless of speed, groove dimensions, or manufacturer.

PLAYBACK DE-EMPHASIS NETWORK



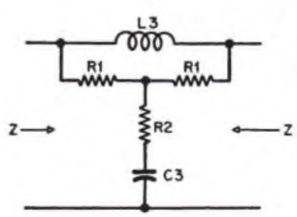
Z (ohms)	L1 (mh)	C1 (μf)	R_0 (ohms)
150	9.55	0.408	150
250	15.9	0.255	250
500	31.8	0.127	500
600	38.2	0.102	600

RECORDING PRE-EMPHASIS NETWORK



Z (ohms)	C2 (μf)	L2 (mh)	R1 (ohms)	R2 (ohms)
150	.0472	1.045	123	30
250	.0284	1.74	204	50
500	.0142	3.49	408	101
600	.0118	4.19	491	121

PLAYBACK L-F BOOST EQUALIZER



Z (ohms)	L3 (Hy)	C3 (μf)	R1 (ohms)	R2 (ohms)
150	0.545	24.08	123	30
250	0.910	14.44	204	50
500	1.81	7.22	408	101
600	2.18	6.02	491	121

Fig. 3. Constant impedance networks suitable for line impedance indicated.

Editor's Note. This British patent, a portion of which appears in the following pages, is reproduced with the kind permission of the Controller of Her Britannic Majesty's Stationery Office. It is of historic importance in the development of stereophony. I have felt that since this issue of the Journal is substantially devoted to this subject its appearance is timely and appropriate. When it is realized that many of the ideas, psychoacoustic, mechanical, and electrical, set forth in this document of 1931 are only now gaining wide, popular currency, one may well reflect on the magnitude of the economic forces which control the viability of inventions.

British Patent Specification 394,325

Alan Dower Blumlein

Application Date: Dec. 14, 1931 Complete Accepted: June 14, 1933.

COMPLETE SPECIFICATION

Improvements in and relating to Sound-transmission, Sound-recording and Sound-reproducing Systems.

The fundamental object of the invention is to provide a sound recording, reproducing and/or transmission system whereby there is conveyed to the listener a realistic impression that the intelligence is being communicated to him over two acoustic paths in the same manner as he experiences in listening to everyday acoustic intercourse and this object embraces also the idea of conveying to the listener a true directional impression and thus, in the case in which the sound is associated with picture effects improving the illusion that the sound is coming, and is only coming, from the artist or other sound source presented to the eye.

The invention is not, however, limited to use in connection with picture effects, but may, for example, be used for improving the qualities of public address, telephone or radio transmission systems, or for improving the quality of sound recordings. When recording music considerable trouble is experienced with the unpleasant effects produced by echoes which in the normal way would not be noticed by anyone listening in the room in which the performance is taking place. An observer in the room is listening with two ears, so that echoes reach him with the directional significance which he associates with the music performed in such a room. He therefore discounts these echoes and psychologically focuses his attention on the source of the sound. When the music is reproduced through a single channel the echoes arrive from the same direction as the direct sound so that confusion results. It is a subsidiary object of this invention so to give directional significance to the sounds that when reproduced the echoes are perceived as such.

In order that the physical basis of the invention can be appreciated and the stages of its development understood, known and established facts concerning the physical relations between sound sources, sound waves emitted thereby, and the human ears will be briefly summarized.

Human ability to determine the direction from which sound arrives is due to binaural hearing, the brain being able to detect differences between sounds received by the two ears from the same source and thus to determine angular directions from which various sounds arrive. This function is well known and has been employed to considerable extent for example in subaqueous directional detection in which two microphones are connected by headphones, one to each ear of an observer, the two channels between the microphones and the two ears being kept entirely separate.

With two microphones correctly spaced and the two channels kept entirely separate e.g. by using headphones it is known that this directional effect can also be obtained for example in a studio. If, however, the channels are not kept separate (as, for example is the case in previously proposed arrangements for recording and/or reproducing sound, in which sounds picked up by a plurality of pressure microphones are led to loud speakers which take the place of the headphones) the effect is almost entirely lost and such systems have therefore not come into common use since they are quite unsatisfactory for the purpose. The present invention contemplates controlling the sound, emitted for example by such loud speakers, in such a way that the directional effect will be retained.

The operation of the ears in determining the direction of a sound source is not yet fully known but it is fairly well established that the main factors having effect are phase differences and intensity differences between the sounds reaching the two ears, the influence which each of these has depending upon the frequency of the sounds emitted. For low frequency sound waves there is little or no difference in intensity at the two ears but there is a marked phase difference. For a given obliquity of sound the phase difference is approximately proportional to frequency, representing a fixed time delay between sound arriving at the two ears, by noting which the brain decides the direction from which the sound arrives. This operation holds for all frequencies up to that at which there is a phase difference of π radians or more between sounds arriving at the two ears from a source located on the line joining them; but above such a frequency if phase difference were the sole feature relied upon for directional location there would be ambiguity in the apparent position of the source. At that stage however the head begins to become effective as a baffle and causes noticeable intensity differences between the sounds reaching the two ears, and it is by noting such intensity differences that the brain determines direction of sounds at higher frequencies. It has been stated that the frequency at which the brain changes over from phase- to intensity-discrimination occurs at about 700 c.p.s. but it must be understood that this may vary within quite wide limits in different circumstances and from person to person, and that in any case the transference is not sudden or discontinuous but there is considerable overlap of the two phenomena so that over a considerable frequency range differences of both phase and intensity will to some extent have an effect in determining the sense of direction experienced.

From the above considerations it will be clear that a directional effect is to be obtained by providing impressions at the two ears of low frequency phase differences and high frequency intensity differences, and it would appear that in reproducing from two loud speakers the differences received by two microphones suitably spaced to represent human ears would give this effect to a listener if each microphone were connected only to one loud speaker. It can be shown however that phase differences necessary at the ears for low frequency directional sensation are not produced solely by phase differences at two loud speakers (both of which communicate with both ears) but that intensity differences at the speakers are necessary to give an effect of phase difference: while initial intensity differences from the sources necessary for high frequencies are not sufficiently marked when the sounds reach the ears, and to produce suitable effects therefore the initial intensity differences must be amplified. It is for this reason that the aforementioned methods previously proposed (wherein only pressure microphones were used) are not succeessful in achieving the desired effect, these necessary alterations not having been understood or in any way attained in those prior arrangements.

It will be seen therefore that the invention consists broadly in so controlling the intensities of sound to be, or being, emitted by a plurality of loud speakers or similar sound sources, in suitable spaced relationship to the listener, that the listener's ears will note low frequency phase differences and high frequency intensity differences suitable for conveying to the brain a desired sense of direction of the sound origin. In other words the direction from which the sound arrives at the microphones determines the characteristics (more especially, as will become apparent hereafter, the intensities) of the sounds emitted by the loud speakers in such a way as to provide this directional sensation.

It must be understood that the manual control by an observer of intensities of a plurality of loud speakers spaced round a motion picture screen has previously been proposed but this method suffers considerably from the defects indicated above, and in any case is very difficult and inconvenient to operate. No novelty for mere intensity control per se is however claimed, except insofar as the nature of the control is such as to provide the necessary relative phase and intensity difference sensations.

If in accordance with the invention the sound is first recorded and subsequently reproduced from the records, the control may wholly be effected either during the recording or during reproduction, or may be partially carried out in each stage. It must be understood that wherever throughout this specification the words "sound transmission" are employed (more especially in the claims specified below), they cover (unless the context otherwise requires) not only the case in which impulses pass directly from the microphones to the loud speakers, but also those arrangements embodying an intermediate process or system of recording; and in the latter cases the said words apply to either, or both, the passage of impulses from the microphones to the recording system, and from reproducer to the loud speakers.

More specifically the invention consists in a system of sound transmission wherein the sound is picked up by a plurality of microphone elements and reproduced by a plurality of loud speakers, comprising two or more directionally sensitive microphones and/or an arrangement of elements in the transmission circuit or circuits whereby the relative loudness of the loud speakers is made dependent upon the direction from which the sounds arrive at the microphones.

The invention also consists in a system of sound transmission wherein the sound is received by two or more microphones, wherein at low frequencies difference in the phase of sound pressure at the microphone is reproduced as difference in volume at the loud speakers.

The invention further consists in a system of sound transmission in which the original sound is detected by two or more microphones of a type such as velocity microphones whose sensitivity varies with the direction of incident sound, and in which the dependence of the relative responses of the microphones to the direction of an incident sound wave is used to control the relative volumes of sound emitted by two or more loud speakers.

The invention also consists in a system of sound transmission wherein impulses from two microphones transmitted over individual channels are adapted to interact whereby two sets of impulses are further transmitted consisting in half the sum and half the difference respectively of the original impulses, said impulses being thereafter modified to control the relative loudness of loud speakers whereby the sound is to be reproduced.

The invention also consists in a system of sound transmission wherein the sound is picked up by two directionally sensitive microphones which are so spaced and/or with their axes of maximum sensitivity so directed relative to one another and to the sound source, that the relative loudness of loudspeakers which reproduce the impulses controlled by the direction from which the sound reaches the microphones.

The invention also consists in a system as set forth above wherein two sets of impulses are mechanically recorded in the same groove.

The invention also consists in a system as set forth above wherein the impulses are transmitted by radio telephony.

The invention also consists in a system as set forth above in combination with means for the photographic recording or transmission and/or reproduction of pictures.

The word channel, as employed herein means an electric circuit carrying a current having a definite form depending upon the original sounds in the studio. Thus two channels may be different not only because the average intensities or types of current in them differ but also because they originate from two microphones in different positions in the studio.

The nature of the invention will become apparent from

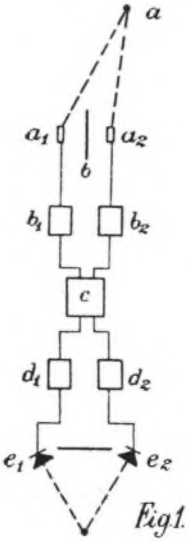


Fig.1.

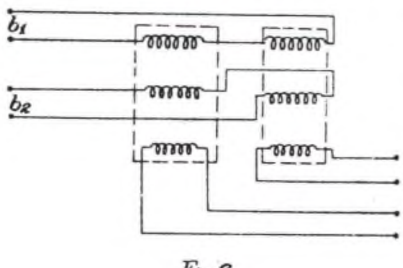


Fig.3

Fig.2.

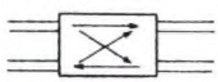


Fig.4.

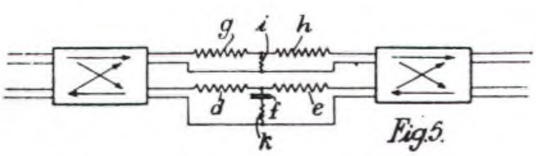
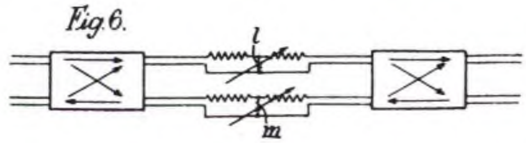


Fig.5.

Fig.6.


the following description of various methods and modes of carrying it into effect but it must be understood that the different forms described are given merely by way of example and do not impose any restrictions upon the scope of the invention or the manner and means whereby it may be accomplished.

The description will be more readily understood by reference to the accompanying drawings, wherein

Figure 1 represents diagrammatically the assembly of one system according to the invention:

Figure 2 represents a microphonic arrangement for use according to one form of the invention:

Figure 3 represents a transformer arrangement employed in one form of the invention: and

Figure 4 shows a symbolic representation of the arrangement shown in Figure 3.

Figures 5, 6 and 7 represent various circuit arrangements applicable to various forms of the invention, while

Figures 8, 9, 10 and 11 represent different forms of sound recorders which may be employed.

It will be clear that the invention is particularly applicable to talking motion pictures and the following description will therefore be given with reference to this application. In one form of the invention convenient for this purpose shown in Figure 1 the sounds to be recorded and reproduced with the pictures may be received from a source a by two pressure microphones a_1, a_2 mounted on opposite sides of a block of wood or baffle b which serves to provide the high frequency intensity differences at the microphones in the same way as the human head operates upon the ears as indicated above. The outputs from the two microphones are after separate amplification by separate similar amplifiers b_1, b_2 taken to suitably arrange circuits c comprising transformers or bridge or network circuits which convert the two primary channels into two secondary channels which may be called the summation and difference channels. These are arranged so that the current flowing into the summation channel will represent half the sum, or the mean, of the currents flowing in the two original channels, while the current flowing into on each of two transformers, the secondary winding of the currents in the original channels.

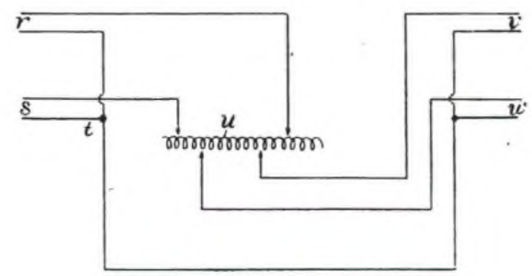


Fig.7.

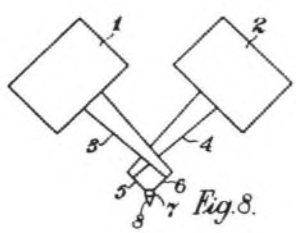


Fig.8.

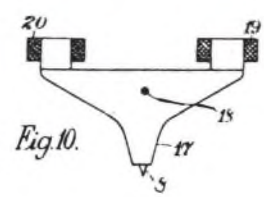


Fig.10.

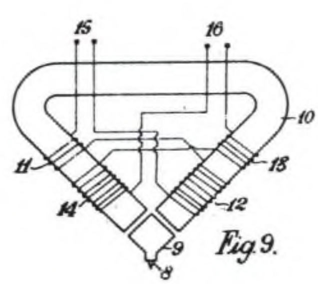


Fig.9.

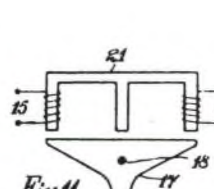


Fig.11.

One convenient transformer arrangement for this purpose is shown in Figure 3 wherein input currents from amplifiers b_1, b_2 are separately fed each to two primary windings, one on each of two transformers, the secondary winding of each each transformer, providing a "sum" or "difference" output current on account of the senses in which the primary coils are wound as shown. A diagrammatic representation of a sum and difference arrangement (which may consist of a transformer similar to that of Figure 3 or any other suitable arrangement of circuit elements) is shown in Figure 4.

In accordance with the form of the invention being described the two outputs from the sum and difference arrangement are modified in order to obtain subsequently the desired sound effects and one convenient circuit arrangement for effecting this is shown in Figure 5 which represents the portion of the circuits indicated by c in Figure 1. Assuming the original currents differ in phase only, the current in the difference channel will be $\frac{\pi}{2}$ different in phase from the current in the summation channel. This difference current is passed through two resistances d and e in series between which a condenser f forms a shunt arm. The voltage across this condenser f will be in phase with that in the summation channel. By passing the current in the summation channel through a plain resistive attenuator network composed of series resistances g, h and a shunt resistance i, a voltage is obtained which remains in phase with the voltage across the condenser f in the difference channel. These two voltages are then combined and reseparated by a sum and difference process such as previously adopted so as to produce two final channels. The voltage in the first final channel will be the sum of these voltages and the voltage in the second final channel will be the difference between these voltages. Since these voltages were in phase the two final channels will be in phase but will differ in magnitude. By choosing the value of the shunt resistance i in the summation channel and the shunt condenser f in the difference channel for a given frequency, any degree of amplitude difference in the final channels can be obtained for a given phase difference in the original channels. For the low frequencies it can be shown that the phase difference between the waves will, for a given obliquity of the sound source, vary proportionately with frequency, being very small for a very low frequency. Thus for a given obliquity of the sound the current in the difference channel will be increasingly great compared with that in the summation channel the higher the frequency. Hence the use of a shunt condenser f in the difference circuit will have the effect of producing a fixed intensity difference in the final channels for a given obliquity at all low frequencies.

For the higher frequencies as indicated above it is not necessary to convert phase shifts into amplitude differences, but simply to reproduce amplitude differences. The shunt condenser f in the difference circuit is therefore built out with a resistance k whose value is substantially equal to that of resistance i.

In building this circuit the capacity of the condenser f is of such value that its impedance is small compared with that of the series resistances d and e over the whole working range, while the value of resistance k is such that it equals the reactance of the condenser at approximately the frequency above which it is desired not to convert phase differences into amplitude differences. The value of k is in general equal to that of i, in which case the amplitude differences for high frequencies are passed on without modification.

It may be found necessary to employ more complex circuits than the shunt resistance k and condenser f in the difference circuit and shunt resistance i in the summation circuit, which however form the basic arrangement. However it must be understood that the circuits employed may be considerably modified as required without departing from the scope of the invention.

The outputs from the modifying circuit c (Figure 1) are passed to amplifiers d_1, d_2 and thence to loud speakers e_1, e_2 suitably disposed on each side of a picture screen. It is to be understood that Figure 1 merely traces the passage of intelligence from the source a to a recipient and no recording or reproducing system has been shown. Such may however be inserted anywhere along the electrical circuit such for example as between amplifiers b_1, b_2 and modifying assembly c, or between assembly c and amplifiers d_1, d_2.

In the latter case the impulses transmitted through the two channels as indicated above may for example be recorded on two sound tracks on a film by any suitable or known means, each of which records may comprise either a sound track of constant density and variable width (e.g. an oscillograph record), or a sound track of constant width and variable density (e.g. a light valve record). Alternatively both records may be made on a single track comprising a combination of the variable width and variable density forms of recording.

Such a record may be reproduced by passing light from the same slit through the two tracks, separating the beam into the two record portions by means of prisms or like optical means and employing the outputs from two photoelectric cells, excited by these separate parts of the beam (after amplification) to operate two loud speakers disposed one on each side of the screen upon which the cinematograph pictures are projected.

From the above description it will be clear that obliquity of the direction of sound wave propagation relative to the microphones a_1, a_2 will produce differences of intensity at the loud speakers so as to give an impression to an observer of oblique sound incidence.

If two very small microphones are used and placed very close together it may be found possible to obtain microphone outputs which do not differ appreciably in amplitude but only in phase for all working frequencies. In this case the modifying circuit may be arranged to convert phase differences into amplitude differences throughout the entire frequency range. The phase differences dealt with at the low-frequencies however may be so small that in this case slight differences in the two microphone circuits would have very large effects. On this account microphone spacing of the same order as that of the human ears is most suitable.

It will be appreciated that the amount of modification necessary to the impulses transmitted through the summation and difference channels as indicated above depends upon a number of factors, including the relative spacing of the microphones and of the loud speakers, and the size and positioning of the screen. It can be shown that for low frequencies ω the degree of modification required in the difference channel as compared with the modification in the summation channel is given by:—

$$K = \frac{2v}{j\omega} \cdot \frac{y}{\theta k} \cdot \frac{s}{x}$$

where

v = velocity of sound.
y = fraction of half picture film width which the image of the sound source is off centre.
θ = angle of obliquity, in radians, of the source from the median plane between the microphones.
k = effective distance apart of the microphones.
s = width of screen of theatre.
x = distance apart of loud speakers in theatre.

This expression in effect gives the impedance of the shunt capacity f in the difference channel in terms of the resistance i in the summation channel. It holds for all frequencies where k is small compared to the wavelength, and is based on the assumption that the θ is small and that x and s are small compared with the distance of the listener from the screen and loud speakers.

The portion $\frac{y}{\theta k}$ is a factor of the recording, and is constant for a given arrangement if either the camera is in line with the microphones and the centre of the picture, or the action does not move appreciably to or from the microphones and camera. When recording, the relative distances of camera and microphones and the focal length of the lens may be adjusted to maintain this factor a constant.

The expression $\frac{s}{x}$ is a constant for the theatre. As regards low frequencies only, the distance apart of the speakers need not exceed the screen width, but should certainly not be closer than 70 percent. of the screen width. The closer the loud speakers the greater the necessary power handling capacity, but the less the troubles introduced by formation of stationary waves.

For the high frequencies no definite expression can easily be obtained, and the modification, if any, used will probably have to be gauged empirically by trial and error.

The argument and formula given above are based on a direct wave analysis and may have to be considerably modified in order to allow for reflection of other acoustic effects. It is preferred therefore to introduce the modifications it is proposed to employ, at the theatre since all factors will then enter into consideration. It will be clear that, as indicated above, the modifying networks and channel arrangements may be employed between the microphones and the film during recording, or between the record and the loud speakers during reproduction, and the latter course, in addition to allowing of adjustment of the arrangements to suit the particular theatre as indicated above, has the additional advantage that the sound film can be reproduced by a single reproducing head or channel if, for example, one of the dual arrangements breaks down, or in a theatre which, having one installation, does not wish to go to the expense of installing a second apparatus.

In order to employ successfully a system of the kind described above it is necessary to carry out preliminary experiments to determine the most suitable value of modification to be employed for each recording, and it is also necessary to standardise various factors entering into every recording. In the preliminary experiments, before recording, volume indicator measurements may be made with a standard sound source placed at the extremes of the "set", i.e. the space within which recording is to be effected, and from these the proposed modifying network laid out. A further experiment may also be effected to standardise phase angles on the film. At the theatre a simple adjustment may be provided to check and balance the input to the two channels, a length of test film being used for this purpose. It will thus be seen that the total theatre equipment necessary is very simple and consists in a transmission modifier (comprising two or four transformers, for example, artificial line resistances and the control network, which may be no more than a condenser and a resistance) and two normal sound-reproducing heads or pick-ups, or one specially designed head or pick-up adapted to separate the two recordings to two complete reproducing channels. There is no reason why the second channel used should not be the "stand-by" channel now often installed for safety since if, as indicated above, one of the channels breaks down reproduction may be continued without serious consequences on the other channel only.

In connection with the standardization indicated above, while the binaural "transfer" frequency (from phase- to intensity-discrimination) need have no definite significance in recording, since it is a function of the human brain, it is nevertheless necessary to fix a change-over frequency from high- to low-frequency working for recording, since this frequency fixes the values of the elements in the modifier and thus the form of modification to be used, the distance apart of the microphones and the form of baffle between them. Any convenient frequency may be chosen as standard after experience has decided which is most suitable. Instead of standardising it may be possible from the preliminary experiments to allow electrically for variation of microphone positions and/or of microphone spacing (although the latter would be extremely difficult) and it must be understood that this arrangement falls within the scope of the invention.

The above analysis is based upon considerations which take no account of sound reflections or interference during reproduction. The reflected sound waves which arise during recording will be reproduced with a directional sense and will sound more natural than they would with a non-directional reproducing system. If difficulties arise in reproduction they may be overcome by employing a second pair of loud speakers differently spaced and having a different modifying network from the first pair: or a row of speakers may be used with a composite, progressive modifying network to supply them: or the two speakers may be placed comparatively close together.

In this last arrangement the sense of direction of the apparent sound source will only be conveyed to a listener for the full frequency range for positions lying between the loud speakers; but if it is desired to convey the impression that the sound source has moved to a position beyond the space between the loud speakers the modifying networks may be arranged to reverse the phase of that loud speaker remote from which the source is desired to appear, and this will suffice to convey the desired impression for the low frequency sounds. With this arrangement of loud speakers close together, however, it would not be possible to effect a similar illusion in connection with high frequencies.

The system so far described employs to receive the sound waves two nondirectional microphones, e.g. pressure microphones. Directionally sensitive microphones may also be employed spaced a small distance apart, the outputs being modified as indicated so that the relative outputs of the loud speakers are controlled both by differences in phase and differences in magnitude of the microphone outputs. Such directionally sensitive microphones may be, but are not necessarily, of the type known as velocity microphones, and preferably provided with movable conductor elements so light as to move substantially as the surrounding air.

Velocity or moving conductor microphones (e.g. moving strip microphones) are very suitable for any system according to the invention and in addition to use with circuit arrangements described above: they may also be employed with various alterations in the circuits. These microphones give a response varying as the cosine of the angle of incidence of the sound relative to the direction of normal or optimum incidence, and they therefore have the advantage that a certain degree of loud speaker output separation may be obtained without phase-conversion or like network modifications.

Three general arrangements employing velocity microphones are possible and in all cases the microphones are placed as near together as possible instead of being spaced as artificial ears, as in the case of pressure microphones.

(1) Two velocity microphones are placed one with its axis of maximum response directly facing in the direction of the centre of the scene, and the other with its axis at right angles to that direction. Both moving strips are in line, and arranged so that this line is vertical, whereas the sound source moves in a horizontal plane. A performer speaking from the middle of the scene will affect only the face-on microphone, but if he moves to one side both microphones will provide outputs, while if he moves the other way similar outputs are provided but the phase of the edge-on microphone is reversed. Since the microphones are close together no phase differences are experienced between them and if their outputs are summed and differenced after a suitable amount of relative amplification the two final channels differ in magnitude in the correct manner for operating the loud speakers to give the desired directional effect. Such sum and difference arrangement differs from the modifying network employed with pressure microphones in that the pressure type provide phase differences (whereby direction is determined) which have to be converted, whereas with the velocity type the edge-on microphone provides an output proportional to the obliquity of the source. A suitable modifying arrangement for this form of the invention is shown in Figure 6. This is substantially identical with that shown in Figure 5 except that the shunt condenser f and resistance k in series, and the shunt resistance i are replaced by shunt resistances l m which are preferably variable as shown. These lines therefore form artificial attenuators and by altering their relative attenuation the intensity differences in the two lines corresponding to a given obliquity of sound is controlled.

(2) Two velocity microphones or microphone elements may be placed with their axes perpendicular to one another and each axis at 45° to the direction of the centre of the screen. This arrangement is represented diagrammatically in Figure 2 wherein n and o represent two velocity, or directionally sensitive microphones one above the other arranged perpendicular to one another and at equal angles at 45° to the direction of the centre of the field from which sound is to be received. It will be clear that movement of the sound source a laterally to a position of p removed from the centre of the field will result in the sound waves striking o at a more acute angle than they strike n and differences in the microphone outputs will result. The microphones are sufficiently close together to render phase differences of the incident sound negligible and the output amplitudes therefore differ approximately proportionally to the obliquity of the incident sound. They may therefore be amplified similarly, and supplied directly to the loud speakers to which they will give the correct amplitude differences for the desired directional effect provided the relationship between the various dimensions of the recording and reproducing "lay-outs" are correct. If it is desired to accommodate any differences between the "lay-outs" the outputs may be modified by networks, in the manner described, suitably to increase or decrease the differences between them. An arrangement such as shown in Figure 6 is suitable for this purpose, and such an arrangement may of course also be employed even if the lay-out is correct if it is desired for any reason to control or modify the amplitude differences of the loud speaker outputs.

(3) Two microphones may be arranged with the two axes lying symmetrically to the direction of the centre of the field and with an angle between them of say θ degrees, so that sound from a performer at the centre subtends an angle of $\frac{\theta}{2}$ degrees to each microphone. If θ is small a small movement of the performer to one side is sufficient to make one microphone "edge-on" and to reduce its output to zero, while if θ is large a large movement of the performer is necessary to do this. By making θ adjustable different "layouts" may be accommodated without the modification indicated under (2) and it will be clear also that this provides a method of directional sound transmitting, recording and reproduction which avoids the necessity of combining and reseparating the two channels.

