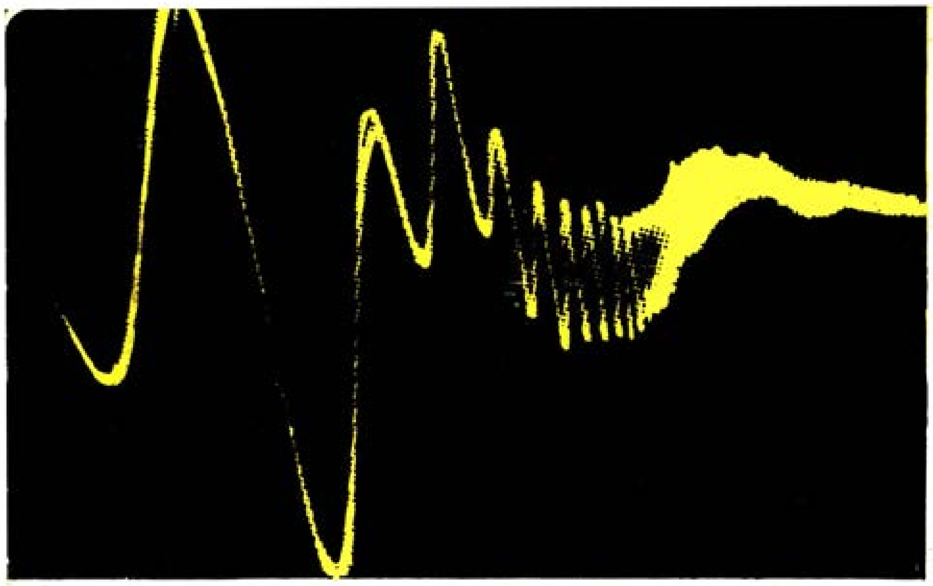
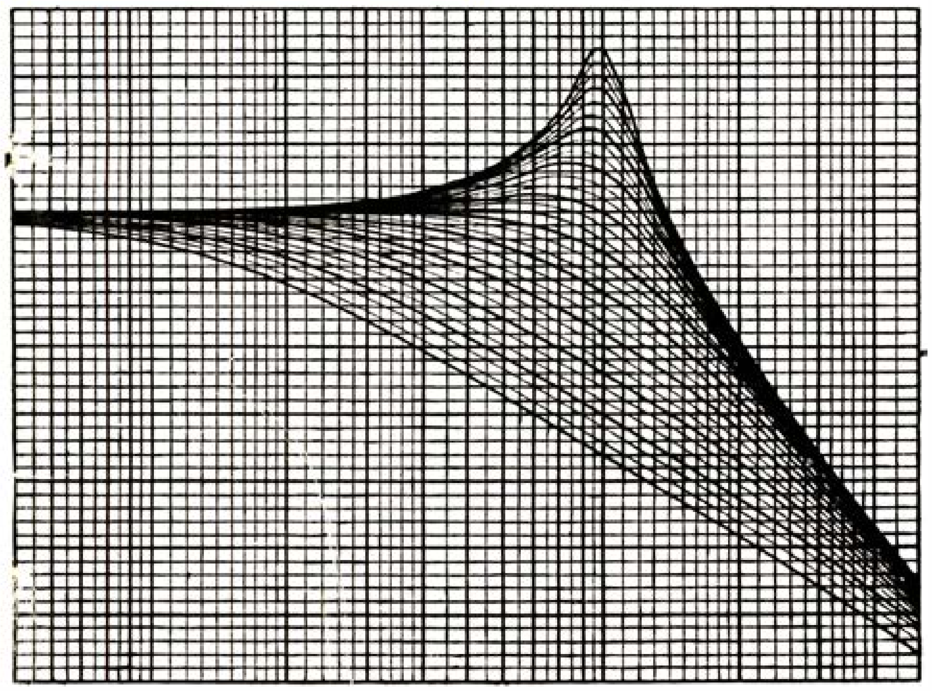