The microphone elements in any of the above cases may be enclosed in a single casing if desired for convenience, and may also be positioned in a single magnetic system common to both. Two velocity microphones set in line with one another and with their axes of maximum response symmetrically inclined to the direction of the centre line of the scene, may, if placed one above the other, be employed also to provide significance of vertical as well as horizontal movement of the sound source in a plane perpendicular to the axis of maximum response of the microphone system. Such vertical displacement of the source will in this arrangement give phase differences to the outputs while lateral displacement gives amplitude differences, and these can be separated, the phase differences converted to intensity differences by modifying networks, as described, and the resulting impulses employed to operate four or more loud speakers distributed round the screen. The transmission in such a system occupies only two channels (one leading from each microphone) up to a point in the system where each of these channels is divided into two parallel channels thus providing four channels in all at this point. Two channels, one from each parallel pair of these divided channels, are connected to one modifying network adapted to deal with phase differences, and the other two channels, one from each pair, connected to another modifying network adapted to augment intensity differences. Each modifying network operates a plurality of loud speakers providing a directional sensation in one direction, and in this manner directional senses in two directions at right angles can be obtained. It will be seen that in such an arrangement the transmission and/or recording (which is the most expensive and difficult operation of the system) may be effected over only two channels although directional sensations in two perpendicular directions are subsequently obtained. A similar effect may be obtained with a plurality of pressure microphones by employing suitable modification previous to transmission.

In obtaining a complete directional "sound picture", i.e. both horizontal and vertical directional effects, the invention is not limited solely to the use of two microphones. A plurality may be employed and their outputs suitably collected, modified and separated to transmit suitable differences of impulses to a plurality of loud speakers. The general feature is that two transmitting channels, receiving impulses from two or more microphones for example, communicate impulses which can be modified and separated to provide two directional senses at right angles to one another, the sounds whereby this is done being provided by a plurality of loud speakers. It will moreover be clear that if the sound source moves away from or towards the microphones the overall intensity of the combined loud speaker propagations will vary and thus provide indication of the position of the source along that axis. Full three-dimensional location of the source is thus obtained by this arrangement.

It will be seen that while with pressure microphones it is preferred to transmit phase differences rather than amplitude differences and convert from one to the other as late as possible prior to reproduction with velocity microphones it is more convenient to transmit the two channels in phase but at different amplitudes, the only modification then necessary being an increase or decrease of the amplitude differences should the reproducing "lay-out" differ from the recording "lay-out" or should more than two loud speaker positions be used.

There is a simple method by which modifications for increase or decrease of differences between channels may be effected if no conversion of phase differences into amplitude differences is required. The method is particularly useful for the operation of more than two loud speakers, and is also useful for working into high impedances such as the grid impedance of a thermionic valve. The arrangement is shown diagrammatically in Figure 7. If the transmission is effected in the form of two channels r s of similar phase but different amplitudes, an alteration of these amplitude differences may be effected by connecting one wire of each channel r and s together at t and connecting a choke u between the other two wires of the two channels. The outgoing channels v and w whose difference is to be a modification of the original difference, are connected by one wire each to the common point t of the original channels, and by their other wires to tappings along the choke u. If the differences are to be increased, the tappings at which the output channels are connected lie outside the tappings to which the input channels are connected, so that the choke operates in effect as an auto-transformer amplifying the difference voltages. Similarly, for a reduction of differences, the output channels are tapped intermediately between the two input channels. Modifications of this arrangement in which the devices are balanced about earth, etc. may be arranged, but the chief advantage is that the modification is varied entirely by altering tappings along a transformer or choke, and that no great power loss is involved.

This arrangement of a choke or transformer is well suited to working a number of loud speakers for binaural reproduction. In this case, the two outputs from power valves are fitted to a choke such as u along which the loud speakers are tapped. The position of the loud speaker tappings can be adjusted to suit their relative positions, and it can be arranged that the valves are working into their best impedances. Transformers may be used to ensure the speakers taking their correct fraction of the output.

While, in connection with the above described systems, it is suggested that when it is desired to record sounds for subsequent reproduction this may be done upon a film, the invention is not limited to that medium since the recording may if desired be effected on discs or cylinders of suitable material. In carrying out the invention in this manner the two channels may if desired be recorded in separate grooves but it is preferred that they be recorded in the same groove having a hill and dale and also a lateral cut movement. For the purposes of television previous proposals have been made whereby a wax disc has a sound record as a hill and dale cut and a picture record as a laterally cut V-shaped groove at the bottom of the hill and dale groove, or vice versa. Such records appear unsuited for separate and distinct sound recordings since undoubtedly considerable cross-talk between the two recordings would occur. They can however be used for two channels of the kind contemplated in the present invention, one being only slightly different from the other, since a certain amount of cross-talk in this case does not matter, or can be allowed for. Furthermore, the records now proposed are distinguished from those previously known in that both channels may be recorded as separate cuts in one groove and may be recorded by a single recording tool (either of moving iron or moving coil type) and be reproduced therefrom by a single reproducing device or pick-up.

If the two channels being recorded are directly picked up from two microphones, or are intended to work unmodified into two speakers, that is with intensities and qualities similar to those of the original sounds received, it is preferred not to cut one track as lateral cut and the other as hill and dale, but to cut them as two tracks whose movement axes lie at 45° to the wax surface, or at some other convenient angle dependent on the relative available intensities from lateral cut and hill and dale respectively. If, however, the two channels recorded are such as summation and difference channels, it is preferred to separate them completely into pure hill and dale and pure lateral cut, i.e. to make the recording axes normal and tangential to the wax surface.

The result in the two above suggested cases is very similar since channels recorded at 45° to the wax surface give their sum and difference as the effective lateral and hill and dale amplitudes.

It will be appreciated that a record, cut as a combined hill and dale and lateral, may be reproduced if desired as two skew direction cuts, the basic principle being that the groove has amplitude in any direction in the plane at right angles to the direction of wax movement, and the recording and reproducing directions may be chosen as any pair of axes lines, not necessarily at right angles, in this plane.

It would appear that for such a record, a material other than that now used for lateral cut records would be desirable, and a material of the nature of cellulose acetate is indicated.

The track section is preferably adapted to work with a sapphire and have a sufficiently fine angle to give lateral as well as vertical control to the sapphire.

The recorder whereby both channels may be cut by a single tool on the same groove may take various forms, the underlying feature being that a light stylus is pulled into two directions at right angles to one another and each preferably at 45° to the wax surface.

Figure 8 shows schematically a recorder of this kind suitable for producing records having complex cuts. 1 and 2 represent the driving elements of two recorders normally adapted for cutting lateral cut records. These driving elements drive arms 3 and 4 about axes at right angles to the plane of the paper within 1 and 2. The ends of these arms are connected by ligaments 5 and 6 to the end of a reed 7 which extends backwards along an axis perpendicular to the paper to supports not shown. This reed carries a cutting sapphire 8. Movements of the recording arms 3 and 4 produce movements in the end of the reed 7. Thus, currents in movement 1 will cause the reed 7 to move along an axis approximately 45° to the vertical rising from left to right across the figure. Similarly, currents in movement 2 will produce movement of the reed 7 in an axis at right angles to the former axis, while currents in both movements will of course result in vertical movement of the reed.

Another such form of recorder shown in Figure 9, representing a moving iron recorder, may consist in a short reed 9 mounted close above and parallel with the wax track and carrying the cutting sapphire 8. This reed 9 may extend backwards perpendicularly to the paper to supports (not shown) which join the top of a laminated pole system 10 to complete a polarising magnetic system therewith. The two laminated arms of the pole piece 10 extend down towards the free end of the reed 9. These arms form two poles adjacent to a square portion of the reed at its free end, each pole being adapted to pull the reed in a direction at 45° to the wax surface. The reed may be suitably damped, e.g. by a rubber line and have a resonant frequency at the top of, or above, the working range. The two pole pieces may be wound with speech coils, and the energisation of one of these moves the sapphire in an upward direction at 45° to the wax surface. The terminals 15 of one channel are connected to main winding 12 and compensating winding 11. The terminals 16 of the other channel are connected to main winding 14 and compensating winding 13. Current in either channel will pull the reed towards the pole carrying the main winding, the purpose of the compensating winding being to prevent movement of the reed away from the other pole due to the flux drawn away from this pole by the main winding. With the winding shown, currents in either channel will cause the reed to cut a track at approximately 45° to the vertical. By a suitable rearrangement of windings, or by a suitable transformer connection between the channels and the terminals of the recorder as shown, any other movement axes may be obtained. Thus for example the tool may have one movement by torsion of its supporting reed and another by flexure thereof.

An alternative moving coil design which may employ electromagnetic damping may consist of a moving member in the shape of a **T** as shown in Figure 10. The recorder sapphire 8 is supported on a light **T** member 17, which is supported at 18 by elastic means such that it may rotate about this point, and may also translate vertically, though it is resistant to horizontal movements in the plane of the paper. The device is driven by moving coils, e.g. speech coils, 19 and 20 which are freely located and immersed in the steady magnetic field provided in annular gaps in a magnetic system, not shown. Current in one of the moving coils tends to both rotate and translate the device so that the sapphire 8 moves along an axis at approximately 45° to the vertical. The movement of this device may be damped and equalised along the lines described in British Patent Specification No. 350,998. As before any required axes of movement may be obtained by suitable interconnection of the two driving coils. Such a movement preferably has the same natural frequency for both rotation and translation. Further the distribution of mass is preferably such that a small instantaneous force applied at one coil produces no movement at the other.

Figure 11 shows another form of recorder similar in principle to the one shown in Figure 10 except that a moving iron drive is employed. The member 17 moving about axis 18 is constructed of magnetic material, or has a magnetic upper portion. The "E" shaped member 21 is polarised either by being partially permanently magnetised, or having a magnetising winding on it, so that the central pole is of opposite polarity to the two outer poles. Speech windings on the outer poles are brought out to terminals 15 and 16 to which the two channels are connected.

In all the devices described above, the angles of the axes defining the movements of the sapphire can be altered by suitably connecting the speech windings; for instance, axes which are normally inclined at 45° to the wax surface can be converted into pure hill and dale and lateral cut axes by arranging that the speech windings are in series aiding for one channel and opposing for the other channel. In like manner any axis conversion can be effected by suitably combining the channels through transformers.

In designing an electric pick-up to reproduce both channels care must be taken that the inertia is kept as low as possible, and with this in mind a very light replica of any of the above described recorders may be employed. Preferably, a moving system in the form of a **T** following the lines of the moving iron recorder shown in Figure 11 is employed as best suited for the purpose. Since the fundamental resonant frequency of a pickup appears to be of no critical importance as regards its characteristic, it may not be necessary to adjust the resonant frequency in the two modes to the same value, which would simplify the design. Adjustments for sensitivity in the two modes may be made by suitably connecting coils wound on the two limbs of the magnetic circuits. As in the recorder design the distribution of mass in the reproducer is preferably such that forces producing motion in one direction (e.g. lateral movements) leave it substantially undisturbed in its reproduction by motions in another direction (e.g. hill-and-dale).

A good binaural effect may be obtained by giving directional significance to only a limited range of frequencies. For example, although good reproduction requires the transmission of all frequencies up to, say, 10,000 c.p.s. yet a good directional effect is obtained from frequencies up to, say, 3,000 c.p.s. This would assist disc recording of the binaural impulses since the lateral cut which represents the sum of the two channels to the speakers might have a frequency range extending to 10,000 c.p.s. whereas the hill-and-dale cut need transmit frequencies no higher than 3,000 c.p.s. This would considerably simplify the design of the recorders and pick-ups in that low inertias would only be required for the lateral cut and design would thus be greatly simplified.

These frequencies are given merely by way of example, are not necessarily the optimum frequencies for design of this character, which will be determined by other considerations.

In transmitting the two channels indicated in the various systems above described, instead of employing line transmission, radio transmission may if desired be employed. Each channel may be separately transmitted or preferably the two channels may be sent as different modulations of the same carrier wave. Thus one channel may be transmitted as an amplitude modulation and the other as a phase or frequency modulation of the same carrier wave. Alternatively the two channels may be transmitted as amplitude modulations of different carrier waves which are 90° out of phase, the two waves being radiated from the same aerial in combination as a single wave propagation. Various systems for the transmission and reception of duplex radio signals along these lines are known and any one of such or similar arrangements may be used in connection with the invention described herein according to its applicability or convenience in the circumstances under consideration. It must be understood that with such a system of duplex radiation, it is possible, if desired, to perform one of the summing and differencing processes in the radio link. For example, by demodulation at the receiving end with two carrier waves 90° out of phase, which carrier waves are 45° out of phase with the original modulating carriers, the resultant low frequency channels are the sums and differences of the original low frequency channels at the transmitter.

The hereindescribed system while being especially applicable to talking pictures is not limited to such use. It may be employed in recording sound quite independently of any picture effects and in this connection (as well as when used in cinematograph work) it seems probable that the binaural effect introduced will be found to improve the acoustic properties of recording studios and to save any drastic acoustic treatment thereof while providing much more realistic and satisfactory records for reproduction. Furthermore, the system may clearly be employed when the microphone outputs are led to the loud speakers instead first of being recorded, and such an arrangement may for example be employed in public address systems in which directional sound effects are desired. In general the invention is applicable in all cases where it is desired to give directional effects to emitted sound. Also in all cases, both when the impulses are fed to the loud speakers without recording and when they are recorded for subsequent reproduction the total modification and/or interaction of the channels may be accomplished in more than one stage. For example, using pressure microphones, the low frequency phase differences may be augmented, the medium frequency phase differences converted to amplitude differences, and the high frequency amplitude differences augmented in a first stage of modification; the low frequency phase differences may then be converted to amplitude differences in a later stage of modification. One or both of these stages may occur either before or after the sound has been recorded. In this manner the very small low frequency phase differences are augmented before they are amplified, so avoiding troubles due to small low frequency phase shifts in amplifiers.

A Report on the Proposed NAB Disc Playback Standard*

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The 1963 (Proposed) *NAB Audio and Reproducing Standard for Disc Recording and Reproducing* are reviewed and discussed. Direct comparisons are made with the 1953 Standards and the changes are explained. The new sections covering stereophonic reproduction are discussed from the point of view of the state of the art and the requirements of the broadcaster.

INTRODUCTION

PHILOSOPHIES have been variously defined as being created out of historical pattern, present necessity or idealized contemplation. It is hoped that the philosophy of the 1963 NAB Disc Standard represents the best of the historical data and experience as well as the present necessity of the broadcaster, while contemplating the best way to achieve a forward-looking Standards document.

The foreword of the 1963 *NAB Disc and Recording and Reproducing Standard* contains the following statement: "These standards and good engineering practices are for the guidance of the broadcasting industry . . ." With this basic concept the entire Standard becomes a tool defining good engineering practices not specifically covered in the FCC Rules and Regulations. While the FCC defines the basic audio requirements for broadcast systems in terms of response, distortion products, signal-to-noise ratio, etc., it does not specify disc or tape playback standards. It is therefore incumbent upon the broadcast industry to elaborate these standards so as to achieve technical uniformity between the various broadcast stations to permit exchange of recorded material while satisfying the basic requirements inherent in the Communications Regulations.

A section of the report of the disc subcommittee says: "With the ever increasing advancements which were being made in the art of recording and reproducing, it soon became evident that the 1953 Standards were woefully out-of-date and in need of revision. During the latter part of 1961, the Association appointed the NAB Recording and Reproducing Standards Committee which was charged with the responsibility of guiding the Association's efforts associated with this task. During November, 1961, the Committee met and decided that the standardization efforts should be broken down into two categories, namely, Disc and Magnetic Tape. As a result of this meeting, two subcommittees were established to deal with these two areas of standardization."

Brief historical data show that 15 companies[1] and their representatives contributed to the 1963 NAB Standard at 13 meetings, representing 16 committee working days between the first meeting on December 15, 1961 and July 31, 1963. Several companies contributed proposals, exhibits and verification of exhibits, representing uncounted hours devoted to the completion of the disc standard. Representation at these meetings was from broadcasters, disc manufacturers and equipment manufacturers.

While the Standard is essentially for use by U.S.-based radio stations, the committee work took cognizance of published international standards to make the Standard universally acceptable and usable.

The 1963 Standard is divided into 5 sections: Turntable

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

[1] See *Acknowledgements.*

Specifications, Disc Specifications, Electrical Specifications, Test Record Specifications and a Glossary of Terms. It was felt that dividing the Standard into these categories would delineate the various requirements, and provide a more readily usable tool.

TURNTABLE SPECIFICATIONS

Speed. The mean speed of the reproducing table shall be held to ± .3%. While this has not been changed since the 1953 NAB Standard, the recording speed, which was not specified previously, is now specified and held to ± 0.1% tolerance. The standard refers only to 45 rpm and 33⅓ rpm discs; while 78.26 rpm discs are occasionally still used, this speed is no longer considered standard.

Wow and flutter. The wow and flutter factor over the frequency range of 0.5 to 200 cps has been reduced from 0.1% to 0.04% during recording and from 0.2% to 0.1% for playback. It should be noted that no weighting curve has been introduced in this method of measurement. The technological advances in bearing and motor design and construction are reflected in this improved wow and flutter requirement.

Review of the available broadcast turntables showed that most of the broadcast units available do meet the new requirements.

Miscellaneous turntable requirements. The remainder of the turntable requirements are concerned with specific problems of the broadcaster: starting time, turntable height, and platen dimensions. With the exception of the center pin size, these standards will not be discussed here, since these parameters are not directly related to the quality of reproduction.

The center pin dimension remains at 0.285 inch ± 0.0005 inch: it will be discussed later as it relates to off-center grooves and records.

DISC SPECIFICATIONS

As in the turntable specification, several parts cover manufacturing standards that do not affect the disc playback system capability and will not be covered here.

Center hole diameter and concentricity of center hole. The 1953 NAB Standard called for a center hole of .286 in. ± .001 in. and a concentricity of .002 in. The 1963 Standard has actually been opened up to 0.286 in. + 0.001 in. — .002 in., while the center hole concentricity is now specified at ± .005 in. The relaxing of the concentricity requirement may seem to be a step backwards, but it should be pointed out that in practice the old ± .002 in. concentricity requirement was in fact barely ever attained. Even the ± .005 in. concentricity required in the new Standard places a high demand upon the disc manufacturer's facilities. If all of the dimensions and tolerances for the center pin, center hole and concentricity are observed, we will have records with a total run-out of .0095 in. How often have there been systems running with double this total eccentricity!

Disc warp. "It shall be standard that the total indicator reading (TIR) of the surface of the disc because of warping shall not be in excess of 1/16 inch and that within any 45° segment the total indicator reading (TIR) shall not exceed 1/32 inch." This is the new Standard while the 1963 Standard calls for 1/16 in. with no regard for the rate of change of the warp.

The advent of lower tracking forces and of stereo have lowered the permissible rate of vertical acceleration, thus necessitating the tighter requirement.

Groove shape. The first reference to a difference between monophonic and stereophonic discs in the 1963 Standard appears under the Groove Shape Standard. The elimination

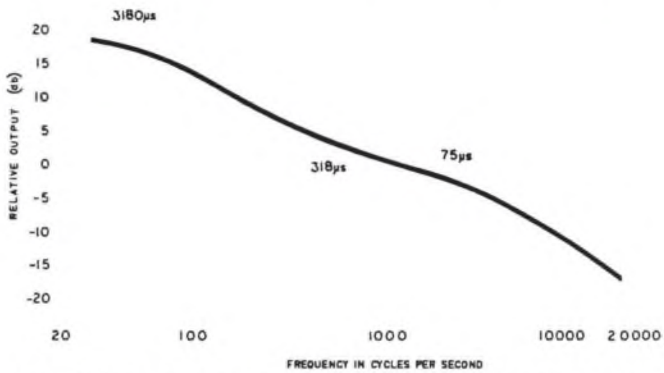

FIG. 1. NAB Standard disc reproducing characteristic: relative output *vs* frequency for constant velocity input. Tolerance ±2 db, 50 cps to 10 kc; +2 db, -3 db, below 50 cps and above 10 kc.

of the 78 rpm disc and its so-called "standard groove" leads to the use of the microgroove or long-playing groove as the standard.

The monophonic groove is standard if the groove of the finished disc has a top width of not less than .0022 in., contains an enclosed angle of 90° ± 5°, and has a bottom radius of not more than .00025 in. The stereophonic groove is standard in the finished disc when the top width is not less than .001 in., has an enclosed angle of 90° ± 2° and a bottom radius of not more than .0002 in.

The playback stylus tip radius appears as a recommendation of .001 in. + .0001, —.0002 in. for monophonic discs, and .0005 to .0007 in. for stereophonic discs, with an enclosed angle of 40° to 55°. Since the broadcaster frequently has the problem of playing both monophonic and stereophonic discs on the same turntable, a footnote points out that the use of a .0007 in. radius is recommended.

ELECTRICAL SPECIFICATIONS

The electrical specifications of the 1963 NAB Standard are based upon system playback response characteristics. The electrical playback system has been considered from the disc-stylus interface through the pickup, equalizer and/or preamp equalizer. In brief, the playback system is complete within itself and must be complete to be fed to the mixing equipment.

Frequency characteristics for monophonic and stereophonic discs. The 1963 Standard covering frequency response characteristics remains the same as in 1953. The tolerance of ± 2 db of the 1953 Standard has been modified to +2 db and —3 db below 50 cps and above 10 kc. (See Fig. 1.) This tolerance change permits slightly greater latitude for AM broadcasters encountering low-frequency overloading; because the FM broadcaster frequently encounters overmodulation in the high frequency audio spectrum with certain records, combined with the transmitter preemphasis, the additional 1 db negative tolerance below 50 cps and above 10 kc was introduced.

It is of considerable interest to the broadcaster that he utilize maximum bandwidth without feeding information into his transmitters that would generate a load outside the usable bandpass of the system. It is therefore recommended in the Standard that the disc playback system be rolled off at least 6 db per octave below 30 cps, with the 3 db point at 20 cps, and above 15 kc with the 3 db point at 16 kc.

The new standard also contains a numerical chart of response *vs* frequency in db. This should provide the measurement engineer with an accurate point reference.

Reference recorded program level, monophonic and stereophonic. The standard level for monophonic discs remains at 7 cm/sec peak at 1 kc, while the stereophonic reference level is 5 cm/sec peak at 1 kc.

The relationship of 7 cm/sec peak for monophonic to 5 cm/sec peak for stereophonic represents approximately 3 db reduction for the stereophonic channel compared to the monophonic channel.

Low frequency noise, monophonic and stereophonic. The definition of low frequency noise covers the spectrum from 10 cps to 500 cps. (See Fig. 2.) Since the turntable, the

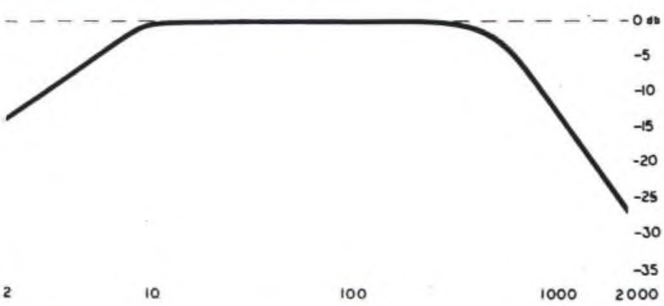

FIG. 2. Bandpass of low-frequency noise as per 1963 Proposed NAB Standard (1.4 cm/sec at 100 cps monophonic, 1.0 cm/sec at 100 cps stereophonic).

pickup and the equalizer or preamp equalizer may contribute to the low frequency noise, the measurement must cover the entire system to be functionally meaningful. The 1953 Monophonic Standard of —35 db below a reference of 1.4 cm/sec peak velocity at 100 cps has been increased to —40 db. The minimum stereo rumble figure is —35 db, using the same response spectrum and the reference at 1 cm/sec peak velocity at 100 cps.

The —40 db for monophonic discs and —35 db for stereophonic discs represents a well designed modern system, and measurements by the subcommittee confirm that the low frequency noise figures represent a high state of the art.

High frequency noise, monophonic and stereophonic. This measurement serves to measure the high frequency noise content of the disc and playback system over the spectrum from 500 cps to 15 kc referred to 7 cm/sec peak at 1 kc for monophonic, while the stereophonic reference is 5 cm/sec peak at 1 kc. (See Fig. 3.)

The high frequency signal-to-noise ratio is now 55 db minimum; the stereophonic signal-to-noise ratio is 50 db.

The 1953 Standard specified a 40 db minimum signal-to-noise ratio at the same reference, i.e. 7 cm/sec peak at 1 kc, over the 500 cps to 10 kc spectrum. The 1963 Standard

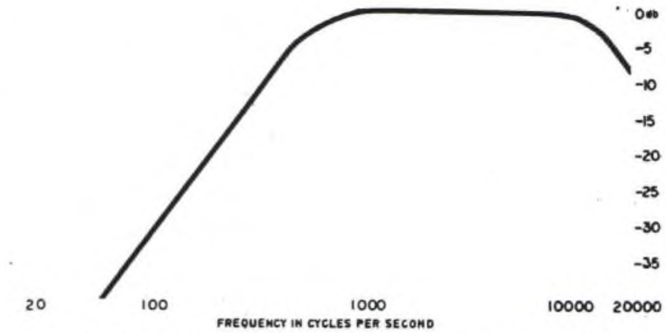

FIG. 3. Bandpass of high-frequency noise as per 1963 Proposed NAB Standard (7 cm/sec at 1,000 cps monophonic, 5 cm/sec at 1,000 cps stereophonic).

reflects primarily the advances made by the disc manufacturers. Anyone who has compared early long-playing discs or shellac 78 rpm discs with those of recent manufacture will readily appreciate this improvement.

Additional electrical specifications, stereophonic. Definitions of planes of modulation, channel orientation, and phasing appear in the Standard but will not be discussed here. Added to these are the requirements for channel separation, channel balance and channel phasing.

Briefly, the system separation between a recorded and unrecorded channel shall be at least 26 db over the frequency range 100 cps to 7500 cps; below and above these frequencies the separation shall not degenerate at a rate greater than 6 db per octave. The channel balance shall be maintained within ¼ db at 1 kc.

TEST RECORD SPECIFICATIONS

The broadcast station operator must have discs available to check system capabilities and conformance to the Standard. Included in the 1963 NAB Standard are the specifications for such a test record to permit the operator to run an audio performance check. This disc will contain material to measure all of the specifications of the Standard for monophonic and stereophonic electrical requirements; it will be available from NAB Headquarters.

GLOSSARY

The glossary contains a series of terms peculiar to the disc recording and reproducing industry. It is intended to translate the jargon of these specialized fields to meaningful language for the broadcaster, who is not necessarily a disc recording specialist. While certain terms have become obsolete, certain new expressions have appeared in the glossary. The disc industry has managed to invent new terms to meet new techniques that are generally descriptive of the particular technique.

CONCLUSION

The work of the proposed 1963 NAB Disc Standard is complete and with possible minor modifications will be adopted by the NAB. While no Standard can ever be considered eternally complete, it is hoped that this standard has been sufficiently forward-looking to enhance the broadcasting of discs for some time to come. The areas defined in the Standard have been set on a realistic and high plane. The obvious purpose is to achieve for the broadcaster the best possible equipment and discs by setting high standards. There has not been any assumption that this standard will never be modified, and as newer techniques are better understood they will be incorporated to further advance the broadcasting industry.

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Mr. Bubbers is a member of the Institute of Electrical and Electronics Engineers, the Professional Group on Audio of the IEEE, and the Audio Engineering Society.

The RIAA Engineering Committee*

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A brief review is given of the history and accomplishments of the Engineering Committee of the Record Industry Association of America, which was established to act upon engineering matters for the Association and prepare industry standards for phonograph records. (Standards for prerecorded magnetic tapes were included later as the need arose.) The four standards currently in existence, three for disc phonograph records and one for pre-recorded magnetic tapes, are summarized briefly.

INTRODUCTION The Engineering Committee of the Record Industry Association of America (RIAA) was formed in 1953. At that time microgroove records at operating speeds of 33⅓ and 45 rpm had been in existence for some five years and were replacing the 78 rpm records. There were some record standards in existence due to the activities of the National Association of Broadcasters (NAB) and the Electronic Industries Association (EIA). Record engineers participated in these standardization activities but lacked the advantage of having their own organization, and hence the close cooperation that would permit them to better reflect their own views into standards. To overcome this deficiency the Board of Directors of RIAA authorized the establishment of an RIAA Engineering Committee.

EARLY MEETINGS

At the first meeting held on January 16, 1953 those present were: Glenn Wallichs of Capitol, representing the RIAA Board of Directors, John Griffin, executive secretary of RIAA, and several engineers: E. H. Uecke for Capitol, W. S. Bachman for Columbia, Charles Lauda for Decca, and H. I. Reiskind for RCA. Robert Fine, representing Mercury, was not present at this meeting.

The committee was informed that the first order of business should be a recommendation for the standardization of record speeds and sizes which would have optimum characteristics for both performance and manufacture. A discussion of the problems then followed.

A second meeting was held on January 21, 1953. The discussion of the first two meetings was confined to "product uniformity". In essence, the following conclusions were reached regarding the major items:

1. Ultimately the microgroove record was expected to replace the standard-groove 78 rpm record. Therefore, only the microgroove record was to be considered.
2. Three record diameters, 7, 10, and 12 in., were thought necessary to meet the program time requirements.
3. Two center hole sizes, with nominal diameters referred to as "small" (0.286 in.) and "large" (1.504 in.), were selected as the only ones requiring further consideration.
4. Two turntable speeds, 33⅓ and 45 rpm, were considered to be the only ones requiring consideration.

It is interesting to note, after some 14 years, how correctly the committee selected the main objectives.

FIRST STANDARD

The first standard produced by the Engineering Committee, *Standard Recording and Reproducing Characteristic* (Bulletin E1) was approved by the Board of Directors on January 29, 1954. This basic standard is still in force today, and is also an international standard.[1] It was a milestone in the progress of record standards, for prior to its adoption preamplifiers for record reproduc-

* Presented October 18, 1967 at the 33rd Convention of the Audio Engineering Society, New York.

tion showed many compensation positions such as: *LP, AES, 78,* and *ORTHO.*

The characteristic selected was one adopted by the NAB as a revision of their earlier standard. It was the outcome of a careful study made by the NAB Engineering Committee of the many recording characteristics then in use. Record engineers participated in this committee action.

DIMENSIONAL STANDARDS

Bulletin E 2, *Dimensional Standards of Disc Phonograph Records,* was approved by the Board of Directors in September 1957. This standard was subsequently superseded by Bulletin E 4, *Dimensional Standards of Disc Phonograph Records for Home Use,* which was officially approved March 16, 1961.

STEREO DISC STANDARD

Bulletin E 3, which covers the *Standards for Stereophonic Disc Records,* was approved as an RIAA standard on March 25, 1958. The content of this bulletin was subsequently rearranged and clarified on October 16, 1963, but a new bulletin was not issued. This bulletin contains a definition of a "true stereophonic record": ". . . has two distinct orthogonal modulations derived from an original live recording in which a minimum of two separate channels are employed". This definition, except for a slight change in wording, has been used by the Federal Trade Commission as a guide in policing pseudo-stereophonic recordings.

The evolution of the stereo disc standard deserves special mention, for it is indeed rare when standardization is achieved prior to production. In September 1957, two methods of producing stereophonic records, having two channels of information in the same groove, were demonstrated to members of the record industry. Decca of London utilized a vertical-lateral system; Westrex rotated the axis so that the modes of modulation were at a 45° angle with respect to the record surface. Thus the two systems were similar except for the direction of modulation. RIAA engineers promptly sought a three-month study period for evaluation purposes. The task was not an easy one, since stereo cutters and pickups were not available commercially, and would not be for many months. Producers of cartridges had not yet started the development of stereophonic pickups. Theoretical studies offered the only immediate means of evaluation.

The Westrex system was demonstrated at the AES Fall Convention in New York City on October 11, 1957. On November 7, 1957 the EIA Engineering Committees concerned were advised of the two systems and their aid solicited in evaluating them.

On December 17 and 18, 1957 the RIAA Engineering Committee met in Indianapolis. R. C. Moyer and his engineers gave comprehensive reports on their studies, pointing out the advantages offered by the 45°-45° system due to symmetry. Both channels would be alike with respect to frequency response and distortion and other characteristics, since in reality both were vertical systems operating at a 45° offset angle. On the other hand, the vertical-lateral system consisted of one lateral channel, where push-pull action during reproduction would result in cancellation of even order distortion terms; and a vertical channel, single-ended in action, where cancellation of even order terms would not occur. Hence, tracing distortion and quite likely many other characteristics of the two channels would be different, an undesirable condition. M. S. Corrington reported on his computerized study on the "tracing distortion" of the two systems.[2] He stressed the importance of a small stylus tip for reproduction in order to minimize the strong second-order tracing distortion term in the vertical-cut channels. Other members made their reports and the committee concluded that the 45°-45° system offered many advantages and should be selected as the standard. A most welcome visitor, G. F. Dutton of EMI reported that at a meeting in Zurich on November 28, 1957 the European recording engineers had decided in favor of the 45°-45° system. Thus the decision became a unanimous one, internationally, and one that in retrospect appears to have been remarkably foresighted.

The EIA engineers were notified of the RIAA decision at a meeting in New York City on January 21, 1958. Mr. Corrington again reported on his study of tracing distortion of the two systems, and Arthur C. Haddy of London Decca told of the results of the Zurich Meeting. The EIA engineers then voted in favor of the 45°-45° system. Thus, within a period of four months after the initial demonstrations, standardization was achieved. This is a tribute to the cooperation of engineers both here and abroad, and to their strong desire that there should be one and only one stereophonic disc standard.

MAGNETIC TAPE STANDARD

In preparing the standard for magnetic tape records it was decided that there should be a single standard bulletin complete in all details. Two modes of operation, endless loop and reel to reel, were deemed to be adequate since the mode of operation for cartridges, such as the coplanar type, is the same as that for reel to reel. Both monophonic and stereophonic recordings are covered. Sequence of track utilization, recorded track width and placement are specified for two, four, and eight track recordings on tape ¼ in. wide. Reproducing characteristics are specified for operating speeds of 7½ and 3¾ ips. This standard, Bulletin E 5 entitled *Standards for Magnetic Tape Records,* was issued July 15, 1965. It is currently being revised to include four-track recordings on tape 0.150 in. wide at an operating speed of 1⅞ ips.

COORDINATION WITH EIA AND IEC

Close contact has always been maintained with EIA Engineering Committees. Many of the RIAA engineers are also members of EIA committees. Changes, particularly those dealing with dimensional specifications, are only made when approved by EIA engineers. It is customary to submit RIAA proposals to EIA and recommend that they also be adopted as EIA standards.

RIAA has participated actively with the International Electrotechnical Commission (IEC) and attended committee meetings on disc and magnetic recording standards regularly since 1958. This participation has been advantageous to all involved, for recorded music standards are of international concern. The USA attendance has resulted in many valuable contacts and a better understanding of individual problems. Many USA proposals have been accepted and adopted as IEC standards. In general, differences between IEC and USA recording standards are slight, although some do exist. The RIAA-EIA disc standards meet the IEC requirements, but the reverse is not true. For example, IEC permits a minimum thickness at the center hole area of 0.053 in. for LP records. The RIAA-EIA standard specifies a minimum of 0.060 in. In order to resolve this difference RIAA members provided records having the 0.053 in. thickness to record changer manufactuers for their evaluation. Some found the thin records acceptable but others experienced double dropping, i.e., two records at a time. As a result the USA standard has not been changed. Evidently the difference is not a serious one, for double dropping does not appear to be an operating problem.

CONCLUSION

The RIAA Engineering Committee is a small active group that has responded well to the responsibility of preparing standards for disc and magnetic tape records.* They have maintained a policy of concentrating on dimensional standards for interchangeability and avoided quality specifications, feeling that quality standards are a manufacturer's prerogative. Through active participation with other standards groups, both nationally and internationally, they have kept themselves continually aware and informed of new concepts in the recording field.

REFERENCES

1. *Processed Disk Records and Reproducing Equipment*, IEC Publication 98, second edition, 1964.
2. Murlan S. Corrington and T. Murakami, "Tracing Distortion in Stereophonic Disc Recordings", *RCA Review* **19**, 216 (1958).

* For copies of RIAA Standards contact Mr. Henry Brief, Executive Secretary, Record Industry Association of America, One East 57th Street, New York, N. Y. 10022.

THE AUTHOR

H. E. Roys was born in Beaver Falls, Pennsylvania, in 1902. He received his B.S. degree in electrical engineering from the University of Colorado in 1925 and went directly to work for the General Electric Company in Schenectady, New York in the Radio Department. Not long afterward Mr. Roys with others from the radio departments of General Electric and Westinghouse were transferred to Camden, New Jersey and in 1930 became part of the RCA Victor Company.

Mr. Roys was soon involved with phonographs and from then on has spent the majority of his time in the development of disc and magnetic recording. He became chief engineer of the RCA Victor Record Division in 1956, a position he held until late 1966. He was chief technical administrator of the RCA Victor Record Division until his retirement at the end of Jan. 1967.

Mr. Roys has been active in the standards field for over twenty years. He participated in establishing some of the early EIA and NAB standards. Prior to retirement he was chairman of EIA P 8 Recording and Reproducing System Components committee; chairman of USASI S4 Sound Recording committee; a US National Representative for IEC SC 29A Sound Recording committee as well as being a member of the RIAA Engineering Committee.

Mr. Roys is a fellow of the Institute of Electrical and Electronic Engineers, the Acoustical Society of America, and the Audio Engineering Society.

Weighted Peak Flutter Measurement—A Summary of the New IEEE Standard *

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The existing USA "flutter content" standard (IEEE 193-1953 and ANSI Z57.1-1953) has been replaced by a new "weighted peak flutter" standard (IEEE 193-1971 and ANSI S4.3-Draft). It standardizes the flutter meter specifications, the measuring procedures, and the form for reporting results. Technical requirements are identical with those of CCIR and German standards and an IEC draft.

INTRODUCTION: A newly published IEEE standard 193-1971, "Method of Measurement for Weighted Peak Flutter of Sound Recording and Reproducing Equipment" [1] is now available, and this short paper summarizes its contents. The technical background of the new standard is to be published in the near future [2], [3].

The advantages of this new method over that specified in the previous IEEE and ANSI standard [4] are as follows.

1) The ranking of the degradation of sound quality due to flutter, when measured objectively with the weighted peak flutter measurement, will predict fairly well that which would be given by a listening panel judging "subjective flutter." (The measurement of "flutter content" [4] bore little relationship to how a recorder would sound.)

2) The requirements for the measuring equipment, the flutter meter, that are given in the new standard are sufficiently complete that different equipment built to this standard will not only give the same readings on a calibrating sine wave, but will also give the same readings on a dynamic flutter waveform. (The previous standard [4] gave only general ranges for requirements, and no specific requirements for the dynamic response.)

3) Measurements according to the new standard are identical to those used internationally in standards of the CCIR and in the IEC draft, and to the German standard which has been widely used in Europe. This greatly enhances the exchange of information on recorder performance and facilitates sales and purchases of equipment in overseas areas.

SCOPE

"This standard specifies the weighted peak method of measurement for predicting subjective flutter of sound recorders and reproducers for normal audio usage" [1, sec. 1].

* Presented October 7, 1971 at the 41st Convention of the Audio Engineering Society, New York.

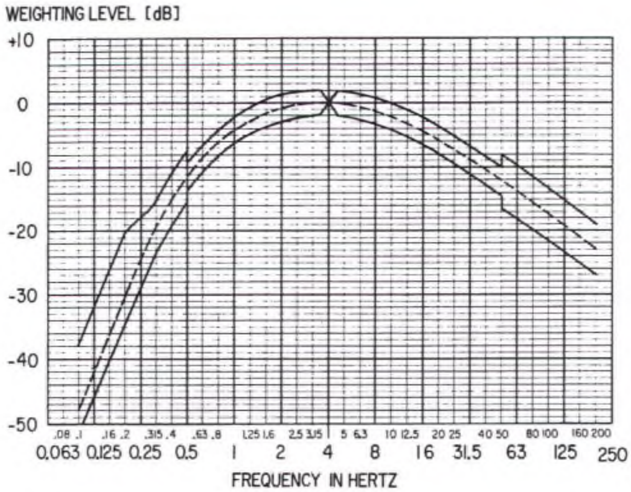

Fig. 1. Weighting curve.

DEFINITIONS

"Flutter, wow, drift, and frequency-modulation noise are all forms of distortion caused by undesired frequency modulation introduced into the signal by an irregular motion of the recording medium during the recording, duplicating, and/or reproducing processes" [1, sec. 2]. Although flutter, wow, drift, and frequency-modulation (friction) noise ("scrape flutter") are defined, the standard covers only the measurement of weighted peak flutter.

Weighting is defined as "the use of a psychoacoustically determined time response and frequency response in an objective measuring equipment. This is done in order to obtain indications which better predict the subjective values than would wideband measurement with a meter having either an instantaneous time response or a long-time average or rms response."

Weighted peak flutter is defined as "flutter and wow indicated by the weighted peak flutter measuring equipment specified in IEEE standard 193-1971" [1, sec. 2].

THE FLUTTER METER

The Flutter Meter Specification

"The measuring equipment shall consist of a frequency demodulator which produces an output voltage proportional to the relative frequency change ($\Delta f/f$), followed by a weighting filter, a peak rectifier, and an indicator" [1, sec. 5].

The test frequency now specified is the "preferred frequency" of 3150 Hz. The response curve of the combination of the demodulator, the weighting filter, and the indicator is to be as shown in Fig. 1. A peak-to-peak rectifier is used, but the meter is calibrated in the peak value (one half of the peak-to-peak value).

The dynamic characteristics of the flutter meter are

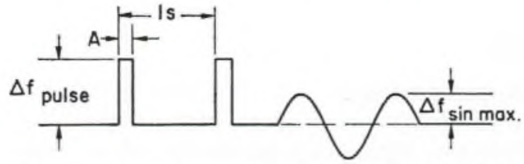

Fig. 2. Pulse for measuring dynamic characteristics.

specified in terms of the indication for a pulse train of frequency modulation as shown in Fig. 2. The pulses have constant amplitude, constant 1-s repetition rate, and adjustable length of 10 to 100 ms. They have the same peak-to-peak amplitude as the 4-Hz sine wave. The flutter meter reading with the sine wave of frequency modulation is taken as reference (100%). Then the relative meter readings are measured for the pulse train of frequency modulation. The flutter meter readings must be as shown below (tolerances are also given in the standard).

Pulse Length A (ms)	10	30	60	100
Relative Indication B (%)	21	62	90	100

The other dynamic requirement is for the decay time. When the 100-ms pulse is used with a 1-s repetition rate, the decay time of the flutter meter must be such that between the pulses the indicator falls to a reading of from 36 to 44% of the maximum.

A number of "good engineering practice" items are given; the instrument should work with test frequencies between 3000 and 3300 Hz in order to allow use with off-speed recorders or reproducers, and also with both old test records at 3000 Hz and new test records at 3150 Hz. A basic accuracy of at least plus or minus 10% of full scale is suggested. A required input voltage of not more than 100 mV is suggested, and an input impedance of not less than 300 kilohms at 3150 Hz. Finally, provision for connecting external equipment (for example, an oscillograph) with or without the weighting filter is suggested.

Availability of Flutter Meters and Test Records

Since the "weighted peak flutter" measurement is identical to the "DIN weighted" measurement which has been used for some 10 years now, commercial flutter meters which measure to the German standard DIN 45 405-1966 may be used. Such instruments are available, for instance from Gotham Audio Corporation, New York, N.Y., and Mincom Division of 3M, Camarillo, Calif. (The Mincom instruments were developed by Bahrs Industries and manufactured originally by Micom, later called DMC).

Test records with a 3150-Hz signal which may be used for flutter measurements according to the new standard may be obtained, for instance, from the following companies:

Tape records:	Ampex Corporation, Redwood City, Calif. Standard Tape Laboratory, Oakland, Calif.
16-mm and 35-mm motion picture film records:	Standard Tape Laboratory, Oakland, Calif.
Disc records:	Gotham Audio Corporation, New York, N. Y.

MEASUREMENT PROCEDURE

"The measurements of normal recording and reproducing systems shall be made on one element only of the system (either the recorder or the reproducer, but not on both) under such conditions that the weighted peak flutter in the remaining parts of the measuring system is negligible.

When this condition cannot be fulfilled, a recorder/reproducer may be measured by recording a 3150-Hz test frequency and subsequently reproducing this record several times, measuring in each case the total weighted peak flutter and calculating the arithmetic average value of these measurements. Weighted peak flutter shall not be measured while simultaneoulsy recording and reproducing" [1, sec. 3.3].

If, because of random flutter or very low-frequency flutter, the reading varies with time, the maximum value shall be read and reported. Since in most systems system conditions vary in such a manner as to give different flutter readings, a choice of reporting forms is given. Either report the reading for each condition, or else give the reading for the worst combination of factors.

REPORTING RESULTS

Weighted peak flutter should be reported in the following manner.

"Weighted peak flutter of the recorder [reproducer] [recording and reproducing system]: ± %" [1, sec. 4].

The sign ± is used to indicate that the peak rather than peak-to-peak value has been given.

A statement of conditions may also be required; for example, for a tape recorder, the speed and the reel size (minimum hub diameter, maximum outside diameter, etc.).

REFERENCES

[1] "Method of Measurement for Weighted Peak Flutter of Sound Recording and Reproducing Equipment," IEEE Standard 193-1971 and ANSI S4.3-Draft, available from IEEE, N.Y., N.Y. Price: $3.00; $2.25 to IEEE members. *IEEE Trans. Audio and Electroacoust.*, vol. AU-20, no. 1 (1972) (to be published).

[2] E. Belger, "On Measuring Frequency Variations," *IEEE Trans. Audio and Electroacoust.*, vol. AU-20, no. 1 (1972) (to be published).

[3] J. G. McKnight, "Development of a Standard Measurement to Predict Subjective Fluttering," *IEEE Trans. Audio and Electroacoust.*, vol. AU-20, no. 1 (1972) (to be published).

[4] "Methods for Determining Flutter Content in Sound Recorders," IEEE Standard 193-1953 and *ANSI* Z57.1-1954.

THE AUTHOR

John G. (Jay) McKnight was born in Seattle in 1931. He received a B.S. in electrical engineering from Stanford University in 1952.

He has been with Ampex Corporation since 1953, with the exception of the years 1954-56 when he was assigned to the engineering staff of the U.S. Armed Forces Radio Service in New York. McKnight has worked for the Professional Audio Division and the Stereo Tape (now Music) Division, and is now a Member of the Research Staff, Magnetic Recording Group, of the Research and Advanced Technology Division.

Mr. McKnight's work has been in research and engineering concerning the dynamics of tape transports; magnetic recording, especially as it concerns the recording of music; and tape recording and audio systems standardization. He has presented and published papers on amplitudes vs. frequency in music, noise considerations and measurements in magnetic recording, equalization in magnetic recording, measurements of signals recorded on magnetic tape, stereophonic recording, and transport speed variations (flutter). He is an amateur musician, playing viola in the Peninsula Symphony Orchestra.

He is a Fellow of the AES, recipient of the AES award, a member of its Editorial Board, a Governor two times, and Chairman of its Standards Committee; a Senior Member of IEEE, member of the G-AE Adcom, member of the IEEE Transactions on Audio and Electroacoustics Editorial Board, and Chairman of the IEEE Standards Sub-Committee on Recording and Reproducing; and a member of standards committees on audio engineering and magnetic sound recording of the American National Standards Institute, the NAB, the EIA, the IEC, the SMPTE, the RIAA, and the CCIR.

LETTERS TO THE EDITOR

THE BENEFITS AND DANGERS OF STANDARDIZATION

The Problem

The recording industry is admittedly within reach of a new technology in the recording process, a technology which has been with us now in other signal processing aspects of audio since 1971. The digital recording form, as such, is fairly old. Its applicability to recording on magnetic tape is likewise many years old. Its appeal to our industry is recent only because its economics appear to be coming within range for the entertainment world. Admittedly, the introduction of digital technology into any phase of recording which requires interchangeability is problematical. But that is nothing new. How long did it take for a U.S. tape machine manufacturer to build a machine capable of also playing back and recording the CCIR tape recording characteristic, or a European one of playing an NAB tape? A mere matter of decades! Such incompatibility has been with us for as long as this industry has existed, and it will continue to be with us. Just try to take your television set to Europe and see what you can tune in. In short, utopia is as far away as ever.

The Dream

A world in which everyone uses the same format, language, sampling rate; in short every tape recorded digitally anywhere is playable everywhere.

The Nightmare

That comes 10 years later when we find out that we knew too little at the time we "standardized" to make a long lasting and intelligent decision.

The Role of the AES

The AES is undoubtedly well intentioned, but, in the opinion of the writer, ill advised. I believe that every topic which concerns audio engineering should be aired within forums of the AES *provided* that such discussion never implies a *search for consensus*. I believe that in this phrase there hides the true danger, for consensus invariably invites its sister phrase the "lowest common denominator." Study groups, I believe, are worthwhile and useful. They, by contrast to standards committees, may be composed of people seeking enlightenment. In other words, they may be tutorial in nature. Whenever such a group gives the impression that it is acting as a go-between among competitive interests, I believe it is not acting in the interest of the public which, were it to choose between compatibility on the one hand and diversification of knowledge and competitive engineering on the other, would be well advised to choose the latter.

The Role of the Consumer

It is interesting to note that the consumer really does not get consulted in these deliberations. And frankly, all that will really be in it for him is higher prices. It is a well-known fact that the principal complaints about music in the home are noisy pressings and poor cassette duplicates. I am afraid neither of these will be significantly improved through the use of digital technology in the studio!

The Benefits of Incompatibility

I believe that incompatibility has benefits to the consumer: it tends to discourage those technologies not worthy of his purchase. Just look at quad records as a good example of just such a case. But video cassette recorders, in at least four incompatible formats, are doing just great. If you want an example of incompatibility which was remedied long after the product appeared on the market place, look at 1-inch helical-scan video tape recorders, where a standard was agreed upon between two *marketed* conflicting systems.

Conclusion

I feel strongly that until digital becomes an all encompassing technology from the microphone right into the home, the possible benefits it holds for the consumer are miniscule. But that fact in and of itself does not justify, to my way of thinking, an effort which will likely stifle competition and creative thinking on a subject which needs innovation much more than it needs standardization for the foreseeable future. Perhaps in 10 or 15 years, we may be able to see more clearly where it is all going. Right now, I am afraid that our ego is running away with our usually good judgment.

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On RIAA Equalization Networks*[1]

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Most current disk preamplifiers, including some very expensive models, have audibly inaccurate RIAA equalization. This severely limits any conclusions that can be drawn from A/B testing of such preamplifiers. These errors are due in part to the perpetuation in print of incorrect formulas for the design of the RIAA equalization networks commonly employed. Other factors include the existence of an uncorrected high-frequency zero too close to the top of the audio band in many noninverting designs, and failure to take adequate account of the limited available loop gain. The situation is surveyed, and tables taking in account the above problems are given for the design of both inverting and noninverting RIAA deemphasis and preemphasis circuits. Examples are furnished to illustrate the various configurations.

0. INTRODUCTION

This paper has been stimulated by the writer's experiences with disk preamplifiers over the past few years. As readers will be aware, many hypothetical causes have been put forward for the subjectively perceived differences between such preamplifiers when A/B tested against each other, and much mystique currently surrounds their design and evaluation. One fact, however, is indisputable, and that is that frequency response differences exceeding a few tenths of a decibel in magnitude between disk preamplifiers are audible. Such deviations tend to be broad band in extent, since they arise from gain and component errors within the RIAA deemphasis circuit. After examining many disk preamplifiers it has become apparent to the writer that this is a problem of significant, if not major, proportions. It is, moreover, not confined to lower priced components only. Some of the most expensive and highly regarded disk preamplifiers on the market deviate audibly from correct RIAA equalization.

There seem to be three major causes for these errors.

1) What the writer, after examining numerous books and schematic diagrams, can only put down to the use of incorrect design equations for the calculation of the resistor and capacitor values used in the equalization networks.

2) Failure to take into account the fact that there is an additional high-frequency corner in the response of an equalized noninverting amplifier stage (the almost universally used configuration), which causes its response to deviate at high frequencies from that required by the RIAA curve. If this corner is placed too close to the top of the audio band, and no corrective action is taken, audible deviations will occur at high audio frequencies.

3) Failure to correctly take into account the limited loop gain available from the amplifier circuit. Many discrete disk preamplifiers have a loop gain at low frequencies which is inadequate to cause them to adhere to the low-frequency portion of the RIAA curve, while many integrated operational amplifiers display insufficient high-frequency loop gain due to their low gain–bandwidth products.

We shall comment further on these points in the sequel. Point 1) is perhaps the most surprising, for there is nothing

* Manuscript received 1977 Oct. 21; revised and presented at the 61st Convention of the Audio Engineering Society, New York, 1978 Nov. 3–6; revised 1978 Dec. 13.

[1] Since the presentation of this paper to the 61st Convention of the Audio Engineering Society in New York on 1978 November 6 the existence of an as yet unpublished thesis on these topics by Paul Skritek ("Optimierung von Entzerrernetzwerken," Diplomarbeit, Institut für Allgemeine Elektrotechnik, Technische Universität, Vienna, Austria, 1977 March) has been brought to the writer's attention. Skritek's approach is somewhat different, but his thesis includes most of the results reported here. In fact, he develops formulas for a very wide class of networks, including all the RIAA cases, and also considers the effects of finite amplifier gain and gain–bandwidth on their performance. The writer has also discovered an article by James Sugden ("Equalizasion," *Audio Annual 1968* (Link House Publications, Croydon, U.K., 1968), pp. 34–37) which correctly develops formulas for two cases of passive and active networks, and which, after Skritek's thesis and [33], represents the single most comprehensive treatment of RIAA equalization networks yet found in print.

extraordinarily difficult about analyzing the standard RIAA equalization configurations.