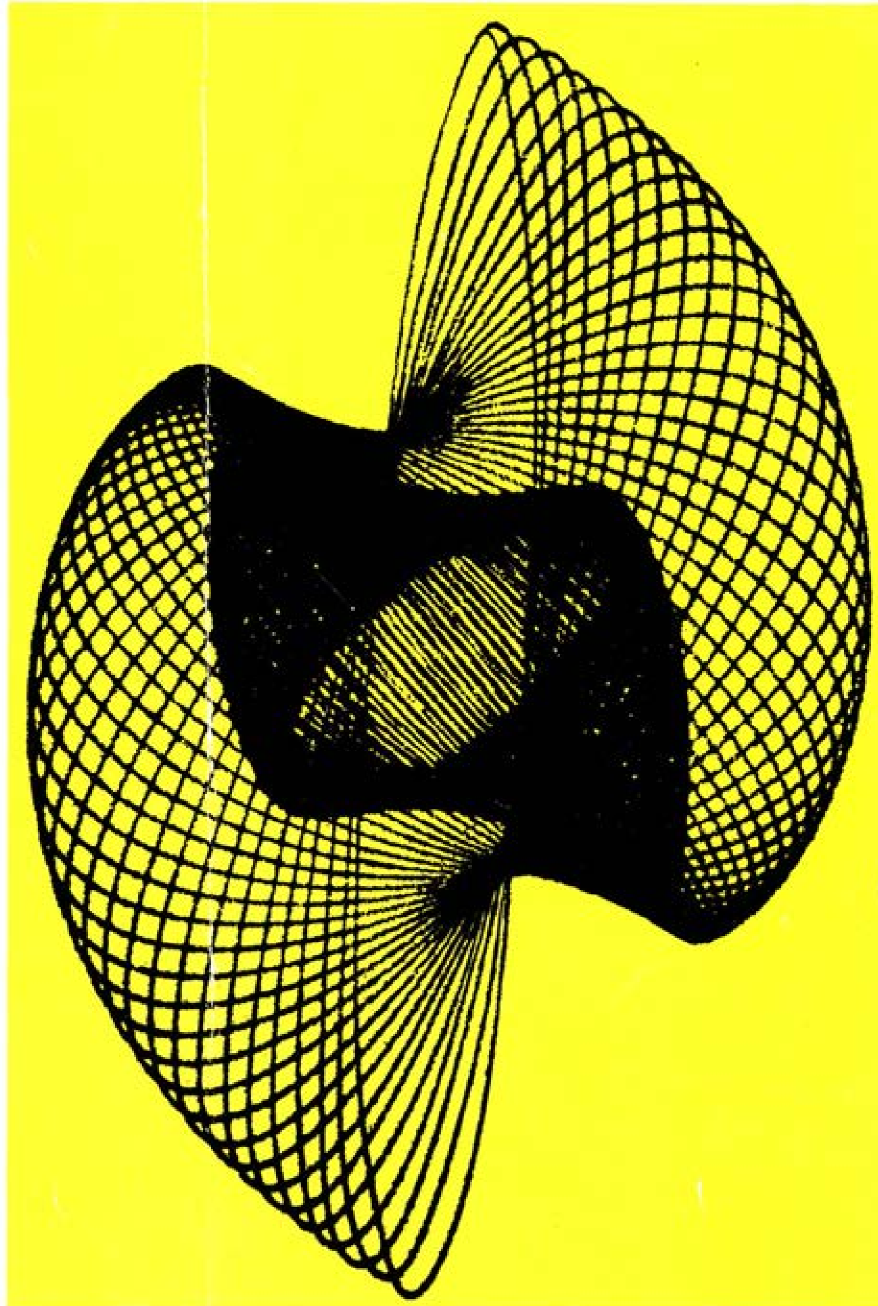


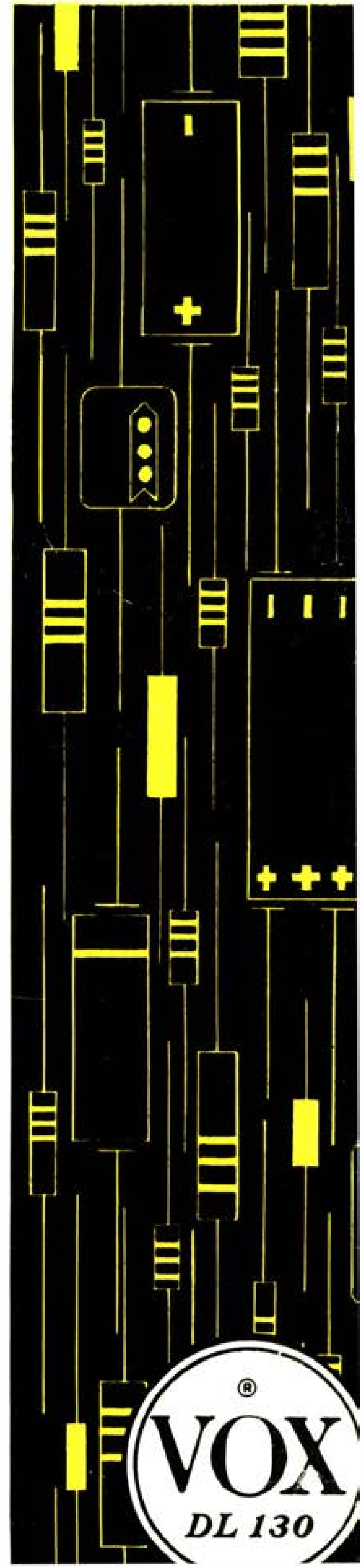
# THIS IS HIGH FIDELITY

written and produced by TYLER TURNER

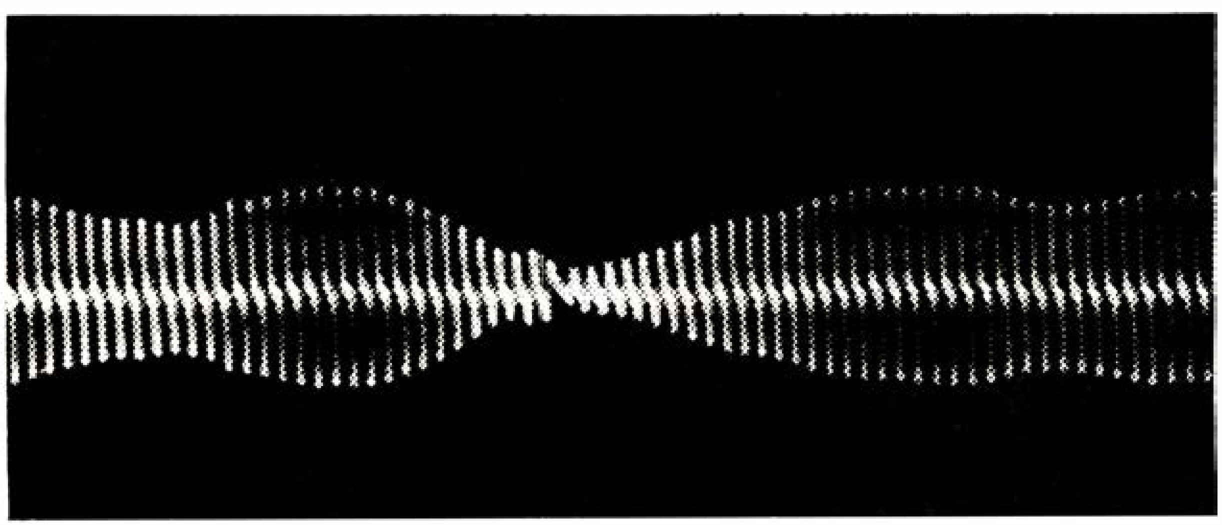
A comprehensive course in the principles of Audio, with tests for frequency response, intermodulation, resonances.











VOX®
DL 130

# THIS IS HIGH FIDELITY

## FOREWORD

by

P. WILSON, M.A. (Oxon.)

Technical Editor, "The Gramophone"

When I was asked to give an opinion on the American Edition of this brochure and record, I took them along to a friend of mine, a Professor at one of the London Schools of Music, and asked him to join me in following them through.

The Professor had no hesitation in saying that the course would be an invaluable aid to all schools of music: he particularly commented on the revealing way in which Bands 2, 3 and 4 of Side B of the record demonstrated the inner qualities of music. He could not think of any other method by which an understanding and appreciation of these qualities could be so effectively achieved.