In case the reader feels that the writer is grossly exaggerating the widespread nature of the problem, we would like to refer him to [1]–[18], drawn from many diverse sources, in support of our contention.[2] As will shortly become apparent, these circuits all suffer from one or more of maladies 1)–3) without showing signs of any adequate corrective action having been taken. All is, however, not bleak, for we have come across a few circuits which do correct for some or all of these sources of error. Without wanting to play favorites, we list some of these circuits [19]–[29], but they are few and far between.

This paper is intended to answer points 1)–3) by providing design formulas for RIAA networks used both passively and actively around inverting or noninverting amplifier stages, and will also give some guidelines for those cases when the loop gain is insufficient for this factor to be ignored. A search of the literature has failed to turn up much in the way of correct formulas; the only sources found which correctly treat a few particular aspects of the problem are [30]–[33]. (See also footnote 1.) It would therefore appear that the time is ripe for a discussion of this topic in some detail. It is hoped that this paper will help fill the gap.

1. THE CIRCUITS AND THEIR CHARACTERISTICS

As is well known, the RIAA disk recording/reproduction standard specifies equalization time constants of $T_3 = 3180\ \mu s$, $T_4 = 318\ \mu s$, and $T_5 = 75\ \mu s$, corresponding to turnover frequencies $f_3 = 50.05$ Hz, $f_4 = 500.5$ Hz, and $f_5 = 2122$ Hz, respectively.[3] The recent IEC amendment [34] to this standard, not yet adopted by the RIAA, adds a further rolloff of time constant $T_2 = 7950\ \mu s$, corresponding to a frequency of $f_2 = 20.02$ Hz, which is applied only on replay. (The reason for this apparently strange nomenclature will shortly become apparent.) Such equalization is commonly achieved by means of frequency-dependent negative feedback around the disk preamplifier stages. The feedback network generally incorporates one of the four electrically equivalent R/C networks N, shown in Fig. 1, for this purpose. The four networks N are listed in the order of popularity, that of Fig. 1(a) being the most popular configuration, while that of Fig. 1(d) is the least frequently used. Also given are their complex impedance formulas, which are easily calculated (see, for example, [35] or [36]). Throughout this paper we shall assume that the components are labeled such that $R_1 > R_2$ and $C_1 > C_2$. (This results in the apparently "reversed" labeling of network 1(c).) Thus $R_1C_1 > R_2C_2$, and so R_1 and C_1 principally determine T_3, while R_2 and C_2 principally determine T_5.

The networks N can be used actively or passively to perform RIAA pre- or deemphasis functions. Of the possible configurations those which appear to be of the most practical utility are listed in Figs. 2–5.[4] Also shown in Figs. 2–5 is the stylized frequency response (Bode plot) of each configuration, $G(\omega)$ representing the magnitude of the gain at angular frequency ω. At this stage it is assumed that the amplifier shown has infinite open-loop gain and can be treated as an ideal operational amplifier. We shall comment later on the very real restrictions and modifications that are necessitated by practical circuits which do not meet these ideal requirements. Two points are at once apparent.

1) There is an additional unavoidable high-frequency turnover with time constant T_6 (corresponding to a frequency f_6, say) which appears in Fig. 3 even when $R_3 = 0$. This departure from the ideal RIAA deemphasis curve does not arise in the inverting case (Fig. 2), unless we deliberately set $R_3 \neq 0$. As mentioned in the Introduction, the appearance of f_6 has almost universally been ignored in practice. While this is not serious if f_6 is at least two octaves above the audio band, this is frequently not the case, as an examination of the circuits cited in the Introduction will show. We shall see, however, that f_6 can be *exactly* compensated for by adding a passive single-pole R/C low-pass filter at the output of the equalized preamplifier, and thus need not concern us unduly. Another reason for wishing to continue the 6-dB per octave RIAA deemphasis beyond f_6 is to prevent ultrasonic signals (from either tracing distortion or radio-frequency pickup) from reaching subsequent possibly slew-rate-limited stages in the chain.[5] It should also be pointed out that the inclusion of R_3 in any of the active circuits under consideration may be *necessary* to enable them to be stabilized.

2) The addition of capacitor C_0 introduces a further pole/zero pair, namely, ω_1 and ω_2, which provides a low-frequency rolloff in the circuits of Figs. 2 and 3 and thus enables a degree of infrasonic filtering of warp and rumble signals to be achieved. If T_2 is chosen equal to $7950\ \mu s$, the inclusion of C_0 will provide equalization as required by the IEC amendment [34]. The reasons behind our labeling of the RIAA time constants T_3–T_5 is now clear. We shall always assume that $T_1 > T_2 > T_3 > T_4 > T_5 > T_6$.

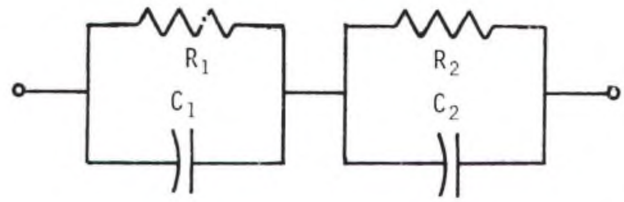
The notes appended to the circuits of Figs. 2–5 will be seen to follow from our calculations in the next three sections. They also refer to the appropriate design table to be

[2] The writer would like to express his appreciation to Walter G. Jung for kindly furnishing him with many of the references cited.

[3] We shall consistently use the symbol f_i to refer to the frequency and ω_i to the angular frequency of a pole/zero of time constant T_i. These quantities are related by $\omega_i = 2\pi f_i$, $\omega_i = 1/T_i$, $i = 1, \cdots, 7$.

[4] The two remaining possibilities which have been omitted for practical reasons are 1) active noninverting preemphasis circuit (this is not feasible due to the enormous HF open-loop gain requirement necessitated by the fact that the minimum signal gain is unity) and 2) passive deemphasis circuit (its wide variation in output impedance renders the circuit of Fig. 2(a) preferable, especially since gain is required in any case); but see Section 8.

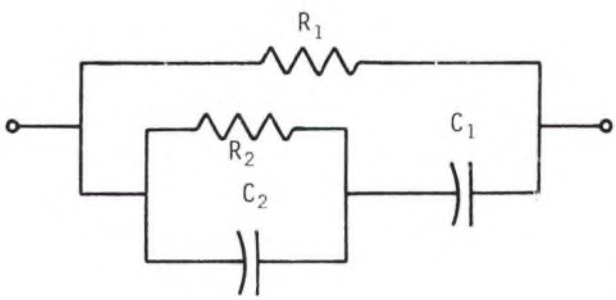
[5] For the same reason, the presence of the T_6 corner in the preemphasis circuits of Figs. 4 and 5 is desirable, provided that it lies at least two octaves above the audio band. One cannot continue preemphasizing at 6 dB per octave much beyond this point. Hence R_3 *should* be used in the circuit of Fig. 4, with T_6 carefully chosen.



$$Z(s) = \frac{(R_1 + R_2)\left[1 + \frac{R_1 R_2}{R_1 + R_2}\,(C_1 + C_2)s\right]}{(1 + R_1 C_1 s)(1 + R_2 C_2 s)}$$

$$R_A = R_1 + R_2$$

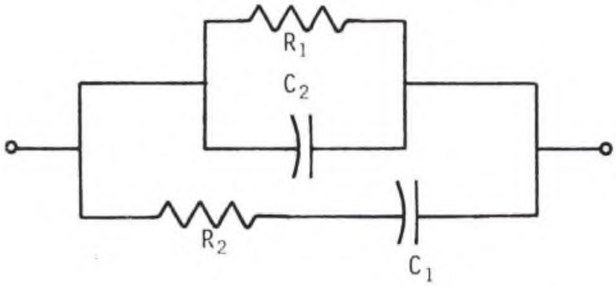
(a)



$$Z(s) = \frac{R_1[1 + R_2(C_1 + C_2)s]}{1 + [R_1 C_1 + R_2(C_1 + C_2)]s + R_1 C_1 R_2 C_2 s^2}$$

$$R_A = R_1$$

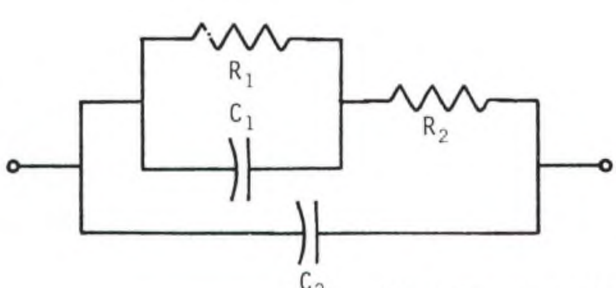
(b)



$$Z(s) = \frac{R_1(1 + R_2 C_1 s)}{1 + [R_1(C_1 + C_2) + R_2 C_1]s + R_1 C_1 R_2 C_2 s^2}$$

$$R_A = R_1$$

(c)



$$Z(s) = \frac{(R_1 + R_2)\left[1 + \frac{R_1 R_2}{R_1 + R_2}\,C_1 s\right]}{1 + [R_1(C_1 + C_2) + R_2 C_2]s + R_1 C_1 R_2 C_2 s^2}$$

$$R_A = R_1 + R_2$$

(d)

Fig. 1. Four commonly used equalization networks N.

used for each configuration, and it will be our purpose in these sections to derive the appropriate formulas upon which these tables are based.

2. CALCULATING THE POLES AND ZEROS FOR FIG. 2

In this section we analyze the inverting deemphasis configurations of Fig. 2.[6] The case $R_3 = 0$ will be referred to as the "ideal case," since it is the only one which avoids the undesirable high-frequency zero at ω_6. We thus write down the signal gain equation for the (complex) signal gain $G(s)$ of this circuit (assuming infinite open-loop gain) and find

$$G(s) = -\frac{Z(s) + R_3}{R_0 + 1/(C_0 s)} = -\frac{\{Z(s) + R_3\}C_0 s}{1 + R_0 C_0 s} \quad (1)$$

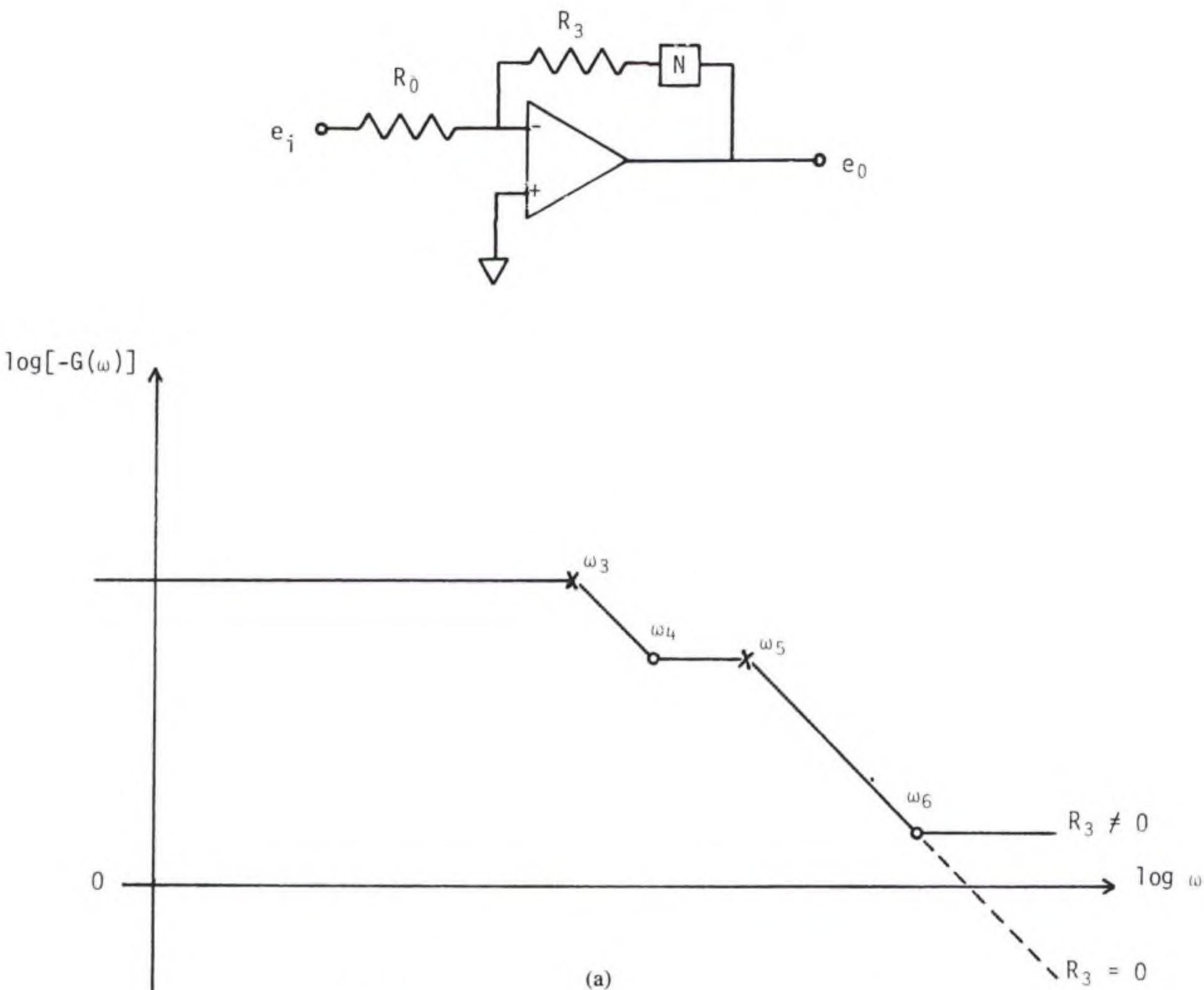
where $Z(s)$ refers to the impedance formulas for the networks N given in Fig. 1. [The case in which C_0 is not present may be obtained by letting $C_0 \to \infty$ in Eq. (1).] Alternately, we may express $G(s)$ in terms of the time constants T_2–T_6 as

$$G(s) = -\frac{R_A + R_3}{R_0} \cdot \frac{T_2 s(1 + T_4 s)(1 + T_6 s)}{(1 + T_2 s)(1 + T_3 s)(1 + T_5 s)} \quad (2)$$

where the resistance R_A, introduced in Fig. 1, represents the resistance of the network N at 0 Hz (its dc resistance).

Equating the right-hand sides of Eqs. (1) and (2) we can, for each of the four networks of Fig. 1, solve for T_2–T_6 in terms of R_0, R_1, R_2, R_3, C_0, C_1, C_2, thus obtaining formulas for the actually realized time constants of this configuration, and more usefully, we can solve for the

[6] The reader is asked to bear with us through this analysis, for the more common noninverting configurations of Fig. 3 will turn out to be reducible to those of Fig. 2, and the latter are easier to analyze first.

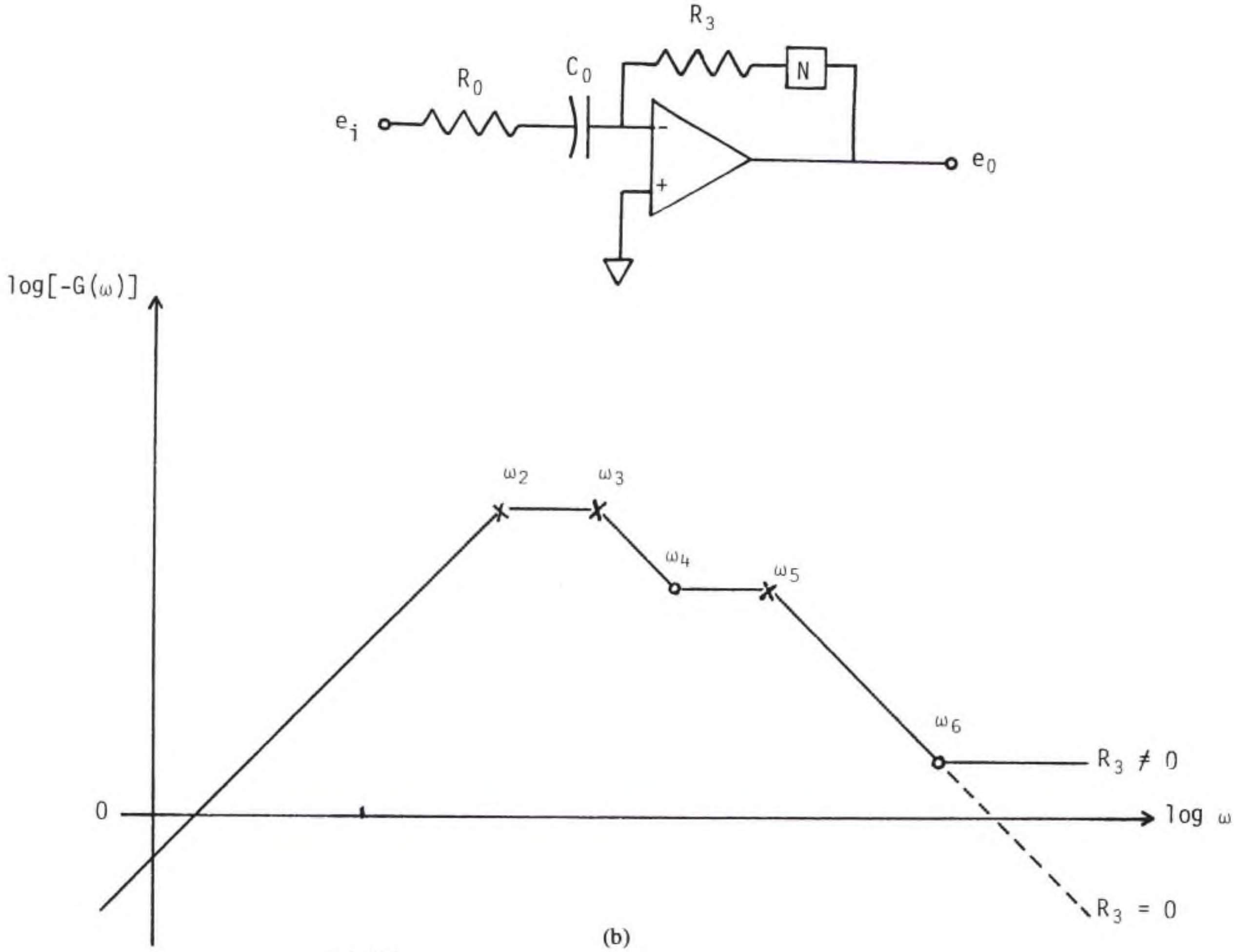


NOTES:
1) R_0 includes source resistance.
2) If $R_3 \neq 0$: Both R_0 and R_3 can be chosen independently. R_0 alone enables gain adjustment without affecting frequency response; R_3 alone determines ω_4, ω_6, while ω_3, ω_5 are not affected by changing R_0, R_3. The ω_6 corner is passively correctable.
3) If $R_3 = 0$: R_0 and *one* of the components of N can be chosen independently. R_0 alone enables gain adjustment without affecting frequency response.
4) Use Tables 1 and 2.

Fig. 2. Active inverting deemphasis circuit. (a) Without C_0.

resistor and capacitor values of the network components in terms of T_2–T_6.[7] These latter formulas can be used in the design of the networks to fulfill the required RIAA function. A different set of formulas results in the case of each of the four networks of Fig. 1. An example of the rather elaborate calculations involved is given in Appendix 1 for the case of the network of Fig. 1(a). The other cases are somewhat more complicated. The results are summarized in Table 1(a)–(d), referring, respectively, to the networks of Fig. 1(a)–(d). The first column in Table 1 gives the design formulas for the ideal case $R_3 = 0$, and the second column lists the corresponding formulas when $R_3 \neq 0$. For simplicity, some of these formulas are given in an approximate form only in the second column. These approximations are to first order in R_3, and are valid to a very high degree of accuracy, provided $R_3 << R_2$, a situation occurring in practice. Table 2 gives the formulas for the magnitude $G(\omega)$ of the complex gain $G(s)$ at angular frequency ω, and is to be used in conjunction with Table 1 in the design process. The design notes appended to Fig. 2 now become relevant. In solving for the formulas given, it is found that both R_0 and R_3 (if nonzero) can be chosen independently. For this reason the formulas in the middle third of Table 1 are "normalized" to give each of the unknown quantities R_1, R_2, C_0, C_1, C_2 in terms of R_0 and R_3 *only,* assuming that the T_i have been chosen in any particular case. Practical design is thus simplified. We shall have more to say about this aspect later. Of considerable significance are the formulas in the first column of

[7] Note that the zeros and poles all lie on the negative real axis in the complex frequency plane.



(b)

NOTES:
1) R_0 includes source resistance.
2) If $R_3 \neq 0$: Both R_0 and R_3 can be chosen independently. Changing R_0 adjusts gain but also affects ω_2. R_3 alone affects ω_4, ω_6, while ω_3, ω_5 are not affected by changing R_0, R_3. The ω_6 corner is passively correctable.
3) If $R_3 = 0$: R_0 and *one* of the components of N can be chosen independently. R_0 alone affects *both* gain and ω_2, but none of the other ω_i.
4) Use Tables 1 and 2.

Fig. 2. Active inverting deemphasis circuit. (b) With C_0.

Table 1, for they represent the ideal RIAA case, and are modified only slightly in numerical value when $R_3 \neq 0$. It should be noted that, not unexpectedly in this simple case ($R_3 = 0$), the formulas for T_3–T_5 are just precisely those for the time constants corresponding to the negative real zero and poles of the impedance expressions $Z(s)$ given in Fig. 1. They also point up what appears to be a very common error committed to print in some of the references cited in the Introduction, and clearly demonstrated by many of the circuits referred to there. For example, the following two situations are not uncommon, and will be found to be represented in the references cited:

1) Use of the network of Fig. 1(a) with the false design equations

$$R_1C_1 = T_3, \qquad R_2C_2 = T_5, \qquad R_2C_1 = T_4 = 318\ \mu\text{s}. \quad (3)$$

As we see from Table 1(a), in fact the network RC products should be (ignoring R_3)

$$R_1C_1 = T_3, \qquad R_2C_2 = T_5,$$

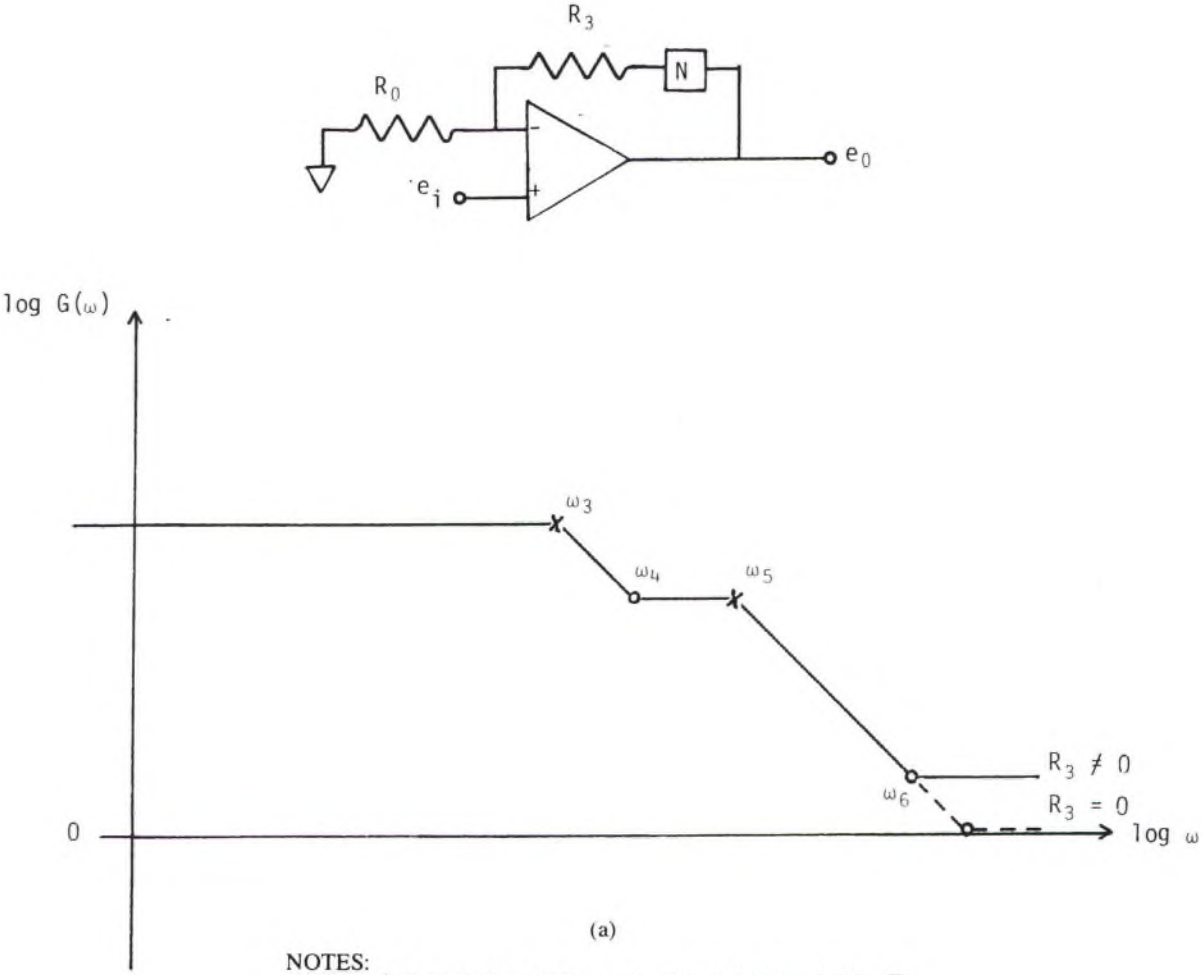
$$R_2C_1 = \frac{T_3(T_4 - T_5)}{T_3 - T_4} = 270\ \mu\text{s}$$

so that the last of formulas (3) is in error by a substantial 18%. This is a very common mistake. The correct formula for T_4, namely,

$$T_4 = \frac{R_1R_2}{R_1 + R_2}(C_1 + C_2)$$

is not difficult to remember, for it represents the time constant of the parallel combination of R_1 and R_2 with the parallel combination of C_1 and C_2.

2) Use of the network of Fig. 1(b) with $R_2C_2 = T_5 = 75\ \mu$s instead of the correct value (ignoring R_3)



(a)

NOTES:

1) If $R_3 \neq 0$: Both R_0 and R_3 can be chosen independently. To adjust gain without affecting frequency response, change R_3/R_0 while keeping $(R_0 + R_3)$ fixed. $(R_0 + R_3)$ alone determines ω_4, ω_6, while ω_3, ω_5 are not affected by changing R_0, R_3.
2) If $R_3 = 0$: Only R_0 can be chosen independently. Changing R_0 affects *both* gain and ω_4, ω_6, while ω_3, ω_5 are not affected by changing R_0.
3) The ω_6 corner is passively correctable.
4) Use Tables 1 and 2 with R_3 replaced by $(R_0 + R_3)$ wherever it occurs, and $G(\omega)$ replaced by $-G(\omega)$.

Fig. 3. Active noninverting deemphasis circuit. (a) Without C_0.

$$R_2C_2 = \frac{T_3T_5}{T_3 - T_4 + T_5} = 81.21\ \mu\text{s}.$$

This represents an error of -8%, which is not negligible.

It must thus be realized that the R/C subsections of the networks N interact in determining the overall poles and zeros, and hence the individual RC products for each subsection do *not* give the time constants of the overall network.

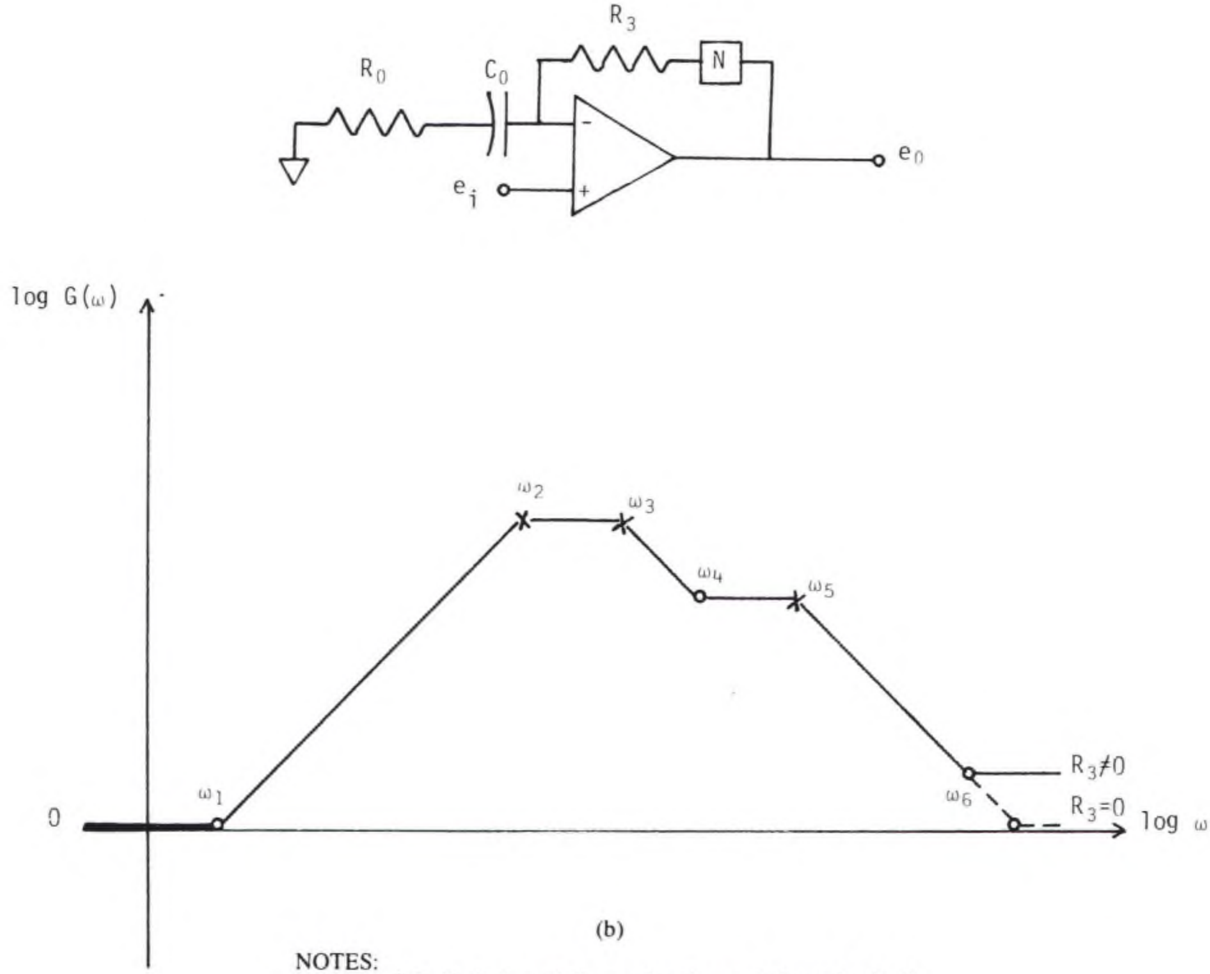
We have placed considerable emphasis on Tables 1 and 2, and for a good reason: With only a few substitutions they will also provide design formulas for the circuits of Figs. 3(a), 4, and 5. Only the circuit of Fig. 3(b) will require a different design table. In fact, since in general R_0 and R_3 should be very much less than R_2 in value, the first column of Table 1 serves as a fairly accurate prototype of the values that will apply in most practical situations. The symbol [·] is used in some of the formulas in Table 1 (and will also be used subsequently) to denote a repetition of the square-bracketed expression that precedes it within the same formula.

As a final note it should be remarked that T_2 is uncoupled from the other time constants T_3–T_6 in the sense that changing or removing C_0 affects only T_2, leaving T_3–T_6 unaltered. (This is not true for the circuit of Fig. 3(b).)

3. EXTENDING THE RESULTS TO FIGS. 3(a), 4, AND 5

The extension of the above results to the active inverting RIAA preemphasis circuit of Fig. 4 is immediate and obvious, for the simple replacement of $G(s)$ by $1/G(s)$ in our previous analysis converts it to this case (that is, the poles and zeros are interchanged). Thus as mentioned in the design notes in Fig. 4, Tables 1 and 2 are easily applied.

The active noninverting RIAA deemphasis circuit of Fig. 3(a) is not much more difficult to handle. For in this case (see Eq. (1)),



(b)

NOTES:
1) If $R_3 \neq 0$: Both R_0 and R_3 can be chosen independently if, say, ω_1 is considered to be dependent. To adjust gain, change R_3/R_0 while keeping $(R_0 + R_3)$ fixed; this also affects ω_2. $(R_0 + R_3)$ alone determines ω_1, ω_4, ω_6, while ω_3, ω_5 are not affected by changing R_0, R_3.
2) If $R_3 = 0$: Only R_0 can be chosen independently. Changing R_0 affects *both* gain and ω_1, ω_2, ω_4, ω_6, while ω_3, ω_5 are not affected by changing R_0.
3) The ω_6 corner is passively correctable.
4) Use Tables 3 and 4.

Fig. 3. Active noninverting deemphasis circuit. (b) With C_0.

$$G(s) = \frac{Z(s) + R_3}{R_0} + 1 = \frac{Z(s) + (R_0 + R_3)}{R_0} \tag{4}$$

which is just precisely the limiting form of Eq. (1) when $C_0 \to \infty$, if we replace R_3 in Eq. (1) by $(R_0 + R_3)$ and delete the minus sign on the right-hand side. Eq. (2) also now applies with the same changes, and so it follows at once that the design Tables 1 and 2 without C_0 also apply directly to the circuit of Fig. 3(a) under the simple substitutions

$$R_3 \to R_0 + R_3 \qquad \text{and} \qquad G(\omega) \to -G(\omega). \tag{5}$$

We see that the poles T_3, T_5 are exactly the same as those of Fig. 2(a); the zeros T_4, T_6 are, however, shifted by the change of R_3 to $(R_0 + R_3)$.

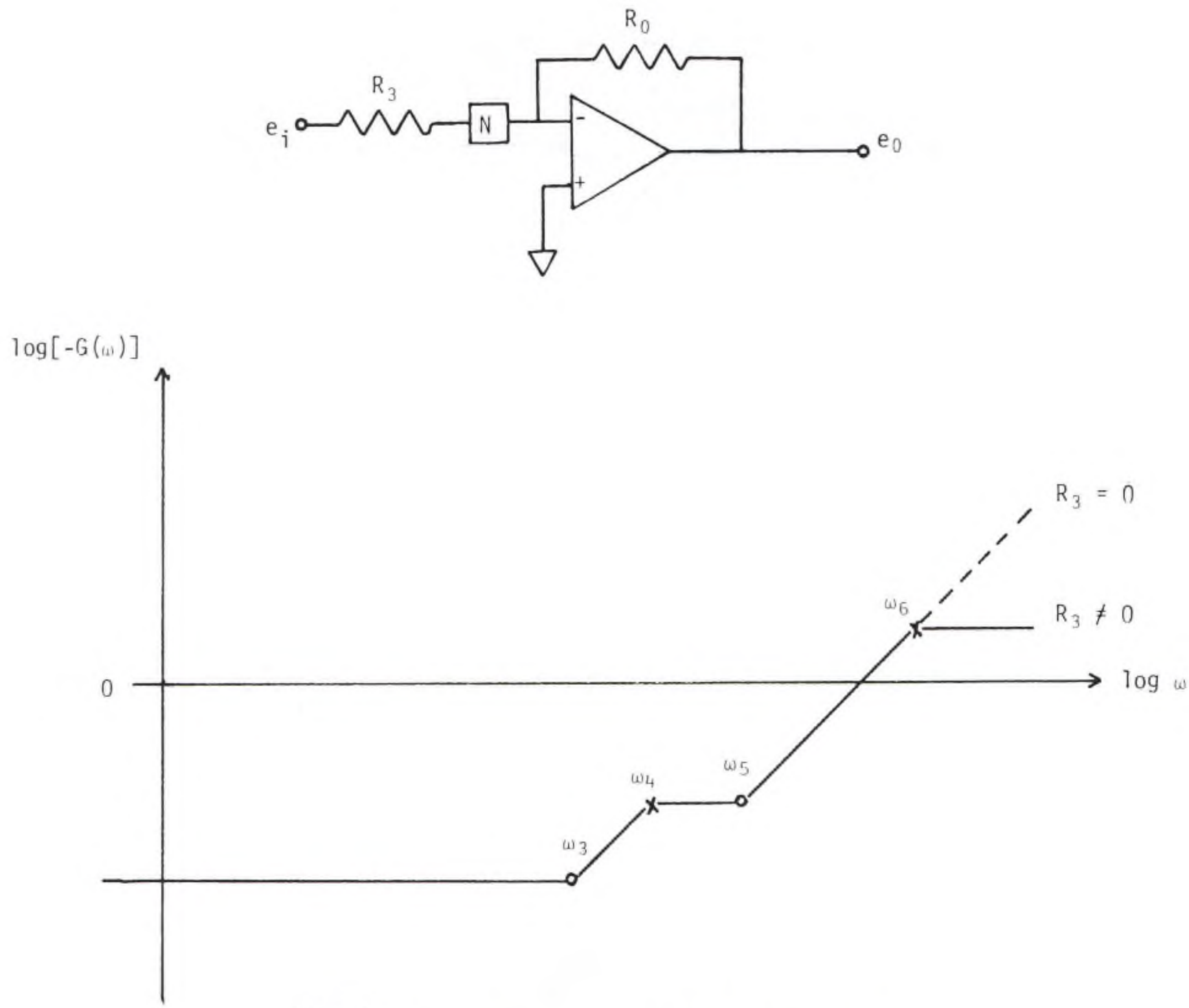
Similarly, the passive preemphasis circuit of Fig. 5 now follows easily from the case of Fig. 3(a), since its gain formula is just the reciprocal of Eq. (4). Thus its design equations also follow from Tables 1 and 2 without C_0 by making the substitutions

$$R_3 \to (R_0 + R_3) \qquad \text{and} \qquad G(\omega) \to -\frac{1}{G(\omega)}\cdot \tag{6}$$

Again, the design notes appended to Figs. 3(a), 4, and 5 should now begin to fall into place. In particular, note that both R_0 and R_3 (if nonzero) can be chosen independently in the design process. In view of the manner in which the time constants are affected by changing R_0 and R_3 in the case of the circuits of Figs. 3(a) and 5, it is preferable to think of the combinations $(R_0 + R_3)$ and R_3/R_0 [or $(R_0 + R_3)/R_0$] as being the independent quantities in these cases. This is so because of the appearance of $(R_0 + R_3)$ in the formulas of Tables 1 and 2 as a result of the substitutions (5) and (6).

4. THE CASE OF FIG. 3(b)

Fig. 3(b) requires a separate treatment. The signal gain formula now reads



NOTES:
1) R_3 includes source resistance and is required for stability. At high frequency the load on the source is R_3.
2) Both R_0 and R_3 can be chosen independently. R_0 alone enables gain adjustment without affecting frequency response; R_3 alone determines ω_4, ω_6, while ω_3, ω_5 are not affected by changing R_3.
3) Use Tables 1 and 2 with $G(\omega)$ replaced by $1/G(\omega)$.

Fig. 4. Active inverting preemphasis circuit.

$$G(s) = \frac{Z(s) + R_3}{R_0 + 1/(C_0 s)} + 1 = \frac{1 + \{Z(s) + (R_0 + R_3)\}C_0 s}{1 + R_0 C_0 s} \tag{7}$$

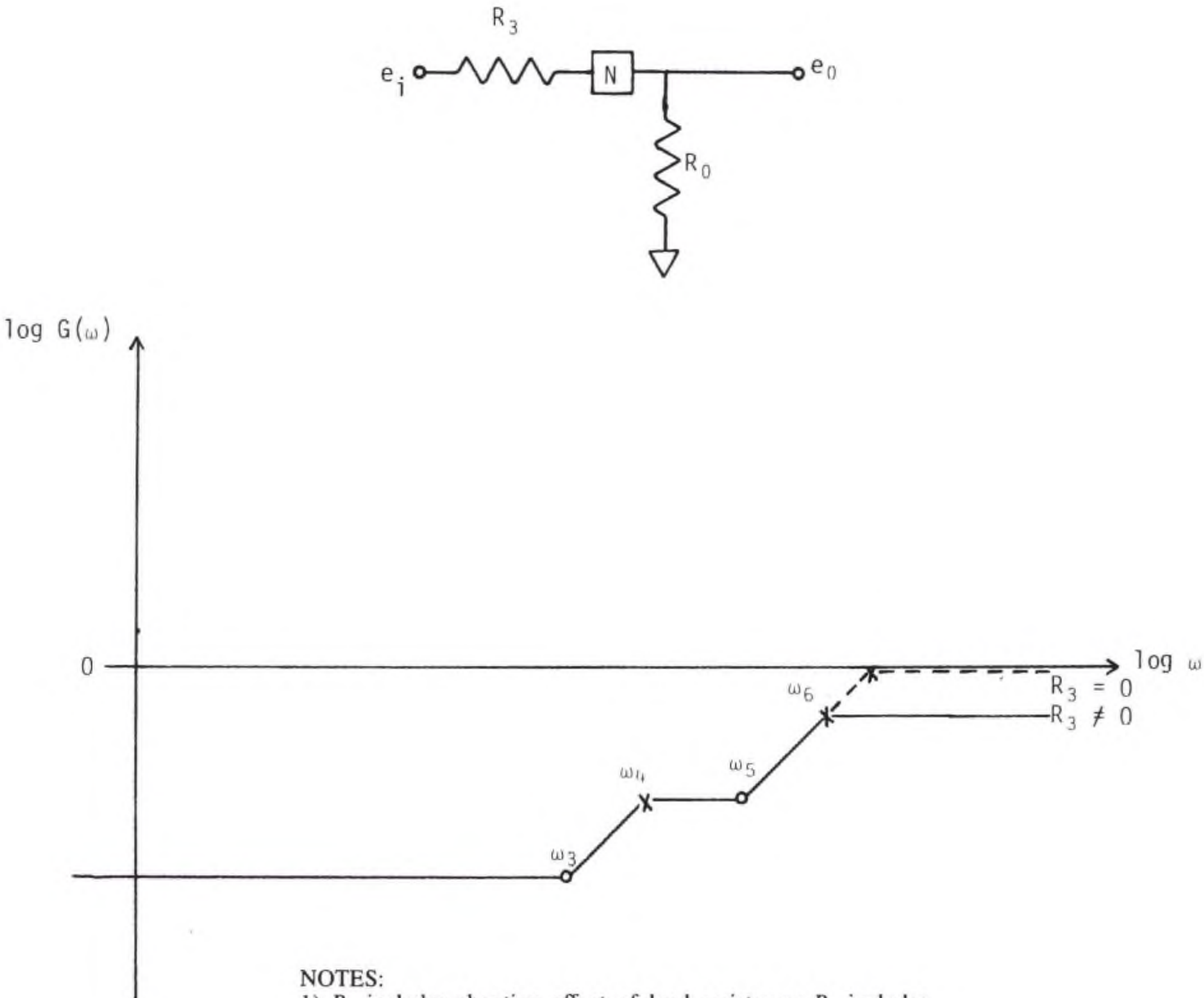
where $Z(s)$ is given in Fig. 1. In terms of the circuit time constants T_1–T_6, $G(s)$ can alternately be expressed as

$$G(s) = \frac{(1 + T_1 s)(1 + T_4 s)(1 + T_6 s)}{(1 + T_2 s)(1 + T_3 s)(1 + T_5 s)} \,. \tag{8}$$

As $C_0 \rightarrow \infty$, Eq. (7) reduces to Eq. (4), as expected. Once again, the poles T_2, T_3, T_5 are exactly the same as those of Fig. 2(b), and moreover, T_3 and T_5 remain unchanged whether or not C_0 is present, but the location of the zeros T_4, T_6 is different from that of both Figs. 2(b) and 3(a) as a result of the presence of C_0 in the noninverting configuration. This is in contradistinction to the inverting case, where only T_2 was affected by the presence or absence of C_0, and T_3–T_6 remained unchanged.

The analysis proceeds by equating the right-hand sides of Eqs. (7) and (8), obtaining a system of six equations which can be solved for T_1–T_6 in terms of R_0, R_1, R_2, R_3, C_0, C_1, C_2, and also for the resistor and capacitor values of the network components in terms of T_1–T_6. An example of the calculations involved in the case of the network of Fig. 1(a) is presented in Appendix 2, while the results are collected in Table 3(a)–(d) for the networks of Fig. 1(a)–(d), respectively.[8] The points made in the preceding paragraph are apparent from the formulas in the upper third of the table. In Table 4 we give the formulas for $G(\omega)$ for the circuit of Fig. 3(b). Reference should also be made to the design notes in Fig. 3(b). For this configuration only *one*

[8] Again, the symbol [·] used in some of the formulas denotes a repetition of the square-bracketed expression which precedes it within the same formula.



NOTES:

1) R_0 includes shunting effect of load resistance; R_3 includes source resistance. At high frequency the load on the source is $(R_0 + R_3)$.
2) If $R_3 \neq 0$: Both R_0 and R_3 can be chosen independently. To adjust gain without affecting frequency response, change R_3/R_0 while keeping $(R_0 + R_3)$ fixed. $(R_0 + R_3)$ alone determines ω_4, ω_6, while ω_3, ω_5 are not affected by changing R_0, R_3.
3) If $R_3 = 0$: Only R_0 can be chosen independently. Changing R_0 affects *both* gain and ω_4, ω_6, while ω_3, ω_5 are not affected by changing R_0.
4) Use Tables 1 and 2 with R_3 replaced by $(R_0 + R_3)$ wherever it occurs, and $G(\omega)$ replaced by $-1/G(\omega)$.

Fig. 5. Passive preemphasis circuit.

of the network components can be chosen independently, and then all the others are fixed by the values of T_1–T_6. In view of the similarities with Fig. 3(a) and the appearance of R_0 and R_3 in the combination $(R_0 + R_3)$ in Eq. (7), the most logical and convenient choice for independent variable is $(R_0 + R_3)$. The formulas in the middle third of Table 3 are therefore expressed in terms of $(R_0 + R_3)$ and T_1–T_6. Since T_1 is an artifact of this circuit configuration and not of primary importance to us from the RIAA point of view, an alternative and more useful way of considering these equations is by choosing *both* $(R_0 + R_3)$ and $(R_0 + R_3)/R_0$ independently, and looking upon T_1 as a *dependent* quantity, related to the others by the formula (from Table 3):

$$\frac{R_0 + R_3}{R_0} = \frac{T_1 T_4 T_6}{T_2 T_3 T_5} \geq 1 \ . \tag{9}$$

This formula is seen to provide a constraint on the allowable values of T_1–T_6, for we must always have the inequality satisfied. If $R_3 = 0$, it reduces to the constraint

$$T_1 T_4 T_6 = T_2 T_3 T_5 \tag{10}$$

and R_0 remains as the only independent variable in this case. We see that a small value for T_6, which is desirable for high-frequency RIAA equalization accuracy, then necessitates a large value for T_1, which results in a long dc stabilization time for the circuit—an undesirable artifact. The time constants thus must be played off one against the other in a practical circuit.

In the next section we shall discuss design procedures using all the formulas so far developed.

Table 1(a). Design formulas for active inverting deemphasis circuits of Fig. 2, using network of Fig. 1(a).

	$R_3 = 0$		$R_3 \neq 0$
Quantity	Formula	RIAA/IEC	Formula
T_2	R_0C_0	7950.00 μs	R_0C_0
T_3	R_1C_1	3180.00 μs	R_1C_1
T_4	$\frac{R_1R_2}{R_1+R_2}(C_1+C_2)$	318.00 μs	$\approx \frac{R_1R_2}{R_1+R_2}(C_1+C_2) + \frac{(R_1C_1-R_2C_2)^2}{(R_1+R_2)^2(C_1+C_2)}R_3$
T_5	R_2C_2	75.00 μs	R_2C_2
T_6	0		$\approx \frac{C_1C_2}{C_1+C_2}R_3$
R_1/R_3			$\frac{T_5(T_3-T_4)(T_3-T_6)}{T_4T_6(T_3-T_5)}$
R_2/R_3			$\frac{T_3(T_4-T_5)(T_5-T_6)}{T_4T_6(T_3-T_5)}$
R_0C_0	T_2	7950.00 μs	T_2
R_3C_1			$\frac{T_3T_4T_6(T_3-T_5)}{T_5(T_3-T_4)(T_3-T_6)}$
R_3C_2			$\frac{T_4T_5T_6(T_3-T_5)}{T_3(T_4-T_5)(T_5-T_6)}$
R_1C_1	T_3	3180.00 μs	T_3
R_2C_2	T_5	75.00 μs	T_5
R_1C_2	$\frac{T_5(T_3-T_4)}{T_4-T_5}$	883.33 μs	$\frac{T_5^2(T_3-T_4)(T_3-T_6)}{T_3(T_4-T_5)(T_5-T_6)}$
R_2C_1	$\frac{T_3(T_4-T_5)}{T_3-T_4}$	270.00 μs	$\frac{T_3^2(T_4-T_5)(T_5-T_6)}{T_5(T_3-T_4)(T_3-T_6)}$
R_1/R_2	$\frac{T_3-T_4}{T_4-T_5}$	11.778	$\frac{T_5(T_3-T_4)(T_3-T_6)}{T_3(T_4-T_5)(T_5-T_6)}$
C_1/C_2	$\frac{T_3(T_4-T_5)}{T_5(T_3-T_4)}$	3.600	$\frac{T_3^2(T_4-T_5)(T_5-T_6)}{T_5^2(T_3-T_4)(T_3-T_6)}$

5. HOW TO DESIGN RIAA CIRCUITS

By this stage the reader should already have a fairly good idea of the correct design procedure, making use of Tables 1–4 as appropriate. We can, however, usefully make a number of additional remarks. We shall assume that T_3–T_5 are given their RIAA values and that T_2, if present, is given either its IEC value of 7950 μs or else is suitably chosen to determine the circuit's low-frequency rolloff point. In any event it is assumed that the values of T_2–T_5 are known and fixed beforehand.

Clearly, once the circuit configuration (Figs. 2–5) has been selected, the next decision is between the four electrically equivalent networks of Fig. 1, and here a choice must be based on practical factors. Please bear in mind that we are still assuming adequate loop gain at all relevant frequencies to ensure adherence to the frequency response curve dictated by the feedback network. We shall show in Section 7 how to deal with cases in which this assumption is not valid. As is evident from the first column of Table 1, the R_1/R_2 and C_1/C_2 ratios are different for each of the four networks. Since, in practice, the range of available capacitor values is more restricted than that of resistor values, a reasonable first question to ask is which networks have a capacitor ratio that is available from, say, the standard E24 series of capacitors.[9] Now as the formulas in the second column of Table 1 show, the capacitor and resistor ratios change from their "ideal" values, given in the first column, as R_3 [or $(R_0 + R_3)$ in the case of Figs. 3 and 5] increases in value from zero. So our question must be in two parts:

1) In the ideal case $R_3 = 0$, which networks realize available E24 capacitor ratios?

2) In the case $R_3 \neq 0$ (or $R_0 + R_3 \neq 0$ for Figs. 3, 5), which networks realize available E24 capacitor ratios *and* give T_6 sufficiently small that their high-frequency zeros lie well above the audio band?

A bit of calculating using a table of E24 values and Table 1 leads to Table 5 and an answer to our questions:

1) In the ideal case only the networks of Fig. 1(a) and (d) are achievable using standard E24 capacitor values. The only three possible "ideal" designs calculated from the first column of Table 1 are given in Table 5(a), with closest E96 resistor values in parentheses.[9] Of course, if one is willing to parallel capacitors to form C_1 and C_2, an infinity of designs is possible.

[9] The E24 series of component values comprises 24 values spanning each decade, the ratio of each value to its predecessor being $10^{1/24}$. The E96 series, in the ratio $10^{1/96}$, contains 96 component values in each decade.

Table 1(b). Design formulas for active inverting deemphasis circuits of Fig. 2, using network of Fig. 1(b).

Quantity	$R_3 = 0$		$R_3 \neq 0$
	Formula	RIAA/IEC	Formula
T_2	R_0C_0	7950.00 μs	R_0C_0
T_3	$\frac{1}{2}\{[R_1C_1 + R_2(C_1 + C_2)] + \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$	3180.00 μs	$\frac{1}{2}\{[R_1C_1 + R_2(C_1 + C_2)] + \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
T_4	$R_2(C_1 + C_2)$	318.00 μs	$\approx R_2(C_1 + C_2) + \dfrac{C_1^2}{C_1 + C_2} R_3$
T_5	$\frac{1}{2}\{[R_1C_1 + R_2(C_1 + C_2)] - \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$	75.00 μs	$\frac{1}{2}\{[R_1C_1 + R_2(C_1 + C_2)] - \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
T_6	0		$\approx \dfrac{C_1C_2}{C_1 + C_2} R_3$
R_1/R_3			$\dfrac{T_3T_5}{T_4T_6} - 1$
R_2/R_3			$\dfrac{R_2}{R_1} \cdot \dfrac{R_1}{R_3}$
R_0C_0	T_2	7950.00 μs	T_2
R_3C_1			$R_1C_1 \cdot \dfrac{R_3}{R_1}$
R_3C_2			$R_1C_2 \cdot \dfrac{R_3}{R_1}$
R_1C_1	$T_3 - T_4 + T_5$	2937.00 μs	$\dfrac{T_3T_5}{R_2C_2}$
R_2C_2	$\dfrac{T_3T_5}{T_3 - T_4 + T_5}$	81.21 μs	$\dfrac{T_3T_5 - T_4T_6}{T_3 - T_4 + T_5 - T_6}$
R_1C_2	$\dfrac{T_3T_5(T_3 - T_4 + T_5)}{(T_3 - T_4)(T_4 - T_5)}$	1007.20 μs	$\dfrac{T_3T_5}{R_2C_1}$
R_2C_1	$\dfrac{(T_3 - T_4)(T_4 - T_5)}{T_3 - T_4 + T_5}$	236.79 μs	$T_3 + T_5 - R_1C_1 - R_2C_2$
R_1/R_2	$\dfrac{(T_3 - T_4 + T_5)^2}{(T_3 - T_4)(T_4 - T_5)}$	12.403	$\dfrac{R_1C_2}{R_2C_2}$
C_1/C_2	$\dfrac{(T_3 - T_4)(T_4 - T_5)}{T_3T_5}$	2.916	$\dfrac{R_2C_1}{R_2C_2}$

2) As R_3 (or $R_0 + R_3$) increases from zero, the C_1/C_2 ratio decreases from its value given in the first column of Table 1, and simultaneously T_6 increases in value from zero. In seeking the best designs possible in this case, which represents the most frequent situation, it is best to proceed backwards. Starting with the formula for C_1/C_2 from the second column of Table 1, we solve it for T_6 in terms of T_3–T_5 and C_1/C_2. Then from the E24 series we choose capacitor values yielding a C_1/C_2 ratio just less than that given in the first column, and calculate the corresponding value of T_6 from our formula. This value of T_6 is then used in the second column to calculate all other component values. In this way we construct the designs given in Table 5(b), listed in the order of decreasing f_6. These are believed to represent the best such designs possible. Again, many more are possible if we are willing to parallel capacitors. Note that all four networks N are represented in Table 5(b). This table can be used to construct very accurate and cheap designs, using few components, for the circuits of Figs. 2, 3(a), 4, and 5, and to a high degree of accuracy, also Fig. 3(b).