This verdict was independently confirmed by another friend of mine whose profession is that of Acoustical Engineer: he also commented with enthusiasm on the success of the record in aurally demonstrating the various factors upon which faithful musical reproduction depends. In his judgment acquiring a copy of the record and brochure would be a **must** for all serious students.

I entirely agree with both these views. I feel, indeed, that I must warmly congratulate both Vox Productions and Mr. Tyler Turner, who devised this issue, on their achievement. I have heard many demonstration records of what is now called "High Fidelity" (or "Hi-Fi", if one must use a contemporary style), but none of them gives anything like the same insight of how musical quality is built up, and how various kinds of distortion may be identified. My only complaint, if it is a complaint, is that it has whetted my appetite for more: I found the fascinating musical illustrations all too short!

The record, then, needs no editing: nowhere could the universal character of both music and science have been more clearly displayed.

The brochure is described as a "comprehensive course in the principles of Audio", and that it certainly is. When I started to read it I expected to find a peculiarly American approach to which I might have some difficulty in adapting my British mode of thought. That fear was not realised. As one who is not inexperienced in these matters, I should have been proud to have written such clear explanations of the basic principles in such simple, yet colourful, English. Only a few verbal alterations, noted below, are required to make the brochure just as closely applicable to the conditions of the British market as it is to those in America.

Only in one respect do I think it necessary to make any reservation. That is to a statement at the top of the second column on page 8, where the combination of a "tone control" with a "volume control" appears to be recommended to deal with what is known as the "Fletcher-Munson effect". It is not yet known with any degree of certainty what adjustment should be made to frequency response to obtain the optimum degree of realism when music is being reproduced at a volume level below that of the original sound. Experiments are still in progress on this question. But whatever the answer may prove to be, most British Engineers and many American Engineers (I have no means of knowing whether they are in the majority or not) consider that the "tone-compensated" volume control is technically unsound.

That, however, is only a small matter and does not affect my warm recommendation of this record and this brochure to all students of music and, indeed, to all who wish to have a reproducing system of the highest possible fidelity.

### Emendations for British Edition

Pages 6-7 Former H.M.V. curve has not been published but can be taken to approximate to that of the "original L.P." Decca curve was the same as the "London".

Page 21 Column 1, last para.
For $50-60, read £15-20.
For 4-8-16 Ohms, read 3-7½-15 Ohms. These are more usual in Britain, but the respective differences are not of great importance

Page 21 Column 2, para. 2.
Single l.p. record players are now virtually obsolete in Britain. But Crystal pickups are in great demand and the best examples have qualities comparable with the best magnetic types.

Page 21 Column 2, para. 3.
Sapphire styli are not considered safe for more than 25 hours' playing time.

Column 2, para. 4.
Alternating current in G.B. has a frequency of 50 cycles. The stroboscope on the label of the British record has been drawn for this frequency.

Page 22 First sentence should read: "There are splendid 8″ and 10″ speakers available for between £3 and £8."


### ACKNOWLEDGEMENTS

*VOX PRODUCTIONS wishes to acknowledge the considerable aid and assistance of the following in the preparation of "This Is High Fidelity":*

*To Commander E. Jay Quinby, formerly of the Naval Sound School, Key West, for permission to use his frequency chart of musical instruments.*

*To R. D. Darrell and Philip Erhorn for helpful suggestions and ideas and to Art Hannes, whose narration adds so much to the enjoyment of this record.*


VOX ® DL 130

# THIS IS HIGH FIDELITY

by

TYLER TURNER

CONTENTS:

# 1

# THIS IS HIGH FIDELITY

A PERSPICACIOUS psychologist would probably find that the growing legion of audio enthusiasts are moved by three elemental drives. The first is the ability to command the services of several hundred musicians at the flip of a switch. The second is the peculiar thaumaturgy which transforms the modest confines of the average home to the dimensions of a concert hall by an aural framework of reverberant, 'big' sound. And the third is the ingrained association of sound with power, the thing which impels children to assert themselves by the great sounds which can emerge from small bodies.

But before such observations are allowed to profane what every audiophile treasures, one must rise to insist that a rose smells just as sweet, even though some Philistine would probably be glad to explain its charm by a silly chemical formula. It may be that man does not grow up, and that is probably a good thing. At least it can be a lot of fun as the Hi-fi enthusiast knows. His pursuit of good sound may take him far from the primitive response which powerful rhythms exercise over the listener. Or like the majority, he may always thrill to sound for its own sake.