Overall it would appear that the network of Fig. 1(a) is perhaps not undeservedly the most popular of the four. An interesting question which springs to mind is whether any one of the networks offers an advantage over the others as regards the ease with which it can be "trimmed" for accuracy. To begin with, trimming is a difficult procedure, for each component affects at least two of the finally realized time constants of the network. Furthermore, to be able to trim accurately one must have either a precision RIAA circuit for reference or else be able to measure over a dynamic range of >40 dB and over a frequency range of >3 decades to an accuracy of tenths of a decibel. This is not an easy task. In fact, it is sufficiently difficult that the writer would suggest that a much better and easier procedure in practice is to produce an accurate design in the first place, and not rely on trimming to adjust the circuit for accuracy. This is, in fact, the whole thesis of this paper. This said, it is interesting to examine Table 6, a table of relative sensitivities of the main network RIAA

Table 1(c). Design formulas for active inverting deemphasis circuits of Fig. 2 using network of Fig. 1(c).

	$R_3 = 0$		$R_3 \neq 0$
Quantity	Formula	RIAA/IEC	Formula
T_2	R_0C_0	7950.00 μs	R_0C_0
T_3	$\frac{1}{2}\{[R_1(C_1 + C_2) + R_2C_1] + \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$	3180.00 μs	$\frac{1}{2}\{[R_1(C_1 + C_2) + R_2C_1] + \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
T_4	R_2C_1	318.00 μs	$\approx R_2C_1 + C_1R_3$
T_5	$\frac{1}{2}\{[R_1(C_1 + C_2) + R_2C_1] - \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$	75.00 μs	$\frac{1}{2}\{[R_1(C_1 + C_2) + R_2C_1] - \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
T_6	0		$\approx C_2R_3$
R_1/R_3			$\frac{T_3T_5}{T_4T_6} - 1$
R_2/R_3			$\frac{R_2}{R_1} \cdot \frac{R_1}{R_3}$
R_0C_0	T_2	7950.00 μs	T_2
R_3C_1			$R_2C_1 \cdot \frac{R_3}{R_2}$
R_3C_2			$R_1C_2 \cdot \frac{R_3}{R_1}$
R_1C_1	$\frac{(T_3 - T_4)(T_4 - T_5)}{T_4}$	2187.00 μs	$T_3 + T_5 - R_1C_2 - R_2C_1$
R_2C_2	$\frac{T_3T_4T_5}{(T_3 - T_4)(T_4 - T_5)}$	109.05 μs	$\frac{T_3T_5}{R_1C_1}$
R_1C_2	$\frac{T_3T_5}{T_4}$	750.00 μs	$\frac{T_3T_5}{R_2C_1}$
R_2C_1	T_4	318.00 μs	$\frac{T_3T_4(T_5 - T_6) + T_5T_6(T_3 - T_4)}{T_3T_5 - T_4T_6}$
R_1/R_2	$\frac{(T_3 - T_4)(T_4 - T_5)}{T_4^2}$	6.877	$\frac{R_1C_1}{R_2C_1}$
C_1/C_2	$\frac{(T_3 - T_4)(T_4 - T_5)}{T_3T_5}$	2.916	$\frac{R_2C_1}{R_2C_2}$

time constants (T_3–T_5) to changes in the values of the components R_1, R_2, C_1, and C_2. They are calculated from the formula

$$S_x^{T_i} = \frac{x}{T_i} \cdot \frac{\partial T_i}{\partial x}, \qquad i = 3,4,5$$

where x is one of R_1, R_2, C_1, or C_2, and represent the percentage change in T_i caused by a 1% change in the component x from its ideal value given in the first column of Table 1. Table 6 must be interpreted with care, but it does show that the network of Fig. 1(c) is the best, and that of Fig. 1(b) the worst, from the interaction (and hence also from the trimming) point of view. A suitable trimming procedure for the Fig. 1(a) network would be to fix R_1, say, and first adjust C_1 at 100 Hz to trim T_3; then adjust R_2 at 1 kHz to trim T_4; and finally adjust C_2 at 10 kHz to trim T_5. Of course, the procedure must be iterated, and is made more complicated by the effect each component change has on the overall gain, as is evident from Tables 2 and 4.

The next point to make is that, for all deemphasis circuits with $T_6 \neq 0$ (that is, Fig. 2 with $R_3 \neq 0$ and all cases of Fig. 3), the high-frequency zero thus introduced can be *exactly* canceled by adding an identical high-frequency pole at the output of the circuit. A passive R/C low-pass filter of time constant T_6 will do this, and if T_6 is small enough, will not significantly degrade output impedance. For example, the second and third designs given in Table 5(b), with $T_6 = 0.4$ μs, can be corrected with a filter having $R = 1.1$ kΩ and $C = 360$ pF. Such a filter *should* be incorporated, especially in those designs where f_6 is rather close to the audio band. Failure to do so will then lead to a rising response (relative to RIAA) in the top octave of the audio band.

This brings us to the next point. In a practical circuit T_6 usually cannot be made arbitrarily small, for decreasing T_6 is equivalent to decreasing R_3 for the circuits of Figs. 2 and 4 or ($R_0 + R_3$) for the circuits of Figs. 3 and 5. Practical questions of amplifier loading and stabilization will generally prevent us from decreasing these components too far, although noise considerations *per se* would dictate using the smallest possible values. In particular, R_3 may be *required* in order to ensure amplifier closed-loop stability without excessive reduction in the gain–bandwidth product and slewing rate. Also, T_1 should be made as small as possible in Fig. 3(b), for it determines the length of time the circuit will take to stabilize its dc operating levels. However, T_1 and T_6 are interrelated according to Eqs. (9),

Table 1(d). Design formulas for active inverting deemphasis circuits of Fig. 2, using network of Fig. 1(d).

Quantity	$R_3 = 0$ Formula	RIAA/IEC	$R_3 \neq 0$ Formula
T_2	R_0C_0	7950.00 μs	R_0C_0
T_3	$\tfrac{1}{2}\{[R_1(C_1 + C_2) + R_2C_2] + \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$	3180.00 μs	$\tfrac{1}{2}\{[R_1(C_1 + C_2) + R_2C_2] + \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
T_4	$\frac{R_1R_2}{R_1 + R_2} C_1$	318.00 μs	$\approx \frac{R_1R_2}{R_1 + R_2} C_1 + \frac{R_1{}^2C_1}{(R_1 + R_2)^2} R_3$
T_5	$\tfrac{1}{2}\{[R_1(C_1 + C_2) + R_2C_2] - \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$	75.00 μs	$\tfrac{1}{2}\{[R_1(C_1 + C_2) + R_2C_2] - \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
T_6	0		$\approx C_2R_3$
R_1/R_3			$\frac{R_1}{R_2} \cdot \frac{R_2}{R_3}$
R_2/R_3			$\frac{R_2C_2}{R_3C_2}$
R_0C_0	T_2	7950.00 μs	T_2
R_3C_1			$R_2C_1 \cdot \frac{R_3}{R_2}$
R_3C_2			$\frac{T_3T_4T_5T_6}{T_3T_4(T_5 - T_6) + T_5T_6(T_3 - T_4)}$
R_1C_1	$\frac{(T_3 - T_4)(T_4 - T_5) + T_4{}^2}{T_4}$	2505.00 μs	$T_3 + T_5 - \frac{(T_3T_5 - T_4T_6)}{T_4T_6} R_3C_2$
R_2C_2	$\frac{T_3T_4T_5}{(T_3 - T_4)(T_4 - T_5) + T_4{}^2}$	95.21 μs	$\frac{T_3T_5}{R_1C_1}$
R_1C_2	$\frac{T_3T_5(T_3 - T_4)(T_4 - T_5)}{T_4[(T_3 - T_4)(T_4 - T_5) + T_4{}^2]}$	654.79 μs	$T_3 + T_5 - R_1C_1 - R_2C_2$
R_2C_1	$\frac{T_4[(T_3 - T_4)(T_4 - T_5) - T_4{}^2]}{(T_3 - T_4)(T_4 - T_5)}$	364.24 μs	$\frac{T_3T_5}{R_1C_2}$
R_1/R_2	$\frac{(T_3 - T_4)(T_4 - T_5)}{T_4{}^2}$	6.877	$\frac{R_1C_1}{R_2C_1}$
C_1/C_2	$\frac{[(T_3 - T_4)(T_4 - T_5) + T_4{}^2]^2}{T_3T_5(T_3 - T_4)(T_4 - T_5)}$	3.826	$\frac{R_1C_1}{R_1C_2}$

(10), and the gain formula, and so decreasing T_1 results in an increase in T_6 and a change in gain. As an example, if T_2 is chosen to be 7950 μs for an IEC design, we find from Eq. (9) that

$$T_1T_6 \geq 5.96 \times 10^{-6}\ [\mathrm{s}^2]$$

and so for $T_6 = 0.4\ \mu$s we would have $T_1 \geq 14.9$ s. For a non-IEC design using Fig. 3(b), T_2 would be larger and so T_1 would be even greater for the same gain. In general, although T_2–T_5 are specified by the RIAA/IEC, T_1 and T_6 and the gain are at our disposal. Since the error caused by T_6 can be exactly compensated for, in a practical circuit we may have to increase T_6 in order to obtain an acceptably small value for T_1 and a suitable gain. Reference to the frequency response curve of Fig. 3(b) shows that T_1 is affected by both the circuit's gain and the location of T_2. If T_2 is specified beforehand, changing T_1 necessitates a change in gain.

A final important practical consideration is the circuit's 1-kHz gain. This can be calculated using Tables 2 or 4 as appropriate. In fact, it may be useful in the course of design to work backwards from these tables, starting with a given desired 1-kHz gain together with Eqs. (9) and (10) and calculating the corresponding values of T_1 and/or T_6 to realize this gain before proceeding to use these values in Tables 1 and 3. Speaking about gain, the design notes in Figs. 2–5 give important information concerning gain adjustment in these circuits. Referring to the upper third of Tables 1 and 3 it is seen that, when changing gain in the circuits of Figs. 2 and 4, R_3 should be held fixed and only R_0 varied, while for the circuits of Figs. 3 and 5, $(R_0 + R_3)$ should be held fixed as R_3/R_0 is varied (that is, the tapping point along $R_0 + R_3$ is varied). This procedure will ensure that the *only* frequency response casualty will be T_2. Any other procedure will affect the important RIAA time constant T_4. This point is of considerable significance, and it appears to be generally ignored in practice.

The only major design problem which can yet affect our considerations above is the lack of suitable loop gain to guarantee adherence to these formulas. We address this problem in Section 7, but first an example.

6. AN EXAMPLE

For the purposes of illustration let us consider the most

Table 2. Gain formulas for active inverting deemphasis circuits of Fig. 2.

Quantity	$R_3 = 0$	
	Without C_0	With C_0
$G(0)$	$-\dfrac{R_A}{R_0}$	0
$G(\infty)$	0	0
$G(\omega)$	$-\dfrac{R_A}{R_0}\sqrt{\dfrac{1+T_4^2\omega^2}{(1+T_3^2\omega^2)(1+T_5^2\omega^2)}}$	$-\dfrac{R_A}{R_0}\sqrt{\dfrac{T_2^2\omega^2(1+T_4^2\omega^2)}{(1+T_2^2\omega^2)(1+T_3^2\omega^2)(1+T_5^2\omega^2)}}$
Quantity	$R_3 \neq 0$	
	Without C_0	With C_0
$G(0)$	$-\dfrac{R_A+R_3}{R_0} \equiv -\dfrac{R_3}{R_0}\cdot\dfrac{T_3T_5}{T_4T_6}$	0
$G(\infty)$	$-\dfrac{R_3}{R_0}$	$-\dfrac{R_3}{R_0}$
$G(\omega)$	$-\dfrac{R_A+R_3}{R_0}\sqrt{\dfrac{(1+T_4^2\omega^2)(1+T_6^2\omega^2)}{(1+T_3^2\omega^2)(1+T_5^2\omega^2)}}$	$-\dfrac{R_A+R_3}{R_0}\sqrt{\dfrac{T_2^2\omega^2(1+T_4^2\omega^2)(1+T_6^2\omega^2)}{(1+T_2^2\omega^2)(1+T_3^2\omega^2)(1+T_5^2\omega^2)}}$
	$\equiv -\dfrac{R_3}{R_0}\cdot\dfrac{T_3T_5}{T_4T_6}\sqrt{\dfrac{(1+T_4^2\omega^2)(1+T_6^2\omega^2)}{(1+T_3^2\omega^2)(1+T_5^2\omega^2)}}$	$\equiv -\dfrac{R_3}{R_0}\cdot\dfrac{T_3T_5}{T_4T_6}\sqrt{\dfrac{T_2^2\omega^2(1+T_4^2\omega^2)(1+T_6^2\omega^2)}{(1+T_2^2\omega^2)(1+T_3^2\omega^2)(1+T_5^2\omega^2)}}$

difficult design case, namely, the circuit of Fig. 3(b), which also represents a sizable proportion of higher priced commercial circuits. Let us set as our criteria a 1-kHz gain of around 35 dB and a frequency response as dictated by RIAA/IEC, that is,

$$T_2 = 7950\ \mu\text{s}, \qquad T_3 = 3180\ \mu\text{s}, \qquad T_4 = 318\ \mu\text{s}, \qquad T_5 = 75\ \mu\text{s}.$$

Reference to a straight-line approximation to the RIAA/IEC frequency response curve defined in [34] shows us that its idealized gain at 20 Hz (corresponding to T_2) is +19.9 dB relative to that at 1 kHz. Hence the desired idealized signal gain at 20 Hz is 54.9 dB, and with reference to Fig. 3(b) we conclude that $f_1 = 0.0360$ Hz, giving $T_1 = 4.419$ s. Then Eq. (9) shows that $T_6 \geq 1.349\ \mu$s, with equality if and only if $R_3 = 0$. The case $R_3 = 0$ corresponds to a high-frequency zero at 118.0 kHz, which is reasonably placed two octaves above the audio band. Let us choose the network of Fig. 1(a) and set $R_0 = 1$ kΩ. Then the following component values are easily calculated from the middle third of Table 3(a) for each of two possible designs (assuming sufficient loop gain):

1) $R_3 = 0$; $f_6 = 118.0$ kHz, $R_0 = 1$ kΩ [Fig. 1(a)], and

$$\begin{aligned} R_1 &= 511.813\ \text{k}\Omega, & R_2 &= 42.722\ \text{k}\Omega \\ C_1 &= 6.213\ \text{nF}, & C_2 &= 1.756\ \text{nF}. \\ & & C_0 &= 7.950\ \mu\text{F} \end{aligned}$$

with high-frequency zero correction filter of 1 kΩ and 1.349 nF. The 1-kHz gain follows from Table 4 as 35.0 dB, as desired.

2) $R_3 = 1$ kΩ, say, to help stabilize the amplifier by increasing the high-frequency noise gain to 2;[10] $f_6 = 59.0$ kHz, $R_0 = 1$ kΩ [Fig. 1(a)], and

$$\begin{aligned} R_1 &= 511.596\ \text{k}\Omega, & R_2 &= 41.940\ \text{k}\Omega \\ C_1 &= 6.216\ \text{nF}, & C_2 &= 1.788\ \text{nF}, \\ & & C_0 &= 7.950\ \mu\text{F} \end{aligned}$$

with high-frequency correction filter of 2 kΩ and 1.349 nF. Again, Table 4 confirms the 1-kHz gain as 35.0 dB.

To give some idea of the accuracy achievable through the use of the formulas, we give in Table 7[11] the measured RIAA frequency response error of a circuit of the Fig. 2(a) type, using the theoretically calculated component values.

It is of interest to note the small changes in the values of R_1, R_2, C_1, and C_2 between these two designs. If it is desired to experiment in order to bring some of the component values closer to standard available values, one can try changing $(R_0 + R_3)$, T_6, the 1-kHz gain, and/or the network to Fig. 1(b)–(d). In this way the design can be optimized.

7. TAKING INADEQUATE LOOP GAIN INTO ACCOUNT

In this last section we shall consider what can be done if the amplifier in one of our circuits does not have enough loop gain at some frequencies to ensure adequate (say,

[10] The noise gain of an amplifier, being the gain experienced by input-referred noise and error, is the reciprocal of the feedback loop attenuation β. It is equal to the signal gain for an ideal noninverting configuration, and is important, for together with the open-loop gain, it determines a feedback amplifier's stability.

[11] Table 7 is extracted from [26, Table IV] and reproduced with the kind permission of the publisher (The Audio Amateur, P.O. Box 176, Peterborough, NH 03458).

Table 3(c). Design formulas for active noninverting deemphasis circuit of Fig. 3(b), using network of Fig. 1(a).

Quantity	Formula
T_1, T_4, T_6	With $T_1 > T_4 > T_6$, they are $-\left(\frac{1}{\text{roots}}\right)$ of the cubic in s:
	$\begin{aligned} 0 = 1 &+ [(R_0 + R_3)C_0 + (R_1 + R_2)C_0 + R_1C_1 + R_2C_2]s \\ &+ [(R_0 + R_3)C_0(R_1C_1 + R_2C_2) + R_1R_2\{C_0(C_1 + C_2) + C_1C_2\}]s^2 \\ &+ [(R_0 + R_3)C_0R_1C_1R_2C_2]s^3 \end{aligned}$
T_2	R_0C_0
T_3	R_1C_1
T_5	R_2C_2
$R_0/(R_0 + R_3)$	$\frac{T_2T_3T_5}{T_1T_4T_6} \rightarrow$ constraint $T_1T_4T_6 = T_2T_3T_5$ if $R_3 = 0$
$R_1/(R_0 + R_3)$	$\frac{T_5(T_1 - T_3)(T_3 - T_4)(T_3 - T_6)}{T_1T_4T_6(T_3 - T_5)}$
$R_2/(R_0 + R_3)$	$\frac{T_3(T_1 - T_5)(T_4 - T_5)(T_5 - T_6)}{T_1T_4T_6(T_3 - T_5)}$
R_0C_0	T_2
$(R_0 + R_3)C_1$	$\frac{T_1T_3T_4T_6(T_3 - T_5)}{T_5(T_1 - T_3)(T_3 - T_4)(T_3 - T_6)}$
$(R_0 + R_3)C_2$	$\frac{T_1T_4T_5T_6(T_3 - T_5)}{T_3(T_1 - T_5)(T_4 - T_5)(T_5 - T_6)}$
R_1C_1	T_3
R_2C_2	T_5
R_1C_2	$\frac{T_5^2(T_1 - T_3)(T_3 - T_4)(T_3 - T_6)}{T_3(T_1 - T_5)(T_4 - T_5)(T_5 - T_6)}$
R_2C_1	$\frac{T_3^2(T_1 - T_5)(T_4 - T_5)(T_5 - T_6)}{T_5(T_1 - T_3)(T_3 - T_4)(T_3 - T_6)}$
R_1/R_2	$\frac{T_5(T_1 - T_3)(T_3 - T_4)(T_3 - T_6)}{T_3(T_1 - T_5)(T_4 - T_5)(T_5 - T_6)}$
C_1/C_2	$\frac{T_3^2(T_1 - T_5)(T_4 - T_5)(T_5 - T_6)}{T_5^2(T_1 - T_3)(T_3 - T_4)(T_3 - T_6)}$

0.2-dB) adherence of the signal gain to that dictated by the feedback network.[12] This will be the case if the loop gain is less than 30–40 dB at any frequency at which the open-loop and noise gain curves are parallel (no relative phase shift), or less than 15–20 dB at any freqeuncy at which the open-loop and noise gain curves have a relative slope of 6 dB per octave (90° relative phase shift).[13] This occurs frequently in practice in disk preamplifiers, and two particular situations are common:

1) The discrete amplifier with large open-loop bandwidth but inadequate low-frequency open-loop gain. Here the main RIAA errors are in the region of the ω_2 and ω_3 poles.

2) The integrated operational amplifier with large low-frequency open-loop gain but small open-loop bandwidth, resulting in inadequate high-frequency loop gain. Here the main errors are around the pole at ω_5.

These errors take the form of deviations in both gain and pole position. These two situations are illustrated diagrammatically in Fig. 6 for the circuit of Fig. 3(b). The solid lines represent the originally intended RIAA response, and the dashed lines show the response actually realized. One solution is, of course, to reduce the desired 1-kHz signal gain in order to increase the available loop gain. If this is not practicable, other solutions must be sought, and these are the topic of the present discussion. Since our aim is mainly to illustrate a design procedure by means of which these errors can be avoided, we shall restrict the discussion to the circuit of Fig. 3(b). It can, however, be applied to the other circuits, but with somewhat greater difficulty.

In the diagrams of Fig. 6 the unprimed quantities are the ones used in the formulas developed earlier for the case of infinite open-loop gain, realizing closed-loop gain G, while the primed quantities are those actually realized due to the finite open-loop gain. Our aim is to force the primed ω_i to take on the *desired* RIAA values by *deliberately* choosing the unprimed ω_i differently. Then the shape of the achieved (dashed) curves will be correct, although their gain will be somewhat below that predicted by Table 4. This is indeed possible.

[12] The writer would like to thank John Vanderkooy for bringing home to him the importance of a discussion of this topic, and for suggesting a possible analytical approach.

[13] See footnote 10.

Table 3(b). Design formulas for active noninverting deemphasis circuit of Fig. 3(b), using network of Fig. 1(b).

Quantity	Formula
T_1, T_4, T_6	With $T_1 > T_4 > T_6$, they are $-\ (\frac{1}{\text{roots}})$ of the cubic in s:
	$0 = 1 + [(R_0 + R_3)C_0 + R_1C_0 + R_1C_1 + R_2(C_1 + C_2)]s$ $+ [(R_0 + R_3)C_0\{R_1C_1 + R_2(C_1 + C_2)\} + R_1R_2\{C_0(C_1 + C_2) + C_1C_2\}]s^2$ $+ [(R_0 + R_3)C_0R_1C_1R_2C_2]s^3$
T_2	R_0C_0
T_3	$\frac{1}{2}\{[R_1C_1 + R_2(C_1 + C_2)] + \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
T_5	$\frac{1}{2}\{[R_1C_1 + R_2(C_1 + C_2)] - \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
$R_0/(R_0 + R_3)$	$\frac{T_2T_3T_5}{T_1T_4T_6} \rightarrow$ constraint $T_1T_4T_6 = T_2T_3T_5$ if $R_3 = 0$
$R_1/(R_0 + R_3)$	$\frac{T_3T_5(T_1 - T_3 + T_4 - T_5 + T_6)}{T_1T_4T_6} - 1$
$R_2/(R_0 + R_3)$	$\frac{R_2}{R_1} \cdot \frac{R_1}{R_0 + R_3}$
R_0C_0	T_2
$(R_0 + R_3)C_1$	$R_1C_1 \cdot \frac{R_0 + R_3}{R_1}$
$(R_0 + R_3)C_2$	$R_1C_2 \cdot \frac{R_0 + R_3}{R_1}$
R_1C_1	$\frac{T_3T_5}{R_2C_2}$
R_2C_2	$\frac{T_3T_5(T_1 - T_3 + T_4 - T_5 + T_6) - T_1T_4T_6}{(T_3 + T_5)(T_1 - T_3 + T_4 - T_5 + T_6) - (T_1T_4 + T_1T_6 + T_4T_6 - T_3T_5)}$
R_1C_2	$\frac{T_3T_5}{R_2C_1}$
R_2C_1	$T_3 + T_5 - R_1C_1 - R_2C_2$
R_1/R_2	$\frac{R_1C_2}{R_2C_2}$
C_1/C_2	$\frac{R_2C_1}{R_2C_2}$

Let

$$G(s) = \frac{N(s)}{D(s)} \quad \text{and} \quad G'(s) = k\,\frac{N'(s)}{D'(s)} \tag{11}$$

where k is a constant, and $N(s)$, $N'(s)$ and $D(s)$, $D'(s)$ are the polynomials in s in the numerators and denominators of the gain formulas $G(s)$ and $G'(s)$, respectively, from Eq. (8). If $A_v(s)$ denotes the open-loop gain, the familiar gain formula

$$G'(s) = \frac{A_v(s)}{1 + A_v(s)/G(s)} = \frac{G(s)}{1 + G(s)/A_v(s)}$$

for a noninverting amplifier yields

$$k\,\frac{N'(s)}{D'(s)} = \frac{N(s)}{D(s) + N(s)/A_v(s)} \tag{12}$$

as the relation between the N and the D. We now specialize Eq. (12) to the two cases of Fig. 6.

7.1 Constant Open-Loop Gain: $A_v = A_{v0}$

This is a good approximation to a wide open-loop bandwidth amplifier. Then we deduce from Eq. (12) that

$$\left.\begin{aligned} k &= \frac{1}{1 + 1/A_{v0}}, \qquad N(s) = N'(s), \\ D(s) &= D'(s) - \frac{N'(s) - D'(s)}{A_{v0}}. \end{aligned}\right\} \tag{13}$$

The first important point to note is the formula for k, relating the 0-Hz gains $G(0)$ and $G'(0)$. Clearly, as $A_{v0} \rightarrow \infty$, these become equal as expected. The second point is the somewhat surprising fact that the zeros ω_1', ω_4', and ω_6' are *not shifted* in frequency by the finite loop gain error. It is only the poles ω_2', ω_3', and ω_5' which are shifted according to the last of Eqs. (13). Again, as $A_{v0} \rightarrow \infty$, they tend to their expected values. Our next step is to substitute the forms of N and D from Eq. (8) into the last of Eqs. (13) and equate coefficients of like powers of s on both sides. If we introduce the notation

$$\left.\begin{aligned} N_1' &= T_1' + T_4' + T_6', \\ N_2' &= T_1'T_4' + T_1'T_6' + T_4'T_6', \\ N_3' &= T_1'T_4'T_6' \\[1ex] D_1' &= T_2' + T_3' + T_5', \\ D_2' &= T_2'T_3' + T_2'T_5' + T_3'T_5', \\ D_3' &= T_2'T_3'T_5' \end{aligned}\right\} \tag{14}$$

Table 3(c). Design formulas for active noninverting deemphasis circuit of Fig. 3(b), using network of Fig. 1(c).

Quantity	Formula
T_1, T_4, T_6	With $T_1 > T_4 > T_6$, they are $-\left(\frac{1}{\text{roots}}\right)$ of the cubic in s: $0 = 1 + [(R_0+R_3)C_0 + R_1C_0 + R_1(C_1+C_2) + R_2C_1]s + [(R_0+R_3)C_0\{R_1(C_1+C_2)+R_2C_1\} + R_1C_1R_2(C_0+C_2)]s^2 + [(R_0+R_3)C_0R_1C_1R_2C_2]s^3$
T_2	R_0C_0
T_3	$\tfrac{1}{2}\{[R_1(C_1+C_2)+R_2C_1] + \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
T_5	$\tfrac{1}{2}\{[R_1(C_1+C_2)+R_2C_1] - \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
$R_0/(R_0+R_3)$	$\dfrac{T_2T_3T_5}{T_1T_4T_6} \rightarrow$ constraint $T_1T_4T_6 = T_2T_3T_5$ if $R_3 = 0$
$R_1/(R_0+R_3)$	$\dfrac{T_3T_5(T_1 - T_3 + T_4 - T_5 + T_6)}{T_1T_4T_6} - 1$
$R_2/(R_0+R_3)$	$\dfrac{R_2}{R_1} \cdot \dfrac{R_1}{R_0+R_3}$
R_0C_0	T_2
$(R_0+R_3)C_1$	$R_1C_1 \cdot \dfrac{R_0+R_3}{R_1}$
$(R_0+R_3)C_2$	$R_1C_2 \cdot \dfrac{R_0+R_3}{R_1}$
R_1C_1	$T_3 + T_5 - R_1C_2 - R_2C_1$
R_2C_2	$\dfrac{T_3T_5}{R_1C_1}$
R_1C_2	$\dfrac{T_3T_5}{R_2C_1}$
R_2C_1	$\dfrac{T_3T_5(T_1T_4 + T_1T_6 + T_4T_6 - T_3T_5) - (T_3+T_5)T_1T_4T_6}{T_3T_5(T_1 - T_3 + T_4 - T_5 + T_6) - T_1T_4T_6}$
R_1/R_2	$\dfrac{R_1C_1}{R_2C_1}$
C_1/C_2	$\dfrac{R_2C_1}{R_2C_2}$

we deduce that

$$\left.\begin{aligned} T_2 + T_3 + T_5 &= D_1' - \frac{N_1' - D_1'}{A_{v0}} \\ T_2T_3 + T_2T_5 + T_3T_5 &= D_2' - \frac{N_2' - D_2'}{A_{v0}} \\ T_2T_3T_5 &= D_3' - \frac{N_3' - D_3'}{A_{v0}} \end{aligned}\right\} \quad (15)$$

and so T_2, T_3, T_5 with $T_2 > T_3 > T_5$, are the roots of the cubic in T:

$$T^3 - \left[D_1' - \frac{N_1' - D_1'}{A_{v0}}\right]T^2 + \left[D_2' - \frac{N_2' - D_2'}{A_{v0}}\right]T - \left[D_3' - \frac{N_3' - D_3'}{A_{v0}}\right] = 0. \quad (16)$$

This equation is exact, and when we insert into its coefficients the *desired* (that is, primed) RIAA time constants, its solutions T_2, T_3, T_5 give us the time constants which, together with

$$T_1 = T_1', \qquad T_4 = T_4', \qquad T_6 = T_6' \quad (17)$$

are the ones that must be used in the design tables and formulas of earlier sections for the circuit to realize the desired frequency response with gain error k given by Eq. (13).

A reasonable approximation in this case is $T_5 = T_5'$ in view of the 20-dB greater loop gain available at ω_5. If this simplification is made in system (15), it follows that, as a good approximation, T_2 and T_3, with $T_2 > T_3$, can be obtained more simply as the roots of the quadratic in T:

$$T^2 - \left[T_2' + T_3' - \frac{N_1' - D_1'}{A_{v0}}\right]T + \left[D_3' - \frac{N_3' - D_3'}{A_{v0}}\right] \cdot \frac{1}{T_5'} = 0, \qquad T_5 = T_5'. \quad (18)$$

7.2 Integrating Open-Loop Gain: $A_v = \omega_0/s$

Here ω_0 denotes the unity-gain angular frequency of the amplifier. This is a good middle- to high-frequency approximation of an integrating operational amplifier (the common type) with small open-loop bandwidth. Then the denominator on the right-hand side of Eq. (12) is quartic in

Table 3(d). Design formulas for active noninverting deemphasis circuit of Fig. 3(b), using network of Fig. 1(d).

Quantity	Formula
T_1, T_4, T_6	With $T_1 > T_4 > T_6$, they are $-\ (\frac{1}{\text{roots}})$ of the cubic in s: $0 = 1 + [(R_0 + R_3)C_0 + (R_1 + R_2)C_0 + R_1(C_1 + C_2) + R_2C_2]s + [(R_0 + R_3)C_0\{R_1(C_1 + C_2) + R._2C_2\} + R_1C_1R_2(C_0 + C_2)]s^2 + [(R_0 + R_3)C_0R_1C_1R_2C_2]s^3$
T_2	R_0C_0
T_3	$\frac{1}{2}\{[R_1(C_1 + C_2) + R_2C_2] + \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
T_5	$\frac{1}{2}\{[R_1(C_1 + C_2) + R_2C_2] - \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2}\}$
$R_0/(R_0 + R_3)$	$\frac{T_2T_3T_5}{T_1T_4T_6} \rightarrow$ constraint $T_1T_4T_6 = T_2T_3T_5$ if $R_3 = 0$
$R_1/(R_0 + R_3)$	$\frac{R_1}{R_2} \cdot \frac{R_2}{R_0 + R_3}$
$R_2/(R_0 + R_3)$	$\frac{R_2C_2}{(R_0 + R_3)C_2}$
R_0C_0	T_2
$(R_0 + R_3)C_1$	$R_2C_1 \cdot \frac{R_0 + R_3}{R_2}$
$(R_0 + R_3)C_2$	$\frac{T_1T_3T_4T_5T_6}{T_3T_5(T_1T_4 + T_1T_6 + T_4T_6 - T_3T_5) - (T_3 + T_5)T_1T_4T_6}$
R_1C_1	$T_3 + T_5 - \frac{T_3T_5\,(T_1 - T_3 + T_4 - T_5 + T_6) - T_1T_4T_6}{T_1T_4T_6} \cdot (R_0 + R_3)C_2$
R_2C_2	$\frac{T_3T_5}{R_1C_1}$
R_1C_2	$T_3 + T_5 - R_1C_1 - R_2C_2$
R_2C_1	$\frac{T_3T_5}{R_1C_2}$
R_1/R_2	$\frac{R_1C_1}{R_2C_1}$
C_1/C_2	$\frac{R_1C_1}{R_1C_2}$

s, and hence the left-hand side must also have a fourth pole at ω_7', say, as illustrated in Fig. 6(b). Thus Eq. (12) becomes

$$k\ \frac{N'(s)}{(1 + T_7's)\,D'(s)} = \frac{N(s)}{D(s) + sN(s)/\omega_0}$$

and we deduce that

$$\left.\begin{aligned} k &= 1, \qquad N(s) = N'(s), \\ D(s) &= (1 + T_7's)D'(s) - \frac{sN'(s)}{\omega_0}. \end{aligned}\right\} \tag{19}$$

As in case 7.1, $G(s)$ and $G'(s)$ have the *same* zeros as expressed by Eq. (17). Now, however, $G(0)$ equals $G'(0)$, while the poles ω_2', ω_3', and ω_5' are shifted according to the last of Eqs. (19), and a further pole ω_7' is added. As $\omega_0 \rightarrow \infty$, $\omega_7' \rightarrow \infty$, and the other poles tend to their expected values.

Substituting into the last of Eqs. (19) from Eq. (8), and equating coefficients of like powers of s, we find, in the notation of Eq. (14), that T_2, T_3, and T_5, with $T_2 > T_3 > T_5$, are the roots of the cubic in T:

$$\begin{aligned} T^3 &- \left[D_1' + T_7' - \frac{1}{\omega_0}\right]T^2 \\ &+ \left[D_2' + T_7'D_1' - \frac{N_1'}{\omega_0}\right]T \\ &- \left[D_3' + T_7'D_2' - \frac{N_2'}{\omega_0}\right] = 0 \end{aligned} \tag{20}$$

where

$$T_7' = \frac{N_3'}{D_3'\omega_0} = \frac{T_1'T_4'T_6'}{T_2'T_3'T_5'\omega_0}\ . \tag{21}$$

Note that by the first formula in the middle third of Table 3 $\omega_7' \leq \omega_0$, with equality if and only if $R_3 = 0$. This is also evident from Fig. 6(b).

Reduction of Eq. (20) to a quadratic equation, in the way in which Eq. (18) was derived from Eq. (16), is *not* justifiable in this case, and the full cubic Eq. (20) should be used.

7.3 General Single-Pole Amplifier Gain: $A_v = \frac{A_{v0}}{1 + A_{v0}\,s/\omega_0}$

As a generalization, we can combine the cases of Fig.

Table 4. Gain formulas for active noninverting deemphasis circuit of Fig. 3(b).

Quantity	Formula
$G(0)$	1
$G(\infty)$	$\frac{T_1T_4T_6}{T_2T_3T_5}$, which becomes 1 if $R_3 = 0$
$G(\omega)$	$\sqrt{\frac{(1+T_1{}^2\omega^2)(1+T_4{}^2\omega^2)(1+T_6{}^2\omega^2)}{(1+T_2{}^2\omega^2)(1+T_3{}^2\omega^2)(1+T_5{}^2\omega^2)}}$

Whichever case we are dealing with, once the modified values T_1–T_6 have been calculated, the appropriate resistor and capacitor values can be obtained from Table 3.[14] One final comment is warranted. In practice it would appear that a procedure frequently adopted, when it transpires that a design is not following the required RIAA

[14] Of course, we assume that the shifts involved are not so large that the roots of Eqs. (16), (18), (20), and (22) become complex, for then the configurations under consideration *cannot* be made to follow the RIAA curve, and the amplifier's open-loop gain must be considered to be totally inadequate.

Table 5. Best possible RIAA network designs using E 24 series capacitors (closest E96 series resistors given in parentheses).

(a) Ideal case: $T_6 = 0$—Circuits of Figs. 2 and 4 with $R_3 = 0$.

Network of Fig. 1	C_1	C_2	$\frac{C_1}{C_2}$	R_1	R_2
(a)	2.7 nF	750 pF	3.600	1.178 MΩ (1.18 MΩ)	100.000 kΩ (100 kΩ)
(a)	3.6 nF	1.0 nF	3.600	883.333 kΩ (887 kΩ)	75.000 kΩ (75.0 kΩ)
(d)	1.8 nF	470 pF	3.830	1.392 MΩ (1.40 MΩ)	202.574 kΩ (205 kΩ)

(b) General case: $T_6 \neq 0$—Circuits of Figs. 2 and 4 with $R_3 \neq 0$, or of Figs. 3(a) and 5 with R_3 replaced by $(R_0 + R_3)$ below.

Network of Fig. 1	C_1	C_2	$\frac{C_1}{C_2}$	f_6	R_1	R_2	R_3 or $(R_0 + R_3)$
(a)	4.3 nF	1.2 nF	3.583	448 kHz	739.535 kΩ (732 kΩ)	62.500 kΩ (61.9 kΩ)	380.4 Ω (383 Ω)
(b)	1.8 nF	620 pF	2.903	398 kHz	1.632 MΩ (1.62 MΩ)	130.924 kΩ (130 kΩ)	871.0 Ω (866 Ω)
(c)	1.8 nF	620 pF	2.903	398 kHz	1.214 MΩ (1.21 MΩ)	176.018 kΩ (174 kΩ)	647.9 Ω (649 Ω)
(a)	2.0 nF	560 pF	3.571	261 kHz	1.590 MΩ (1.58 MΩ)	133.929 kΩ (133 kΩ)	1.402 kΩ (1.40 kΩ)
(d)	9.1 nF	2.4 nF	3.792	227 kHz	274.742 kΩ (274 kΩ)	39.748 kΩ (39.2 kΩ)	293.8 Ω (294 Ω)

6(a) and (b) to realize a formula for the general case of a single-pole amplifier with 0-Hz gain A_{v0} and unity-gain angular frequency ω_0 (that is, open-loop bandwidth ω_0/A_{v0}). Once again we find that the zeros are unchanged, $G(0)$ and $G'(0)$ differ by the same factor k as in case 7.1 (Eq. (13)), and T_2, T_3, and T_5, with $T_2 > T_3 > T_5$, are the roots of the cubic in T:

$$T^3 - \left[D_1' + T_7' - \frac{1}{\omega_0} - \frac{N_1' - D_1' - T_7'}{A_{v0}}\right]T^2 + \left[D_2' + T_7'D_1' - \frac{N_1'}{\omega_0} - \frac{N_2' - D_2' - T_7'D_1'}{A_{v0}}\right]T - \left[D_3' + T_7'D_2' - \frac{N_2'}{\omega_0} - \frac{N_3' - D_3' - T_7'D_2'}{A_{v0}}\right] = 0 \tag{22}$$

where now

$$T_7' = \frac{N_3'}{(1 + \frac{1}{A_{v0}})D_3'\omega_0} = \frac{T_1'T_4'T_6'}{(1 + \frac{1}{A_{v0}})T_2'T_3'T_5'\omega_0}. \tag{23}$$

These formulas clearly generalize those of cases 7.1 and 7.2.

curve due to inadequate loop gain, is to adjust a *single* component's value, for example, R_1 or C_1 in case 7.1 and R_2 or C_2 in case 7.2. *This is incorrect,* for such a change will, according to the formulas in the upper third of Table 3, modify not only the requisite pole T_3' or T_5', but also the zeros T_1', T_4', and T_6' which our analysis shows should be left unchanged. Our whole thesis is that, by appropriate calculation, an extremely accurate design is

Table 6. T-sensitivities to component variations for the ideal case.

Sensitivity	Fig. 1(a)	Fig. 1(b)	Fig. 1(c)	Fig. 1(d)
$S_{R_1}^{T_3}$	1.000	0.922	0.922	0.993
$S_{C_1}^{T_3}$	1.000	0.998	0.783	0.783
$S_{R_2}^{T_3}$	0.000	0.078	0.078	0.007
$S_{C_2}^{T_3}$	0.000	0.002	0.217	0.217
$S_{R_1}^{T_4}$	0.078	0.000	0.000	0.127
$S_{C_1}^{T_4}$	0.783	0.745	1.000	1.000
$S_{R_2}^{T_4}$	0.922	1.000	1.000	0.873
$S_{C_2}^{T_4}$	0.217	0.255	0.000	0.000
$S_{R_1}^{T_5}$	0.000	0.078	0.078	0.007
$S_{C_1}^{T_5}$	0.000	0.002	0.217	0.217
$S_{R_2}^{T_5}$	1.000	0.922	0.922	0.993
$S_{C_2}^{T_5}$	1.000	0.998	0.783	0.783

Table 7. Measured frequency response error of circuit of Fig. 2(a) type, using theoretically calculated component values (from [26, Table IV]).

Frequency [kHz]	Error [dB]
0.01	−0.024
0.02	0.000
0.03	+0.004
0.04	+0.007
0.05	+0.010
0.07	+0.012
0.1	+0.013
0.2	+0.013
0.3	+0.009
0.4	+0.007
0.5	+0.005
0.7	+0.001
1	0.0 (ref.)
1.5	0.000
2	0.000
3	+0.001
5	+0.003
7	+0.003
10	+0.003
15	+0.001
20	−0.002
30	−0.010
40	−0.020
50	−0.033

achievable without the need for any trimming which, as indicated earlier, is extremely difficult to carry out successfully.

8. ADDENDUM

There appears to be some current interest in the use of passive RIAA deemphasis circuits, and since the design equations for such circuits are contained in the tables already presented, it is felt to be worthwhile to provide basic design data for such circuits here as well. The four networks of Fig. 1, when used in this configuration, give rise only to the two distinct circuits shown in Fig. 7 (no low-frequency rolloff is provided). It is found that the relevant design formulas are precisely those given in column 1 of Table 1(b) and (c) for the circuits of Fig. 7(a) and (b), respectively. Note that these circuits provide ideal high-frequency deemphasis, but the preceding flat preamplifier stage does not have the greatly increased high-frequency overload margin achieved by the corresponding active deemphasis circuits.

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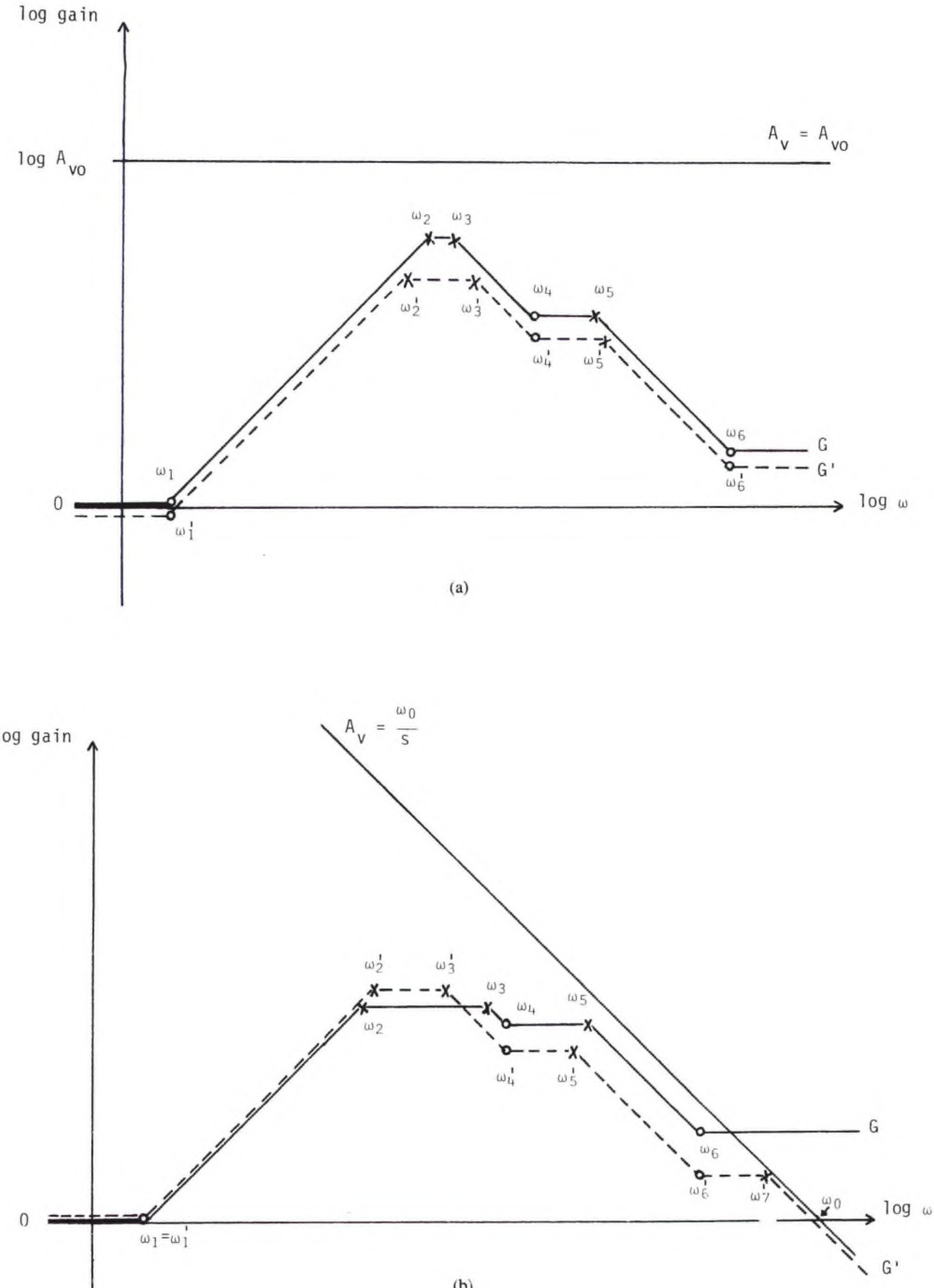

Fig. 6. Effects of inadequate loop gain. (a) Constant open-loop gain: $A_v = A_{v0}$. (b) Integrating open-loop gain: $A_v = \omega_0/s$.

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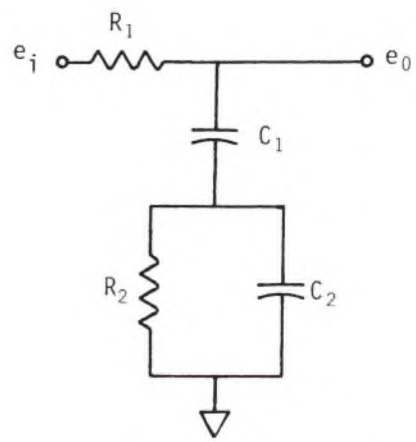
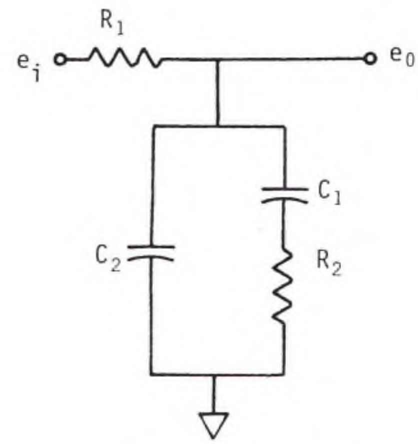
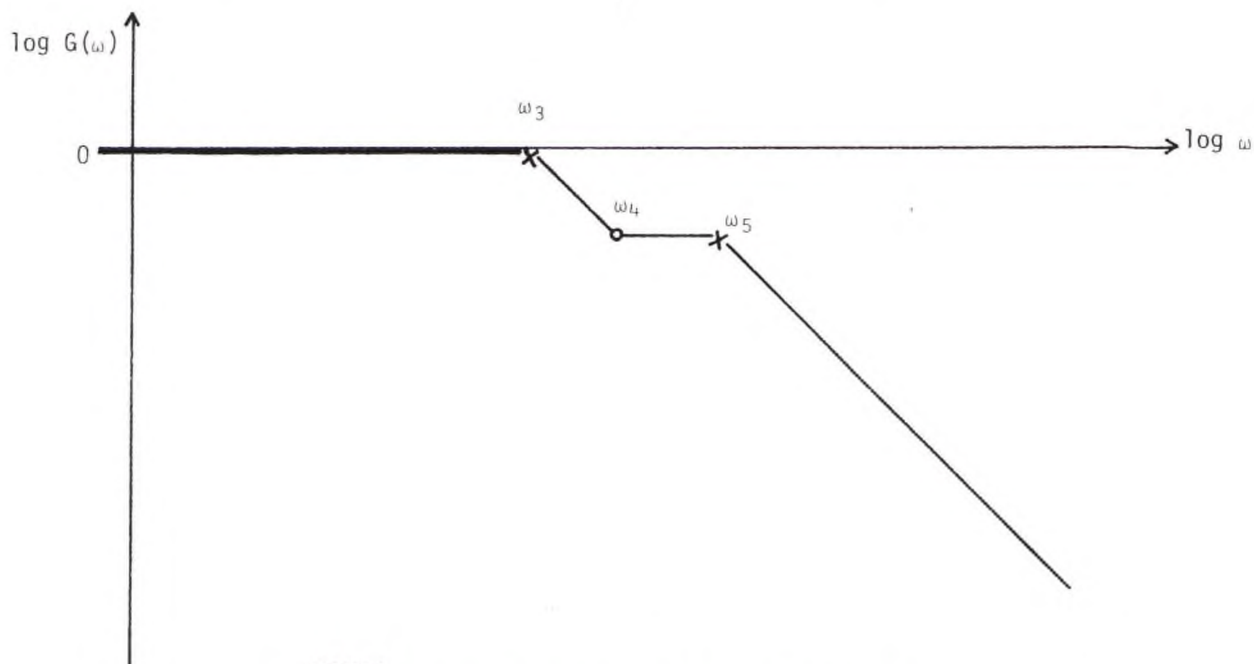


(a) Use Table 1(b) with $R_3 = 0$



(b) Use Table 1(c) with $R_3 = 0$



NOTES:
1) R_1 includes source resistance. The shunting effect of the load resistance modifies R_1, G. At high frequency the load on the source is R_1.
2) *One* of the components can be chosen independently.
3) No gain adjustment is possible; the low-frequency gain is unity.
4) $G(\omega) = \sqrt{(1 + T_4^2\omega^2)/[(1 + T_3^2\omega^2)(1 + T_5^2\omega^2)]}$.

Fig. 7. Passive deemphasis circuits.

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APPENDIX 1

An Example of the Calculations Leading to Tables 1 and 2

As an illustration of the procedure used, we shall consider Fig. 2 with the network of Fig. 1(a). Substituting into Eq. (1) for $Z(s)$, given in Fig. 1(a), and equating the right-hand sides of Eqs. (1) and (2), we obtain, after some simplification:

$$\frac{(R_A + R_3)}{R_0} \cdot \frac{R_0C_0s\left\{1 + \frac{R_1R_2(C_1 + C_2) + (R_1C_1 + R_2C_2)R_3}{R_1 + R_2 + R_3}s + \frac{R_1C_1R_2C_2R_3}{R_1 + R_2 + R_3}s^2\right\}}{(1 + R_0C_0s)\{1 + (R_1C_1 + R_2C_2)s + R_1C_1R_2C_2s^2\}}$$

$$= \frac{R_A + R_3}{R_0} \cdot \frac{T_2s\{1 + (T_4 + T_6)s + T_4T_6s^2\}}{(1 + T_2s)\{1 + (T_3 + T_5)s + T_3T_5s^2\}} \,.$$

We now equate the coefficients of corresponding powers of s in the numerators and denominators on both sides of this equation, and so reduce it to the following system of five equations:

$$\left.\begin{aligned} T_2 &= R_0C_0 \\ T_3 + T_5 &= R_1C_1 + R_2C_2 \\ T_3T_5 &= R_1C_1R_2C_2 \\ T_4 + T_6 &= \frac{R_1R_2(C_1 + C_2) + (R_1C_1 + R_2C_2)R_3}{R_1 + R_2 + R_3} \\ T_4T_6 &= \frac{R_1C_1R_2C_2R_3}{R_1 + R_2 + R_3} \end{aligned}\right\} \quad (24)$$

for the five unknowns T_2–T_6, for which it is easily solved:

$$T_2 = R_0C_0$$
$$T_3 = R_1C_1$$
$$T_5 = R_2C_2$$

and, if $R_3 = 0$,

$$T_4 = \frac{R_1R_2}{R_1 + R_2}(C_1 + C_2)$$
$$T_6 = 0$$

while, if $R_3 \neq 0$,[15]

$$T_{4,6} = \frac{[R_1R_2(C_1 + C_2) + (R_1C_1 + R_2C_2)R_3] \pm \sqrt{[\cdot]^2 - 4R_1C_1R_2C_2R_3(R_1 + R_2 + R_3)}}{2(R_1 + R_2 + R_3)} \,.$$

This latter expression for T_4 and T_6 can now be approximated by standard expansion techniques to derive expressions for T_4 and T_6 which are accurate to first order in R_3, provided $R_3 << R_2$. In this way we obtain the formulas in the upper third of Table 1(a).