Wherever he may stand on such a scale of values, this record will have something for him. Here, in sight and in sound, are set forth the factors which make good and poor audio reproduction. The test material has been chosen to serve three purposes: to help the user evaluate his equipment by exacting program material, to illustrate by direct comparison natural and distorted signals, and to suggest directions in which the user might look for possible improvements to his own equipment. Exaggerated conditions should enable the systematic listener to recognize symptoms of imbalance and distortion. The clarity with which defective or incomplete sections contrast with normal transmissions are a good gauge of the equipment on which they are heard.

For those who wish to torture their friends and neighbors, a suitable sequence of crashes, roars and thumps is available, guaranteed to melt the impassive. (A-III) But one hopes that the other aspects of musical reproduction will prove to be at least equally useful. It is true enough that a savage assault on piano, cymbals and drums imposes the heaviest strain on audio equipment. But these are also the most characteristic musical sounds, the least easily mistaken, and the ones which will appear most natural, even on poor equipment. No one could mistake a harpsichord, a loud cymbal or a loud bass drum. Heard alone and close up, they will identify themselves instantly.

Furthermore, an audio system may be soggy as a swamp, but there is some volume level at which it must command respect. And in the peculiar atmosphere of hazing session and end of the world which pervades the fidelity fiesta, the inflamed and intimdiated imagination easily comes to terms with such overwhelming aural forces, supplying the missing brilliance, the non-existent crispness and the independence which instruments should have of each other.

Stravinsky's sparkling march from *Histoire du Soldat* (A-III) should sound clear even on indifferent equipment. Its sparse and bright instrumentation has a sort of built-in clarity which can be obscured but not destroyed by poor audio components. If it really sounds faded, one can only be sure of it by hearing the same record on other equipment which will do it justice.

Ravel's *Rapsodie Espagnole,* on the other hand, would soon lose its haunting, ethereal magic if brought into the daylight. Yet there is black and black. Velvet is different from spilt ink. One invites, the other discourages, scrutiny. The sultry and voluptuous *Prèlude à la nuit* and *Malaguena* beguile the listener with their distant, caressing voices, their trumpets whispering on the horizon. Unless the listener hears them with impeccable clarity he will not know the subtle aroma which Ravel, probably alone among composers, could compound of sensuous sounds, tempos and deep perspectives. (A-III)

But the ease with which one can overlook these refinements of sound reproduction is, itself, a pitfall. Many an unsuspecting audio enthusiast gropes aimlessly about in a world of meretricious sonic novelties compounded largely of ballyhoo and magnificent packaging, adding one expensive part after another to a rig which never quite comes up to expectations.

There are several reasons why a piano may sound like a banjo or a marimba, a violin like an oboe, or a cymbal like rattling tissue paper. More often it is a combination of several reasons. If the whole problem could be charged up to frequency limitations, there would be happier electronic engineers. But it can't be. There are other factors as well. Perfectly reproduced sound will be

**WIDE RANGE** covering all the notes heard and suggested by the complex tonal patterns of many instruments playing together. (A-I)