From the design point of view it is, however, more useful to have expressions for the values of the resistors and capacitors R_0, R_1, R_2, R_3, C_0, C_1, C_2 in terms of the desired network time constants T_2–T_6. To this end one returns to the system (24), which viewed from this point of view is a system of five equations in seven unknowns, two of which can thus be chosen arbitrarily. We choose to specify R_0 and R_3 beforehand, for they will usually have their values circumscribed by noise and stability considerations. We thus solve system (24) for R_1, R_2, C_0, C_1, C_2 in terms of R_0 and R_3 and, after rather laborious calculations, come up with the formulas which constitute the middle third of Table 1(a). Finally these formulas are combined, eliminating R_0 and R_3, to derive the formulas given in the lower third of the table. These are useful, for they tell us the correct values to expect for the individual RC products and resistor and capacitor ratios in terms solely of T_2–T_6.

It should be remarked that, in the case of the first column of Table 1(a), $R_3 = 0$, and so $T_6 = 0$ and system (24) reduces to a system of only four equations in the six unknowns R_0, R_1, R_2, C_0, C_1, C_2. Now R_0 together with *any one* of the remaining components may be chosen arbitrarily beforehand. The formulas in the second column of Table 1(a) reduce to the "ideal" formulas in the first column as $R_3 \to 0$ (that is, $T_6 \to 0$); the latter thus remain useful as good approximations if R_3 is small. Note that these formulas all remain valid also for the case of Fig. 2(a), that is, as $C_0 \to \infty$. Changing C_0 affects only T_2, leaving T_3–T_6 unchanged. An interesting point is that T_3 and T_5, corresponding to the poles of $G(s)$, occur at *precisely* the poles of $Z(s)$ itself, even when $R_3 \neq 0$, whereas the middle RIAA time constant T_4 is *increased* in value from the zero of $Z(s)$ when $R_3 \neq 0$. This is true for all four networks of Fig. 1 and also for the circuits of Figs. 3, 4, and 5.

Table 2 is derived by putting $s = j\omega$ in Eq. (2) and calculating the magnitude $G(\omega)$ of $G(j\omega)$. The cases with and without C_0 must both be considered, the latter case being obtained from the former by letting $C_0 \to \infty$ (that is, $T_2 \to \infty$). The alternative expressions given in certain cases follow by the use of Table 1. Also given are the limiting values of the low-frequency gain $G(0)$ and the high-frequency gain $G(\infty)$.

[15] Again, the symbol $[\cdot]$ used in some of the formulas denotes a repetition of the square-bracketed expression which precedes it within the same formula.

APPENDIX 2

An Example of the Calculations Leading to Tables 3 and 4

We substitute into Eq. (7) for $Z(s)$ from Fig. 1(a) and equate the right-hand side with that of Eq. (8) to obtain

$$\begin{aligned}
&(1 + [(R_0 + R_3)C_0 + (R_1 + R_2)C_0 + R_1C_1 + R_2C_2]s \\
&\quad + [(R_0 + R_3)C_0(R_1C_1 + R_2C_2) + R_1R_2\{C_0(C_1 + C_2) + C_1C_2\}]s^2 \\
&\quad + [(R_0 + R_3)C_0R_1C_1R_2C_2]s^3) / \\
&\quad (1 + R_0C_0s)\{1 + (R_1C_1 + R_2C_2)s + R_1C_1R_2C_2s^2\} \\
= &\quad (1 + (T_1 + T_4 + T_6)s + (T_1T_4 + T_1T_6 + T_4T_6)s^2 + T_1T_4T_6s^3) / \\
&\quad (1 - T_2s)\{1 + (T_3 + T_5)s + T_3T_5s^2\}
\end{aligned} \tag{25}$$

Comparison of the numerators and denominators on each side of this equation leads at ónce to the formulas in the upper third of Table 3(a). Note again that the poles T_3, T_5 are exactly the same as those of $Z(s)$. Unfortunately it is impractical to provide approximate formulas for the zeros T_1, T_4, and T_6, but they can be evaluated from the given cubic equation by standard techniques in any particular case.

To derive the remaining formulas in Table 3(a) one first equates the coefficients of corresponding powers of s in the numerators and denominators of Eq. (25) to obtain the following system of six equations:

$$\left.\begin{aligned}
T_2 &= R_0C_0 \\
T_3 + T_5 &= R_1C_1 + R_2C_2 \\
T_3T_5 &= R_1C_1R_2C_2 \\
T_1 + T_4 + T_6 &= (R_0 + R_3)C_0 + (R_1 + R_2)C_0 + R_1C_1 + R_2C_2 \\
T_1T_4 + T_1T_6 + T_4T_6 &= (R_0 + R_3)C_0(R_1C_1 + R_2C_2) + R_1R_2\{C_0(C_1 + C_2) + C_1C_2\} \\
T_1T_4T_6 &= (R_0 + R_3)C_0R_1C_1R_2C_2
\end{aligned}\right\} \tag{26}$$

in the seven unknowns R_0, R_1, R_2, R_3, C_0, C_1, C_2. Any *one* of these quantities can thus be chosen arbitrarily, and the obvious choice would appear to be R_0. But in view of the fact that R_0 and R_3 occur in the combination $(R_0 + R_3)$ everywhere in system (26) except its first equation, it turns out that $(R_0 + R_3)$ is a better choice as independent variable. This choice also shows the parallels with the circuit of Fig. 3(a) more clearly. So we choose to solve system (26) for R_0, R_1, R_2, C_1, C_2 in terms of $(R_0 + R_3)$, and after much algebra obtain the formulas in the middle third of Table 3(a). Finally, by combining these results to eliminate $(R_0 + R_3)$ we deduce the lower third of the table. In the particular case $R_3 = 0$ system (26) contains a redundant equation, for then the T_i are constrained to satisfy

$$T_1T_4T_6 = T_2T_3T_5,$$

and so reduces to a system of five equations in six unknowns. Table 3(a) still correctly gives the results in this case, in terms of R_0 now.

Table 4 follows as before by setting $s = j\omega$ in Eq. (8).

THE AUTHOR

Stanley P. Lipshitz was born in Cape Town, South Africa, in 1943. He was educated in Durban and Pretoria, and received his Ph.D. in mathematics from the University of the Witwatersrand, Johannesburg, in 1970. Since then he has been assistant professor in the Department of Applied Mathematics at the University of Waterloo, Waterloo, Ontario. He has always had a keen interest in the whole field of audio and electroacoustics, and has recently become interested in some of the mathematical problems raised by the subject.

Dr. Lipshitz is a member of the Audio Engineering Society, and is the author of a previous *Journal* paper on pickup arm dynamics. Other current research ranges over a wide area, including error feed-forward audio amplifier circuit analysis, the improved characterization of amplifier distortions, and stereophonic recording techniques.

Digital Audio Disk Standardization Conference

STATUS OF THE DIGITAL AUDIO DISK (DAD) STUDY MEETING ACTIVITY

Date: 1980 February

1 DAD STANDARDIZATION CONFERENCE

The DAD Study Meeting was established as a suborganization of the DAD Standardization Conference for the purpose of studying the technology necessary for the standardization and coordination of the working groups which are in charge of actual business.

The concept of the DAD Standardization Conference is as follows:

1.1 The purpose of this newly established conference is to exchange various opinions on the standardization of the DAD system and to investigate standard specifications.

1.2 Twenty-nine hardware and software companies were represented in this conference. The establishment of this conference was announced to the press on 1978 September 26. Since then, the number of member companies has been increased to thirty-four.

1.3 The structure and function of the DAD Standardization Conference is shown in Fig. 1.

2 STATUS OF DAD STUDY MEETING AND WG ACTIVITIES

2.1 The Status of DAD Study Meeting Activity

Upon its establishment, the subjects to be discussed, and its schedule of operation were determined at the meeting.

The target specifications of the DAD system (performance, function, etc.) were discussed and made available at the end of 1978 by the DAD Study Meeting.

During 1979, based on the above target specifications, the DAD systems developed by each company were discussed and evaluated by the DAD Study Meeting.

Since its establishment, five meetings were held during 1978. The first four meetings concerned only a preparation and operation schedule for further meetings. The first target specification for DAD systems was established at the fifth meeting.

The DAD Study Meeting held five meetings during 1979.

After having received proposals on the DAD signal format for the DAD target specifications in 1979 April, the DAD Study Meeting asked WG-2 and WG-3 to discuss and review those proposals.

A new working group, WG-4, was set up for further discussions and evaluation.

At the end of 1979, the DAD Study Meeting renewed the first target specifications and decided to establish the "test format" of the DAD system. Details are given in Table 1.

2.2 Status of WG Activity

At the third meeting, the DAD Study Meeting established three WGs having different specialties, as subsidiary groups.

According to its specialty, each WG will take charge of actual business, such as, for example, the presentation of various experimental data for the study and investigation of specialized DAD system technology. The DAD Study Meeting further established WG-4 at the eighth meeting.

The role and status of each WG are as follows.

2.2.1 WG-1

WG-1 consists of software companies. The desired specifications of the DAD system will be discussed in this WG from the viewpoint of the software manufacturers.

During 1978 three meetings were held which discussed the importance of features that would enable the DAD system, upon its introduction into the consumer market, to coexist with audio records, and the possibility of replacing

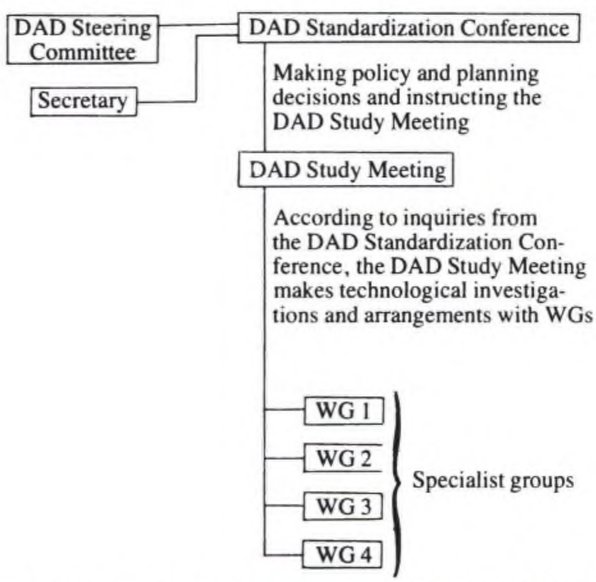

Fig. 1. Structure and function of DAD Standardization Conference.

Table 1. The target specifications of the DAD system prepared by DAD Study Meeting WG-2 and WG-3.*

Item	Content	Revised Requirements from WG-1	Content of Discussion by WG-2 and WG-3: Target	Remark
1. Performance	1. Frequency response	20 – 20 kHz	20 kHz (dc)	
	2. Dynamic range	80 – 90 dB	More than 90 dB	
	3. Distortion		Less than 0.003%	Harmonic distortion
	4. Channel separation	Average quality of the digital master tape	More than 80 dB	To consider discrete 4 channels
	5. Wow and flutter		Less than 0.003%	
	6. Number of uncorrected errors	Best effort is desired.	Less than 10 times/min	
	7. Number of uninterpolated errors		Less than 0.5 times/h	
2. Function	1. Random access	Absolutely to permit cueing of each recorded segment. (To permit repeat is necessary.)	To record address signals at intervals no greater than 1 second.	
	2. Prevention of copying	Prevention of digital copying is necessary.	Prevention of digital copying (to use a sampling frequency different from that of consumer PCM tape).	Prevention of 1) analog copying: impossible; 2) RF copying, and 3) replica copying. Recording address signals on the disk at short intervals will be effective.
3. Specification of disk	1. Diameter	Less than 20 cm is desirable.	30 cm maximum	
	2. Audio channels	4 (At the beginning 2 channels are permitted, but compatibility with 2-channel and 4-channel recording, character display and still picture at the same size, is desirable.)		The use of 4 channels is expected to be reexamined by WG-1.
	3. Playing time	40 min (continuous) 40 – 80 min (per disk)	4 channels, 40 min (continuous)	
	4. Recording method	Real-time mastering is necessary.		
4. Signal format	1. Quantizing bits	Linear; further discussion on 14 bits or 16 bits should be required.	16 slots	
	2. Sampling frequency	Same as the digital master.	Same as master.	
	3. Error correction and interpolation		Error correction necessary.	Interpolation alone will not be considered.
	4. Control signal		The signals for address (item 2.1) identification	Any other proposal on this matter should be discussed, if necessary, by WG-1.
	5. Modulation format		Pending	
5. Reliability	1. Life of disk	More than 100 plays for ordinary use.	More than 100 plays	
	2. Antidust method	To withstand handling (scratches, fingerprints, etc.) and environment (dust, etc.)		Should be accomplished somehow.
	3. Handling		The same or better than an ordinary audio disk.	
	4. Tolerance	Further discussion should be required.	Undecided	
6. Compatibility with video		A common format for all DAD systems world-wide, independent of the video format. Compatibility with player between video and DAD system is a matter of further discussion.	To use at least common pickup devices and rotation mechanisms.	Make audio/video selection by switching possible.
7. Production		Compatibility with present mastering process and production facility should be considered. Recycling of material is very important.		
8. Miscellaneous	Portable player	Necessary		

* Table is based on the results of the joint meeting of WG-2 and WG-3 considering WG-1 reference (WG-1 53-3, 4, 5).

audio records with the DAD system in the future.

WG-1 had established requirements for the DAD system from the viewpoint of software manufacturers, and further renewed the requirements in 1979 October.

WG-1 consists of the following companies: JVC, Nippon Columbia, Victor Musical Ind., King Records, Teichiku, Polydor, Toshiba-EMI, CBS, and Sony.

2.2.2 WG-2

The role of WG-2 is to discuss the signal format, performance, and function of the DAD system. A questionnaire concerned with each company's technology was delivered, and based on the results of this questionnaire, the target specifications were developed after three meetings and presented to the DAD Study Meeting at the end of 1978.

WG-2 held joint meetings with WG-3 starting with its second meeting, as described under 2.2.3.

WG-2 consists of the following companies: Pioneer, Toshiba, Sanyo, Sony, TEAC, JVC, Hitachi, Matsushita, and Mitsubishi.

2.2.3 WG-3

WG-3 was established especially for the study of error-correction technology. However, up to now the subjects discussed by WG-3 were similar to those of WG-2. Therefore, WG-2 and WG-3 were combined. The second and third meetings were held as combined meetings of WG-2 and WG-3.

After having established the first target specifications for DAD systems, WG-3 requested the DAD Study Meeting to ask DAD members for proposals on DAD systems.

Upon receiving proposals from 13 domestic and foreign firms, WG-3 reviewed these proposals.

WG-3 consists of the following companies: Mitsubishi, Sony, Sharp, TEAC, Nippon Gakki, Hitachi, Sanyo, The General Corp., Pioneer, JVC, Matsushita, and Toshiba.

2.2.4 WG-4

WG-4 was established for the purpose of reviewing, discussing, and thus generating the DAD signal format which will fulfill DAD target specifications.

By the end of 1979 five meetings were held. WG-4 has been actively discussing and reviewing concepts for generating a tentative DAD signal format with a view to fulfilling the DAD target specifications proposed by WG-1.

WG-4 consists of the following companies: Pioneer, Mitsubishi, Sanyo, Sony, Sharp, TEAC, Nippon Gakki, Hitachi, JVC, Nippon Columbia, AEG-Telefunken, Philips IDCC, The General Corp., Toshiba, and Matsushita.

related reading

Bibliography of Disk Recording

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The following bibliography on disk recording covers the period from 1921 to 1947. As originally published it was in chronological form. To this has been added an alphabetical list arranged by authors. In both lists a brief abstract of the article is included. This bibliography is the first of four which the *Journal* plans to publish. The others will include a disk recording bibliography covering papers published between 1947 and the present, and bibliographies on tape recording and stereophonic recording and transmission.

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H. D. Brailsford, Recording Machinery Noise, *Electronics*, **15**, 46-51, 164-166 (November, 1942). Cutting and playback apparatus used for accurately recording sounds made by industrial and other equipment. Frequency spectrum analysis is then performed with oscilloscope, camera and sound analyzer.

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R. W. Ehrlich, Volume Expander Design, *Electronics,* **18**, 124-127 (December, 1945). Electronic volume expander provides full expansion in about 10 milliseconds and returns to normal gain in 1 second. Details of construction and oscillograms showing performance.

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A. Badmaieff, Push-Pull Frequency-Modulated Circuit and Its Application to Vibratory Systems, *J. Soc. Motion Picture Engrs.,* **46**, 37-51 (January, 1946). Push-pull action is accomplished by varying the resonant frequencies of both the oscillator and the discriminator in opposite phase relation to each other. Two capacitors with common plate achieve push-pull modulation. Applicable in the calibration of recording heads and the determination of their frequency response and distortion.

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A. N. Butz, Jr., Surgeless Volume Expander, *Electronics,* **19**, 140, 142 (September, 1946). Single-ended expander requiring no push-pull transformer for canceling out the dc surge in the expander tube. Dynamic change in signal tube plate current is balanced out by opposite change of screen current in dummy tube.

A. J. Campbell, Lateral Disk Recording at the Naval Research Laboratory, *Communications,* **26**, 11-15, 50 (September, 1946). NRL recording system facilities and performance. Modifications made on commercial equipment used in the laboratory.

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B. B. Bauer, Measurement of Recording Characteristics by Means of Light Patterns, *J. Acoust. Soc. Amer.,* **18**, 387-395 (October, 1946). Analytical treatment of the geometry of optical patterns formed by light reflection from record groove walls. Influence of angles of light incidence and reflection, included groove angle, and distances between record, light source, and observer.

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W. F. Leidel, Jr., and **N. E. Payne**, Tuned-Ribbon Pickup, *Electronic Inds.*, **5**, 67-69, 100-101 (October, 1946). New reproducer extends playback range to 15 kc, providing at the same time "magnetic cushion" for record noise suppression. Discussion of the various forces acting on a pickup during playback.

I. L. Capps, Recording Styli, *Electronic Inds.*, **5**, 65, 67, 100, 102, 104, 106, 108, 110 (November, 1946). Importance of stylus contour, cutting edge, and burnishing facet is stressed. Correlation between the stylus radius cutting pitch and cutting depth is pointed out.

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L. D. Grignon, Three-Band Variable Equalizer, *Electronics*, **20**, 112-115 (January, 1947). Flexible pre-equalizer provides gain or attenuation adjustment in 1-db steps independently in the low-, mid-, or high-frequency audio ranges.

R. V. Howard, PROGAR—Program Guardian, *Frequency Modulation Business*, **2**, 24-25, 36 (January, 1947). Description of "program gain automatically regulated" amplifier which is a fast-acting peak limiter preceded by an automatic gain control amplifier with variable time constants.

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W. H. Robinson, Lateral Recording, *Communications*, **27**, 26, 28 (February, 1947). Electromagnetic and crystal cutters, groove depths, disks, volume indicators, measuring setups, frequency runs, styli, cutting angles, scratch filters, and pickups.

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Kalmus, H. P., Pickup with Low Mechanical Impedance, *Electronics,* **19**, 140-145 (January, 1946). Amplitude modulation is produced in a radio-frequency oscillator owing to variation in its losses, which is caused by the movement of a metallic vane attached to the stylus. Pickup has low mechanical impedance.

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Kaufmann, K., Saja, Single-Phase Synchronous Motor for Recorders, *Elektrotech. Z.,* **55**, 213-214 (March, 1934). Data for a heavy-duty type motor, with hand starting and a speed of 78 rpm at 50 cps line frequency.

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Keller, A. C., Direct Recording and Reproducing Materials for Disk Recordings, *J. Soc. Motion Picture Engrs.,* **28**, 411-426 (April, 1937). Materials available for direct disk recording classified chemically in five groups, and measurements of frequency characteristics, surface noise, life, and distortion. Data for both lateral and vertical recording.

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Knapp, D. G., Offset Head Crystal Pickup, *Communications,* **18**, 28-29 (February, 1938). An exact formula is derived for optimum value of the ratio of overhang to pickup arm length (for nearly constant tracking error).

Kornei, O., On the Playback Loss in the Reproduction of Phonograph Records, *J. Soc. Motion Picture Engrs.*, **37**, 569-590 (December, 1941). A theory explaining the level losses occurring in playback of laterally cut records at higher audio frequencies. The performance of a spherical pickup stylus from the viewpoint of an elastic deformation of the record groove walls. Playback loss and translation loss related to pickup design.

LeBel, C. J., Advanced Disk Recording, *Electronics*, **11**, 34-36, 82 (November, 1938). How to extend the frequency range of a recording system, reduce its distortion, and control the signal-to-noise ratio, factors which make the difference between passable and excellent results.

Direct Disk Recording, *Electronics*, **11**, 22-25 (March, 1938). In producing high-fidelity instantaneous recordings, careful attention has to be paid to turntable, cutting head, record blank, and reproduction system.

Disk Cutting Problems, *Electronics*, **12**, 17-19 (December, 1939). Problems of high-frequency response, harmonic distortion, disk durability, and noise level in instantaneous recording. Practical advice on how to overcome some of the difficulties and how to make performance checks on a number of units in a recording system.

Extended Experimental Study of the Optical Pattern, *Communications*, **20**, 22, 24 (December, 1940). A discussion is presented of the influence of hum, surface noise, and deviation from exactly lateral cut on an optical test pattern. Causes of unsymmetrical patterns and multitone test bands.

High-Frequency and Noise Level Characteristics of an Instantaneous Recording Disk, *A.T.E. J.*, **8**, 6 (January, 1941).

Properties of the Dulled Lacquer Cutting Stylus, *J. Acoust. Soc. Amer.*, **13**, 265-273 (January, 1942). Experimental study of the influence of burnishing standard lacquer cutting styli. Relations between recorded velocities, frequency and recording diameter for different lengths of dulling.

Quality in Disk Reproduction, *Electronics*, **10**, 25-27, 77 (October, 1937). Turntables and pickups and suggestions for improvement in reproducing systems given.

Recent Improvements in Recording, *Electronics*, **13**, 33-35, 79-81 (September, 1940). New mechanical designs of turntables, improved cutters, playback heads, and equalizers show the trend toward higher fidelity in disk recording.

Research Beats the Priorities, *Electronics*, **14**, 27-30, 78, 80, 82-83 (October, 1941). Replacement of aluminum by glass as a base material in disk recording and some of the new techniques in using this material.

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Leitner, R. G., New Vibrating Reed Magnetic Pickup, *Radio*, **29**, 25, 63 (December, 1945). Design, construction, and operating characteristics of a new type of magnetic pickup. Frequency-response curves with and without damping.

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Lindstrom, E., Recording Disks, *Ericsson Rev.*, 128-132 (1938). Records coated with cellulose lacquer giving good sound quality combined with freedom from noise.

Loefgren, E. G., On the Nonlinear Distortion in the Reproduction of Phonograph Records Caused by Angular Deviation of the Pickup Arm, *Akust. Z.*, **3**, 350-362 (1938). Theoretical investigation into pickup tracking. Sinusoidal and multitone groove modulation. It is shown that in the latter case the distortion represented by the combination tones may be quite considerable. Optimum off-set is determined for minimum distortion.

Lottermoser, W., see Gruetzmacher, M.

Lynch, E. T., Some Considerations in Phonograph Pickup Design, *Brush Strokes*, **3**, 3 (April-June, 1939).

Lynch, T. E., and Begun, S. J., General Considerations of the Crystal Cutter, *Communications*, **20**, 9-11, 26, 28, 29 (December, 1940). Rochelle salt crystal cutter, its operation and use in a recording system.

See Begun, S. J.

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MacDonald, G. E., Reduction of Pickup Tracking Error, *Communications*, **21**, 5-8, 22-24 (January, 1941). Definitions of tracking error and tracking angle and formulas for minimum tracking error conditions.

Mackenzie, D., see Albersheim, W. J.

MacNair, W. A., see Shea, T. E.

Massa, F., Permissible Amplitude Distortion of Speech in an Audio Reproducing System, *Proc. I.R.E.*, **21**, 682-689 (May, 1933). Experimental investigation of the effect of distortion on the character of audio reproduction in wide and narrow frequency channel systems.

Maxfield, J. P., and Harrison, H. C., Methods of High-Quality Recording of Music and Speech Based on Telephone Research, *J. Am. Inst. Elec. Engrs.*, **45**, 334-348 (February, 1926). Analysis of requirements for recording and reproducing sound. Complete description of application of electrical transmission theory to mechanical vibratory systems is most important feature of paper.

Maxwell, D. E., Automatic Gain Adjusting Amplifier, *Tele-Tech*, **6**, 34-36, 128 (February, 1947). Information on design and engineering of the CBS 1-A peak-limiting amplifier, characterized by short attack time to prevent splatter, low compression level, and elimination of pumping.

Meyer, E., see Buchmann, G.

——— and Just, P., Frequency Curves of Electric Pickups and Mechanical Gramophones, *Elek. Nachr.-Tech.*, **6**, 264-268 (July, 1929). Characteristic frequency curves for a number of pickups and the calibration of a test record.

Miessner, B. F., Frequency-Modulation Phonograph Pickup, *Electronics*, **17**, 132-133 (November, 1944). Stylus movement causes a capacitance change in an oscillator circuit, thus producing frequency modulation. Pickup is claimed to have flat frequency characteristic, self-contained volume expansion, and operates with less than one ounce of needle pressure.

Michel, B. M. H., see Goodell, J. D.

Miller, G. H., Improving Broadcast of Recorded Programs, *Electronics,* **7**, 178 (June, 1934). By using a wave filter and resistance pad during playback, giving a characteristic complementary to that used in recording, an over-all system with improved low-frequency response is obtained.

Miller, W., see Downs, G. W.

Miller, W. C., and Kimball, H. R., A Rerecording Console, Associated Circuits, and Const. *B* Equalizer, *J. Soc. Motion Picture Engrs.,* **43**, 187-205 (September, 1944). Two-position rerecording console with sliding volume controls and preset equalization, also a variable-type attenuation equalizer circuit.

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Misener, G. C., Optical Cueing Device for Disk Playback, *J. Soc. Motion Picture Engrs.,* **45**, 297-301 (October, 1945). Simple optical arrangement which permits accurate placement of pickup stylus on the disk for cueing purposes.

Morgan, A. R., see Kellogg, E. W.

Morgan, K. F., and Shea, T. E., The Electrical Engineering of Sound Picture Systems, *J. Am. Inst. Elec. Engrs.,* **48**, 766-770 (October, 1929). Methods and apparatus for sound recording and reproducing with emphasis on electrical engineering aspects.

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Niemann, A. W., Measuring the Recording System with Limited Equipment, *Communications,* **18**, 14-15, 36-37 (May, 1938). Procedures for checking recording equipment with relatively simple measuring apparatus. Discussed are: impedance measurements, frequency characteristics, peak load capacity, nonlinear distortion, and maximum permissible recorded amplitude.

Nixon, G. M., and Volkmann, J., Design of Recording Studios for Speech and Music, *Tele-Tech,* **6**, 37-39 (February, 1947). Data on the adjustment of reverberation time *vs* frequency characteristics to fit requirements for broadcast transcriptions and home records.

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