**UNDISTORTED** by sympathetic or extraneous mechanical or electrical vibrations or interference. (A-III)

**PROPORTIONAL** to the original sound without accentuation or diminution. (A-II)

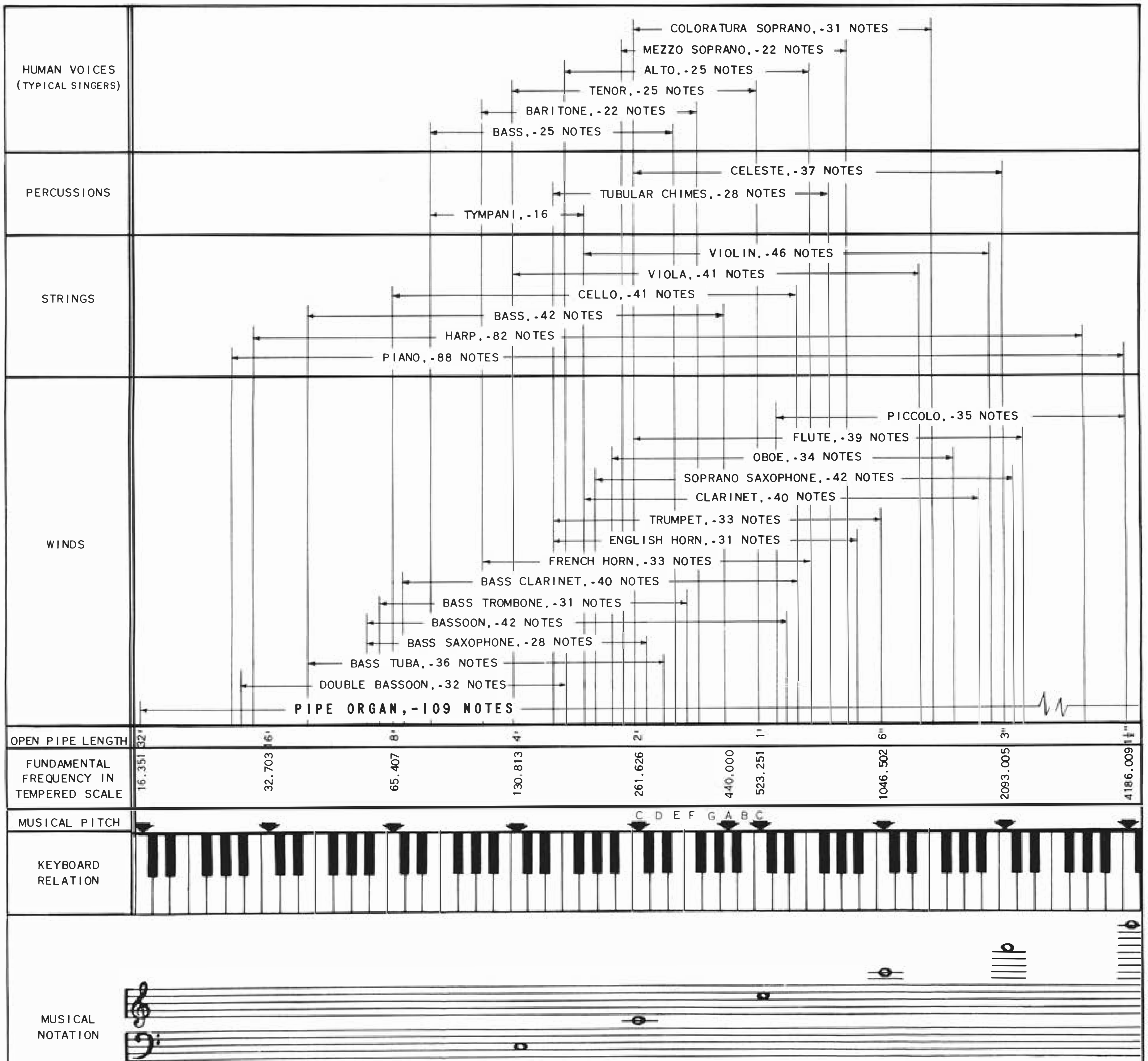
Those who have studied the aural consciousness are emphatic that an extended upper frequency range is agreeable only when it is very pure and balanced by a corresponding bass. It is therefore unsatisfactory to build equipment (or at least to use it) in such a way that it is more limited at one end of the spectrum than at the other. If the low frequencies are attenuated, a corresponding limitation of high frequencies will make the instrument more agreeable for steady listening. (A-II) If one can be extended, it is desirable to improve the other similarly. And if there is an appreciable distortion factor, a modest and balanced frequency band is advisable. (A-III)

The fundamental tones or pitches of various musical instruments range from below 20 to 8000 cycles per second with the organs extending both above and below the limits of the others. The overtones of most instruments extend beyond this band, and subaudible sounds and ultrasonic vibrations modify the associated sounds which we can hear. The only instruments which can be reproduced without detectable loss of fidelity with an 8000 cycle cut off are timpani, bass drum, bass tuba, piano and horn and trombone playing mezzo without mute. Even a double bass requires a band up to 9000 cycles and the overtones of all other instruments approach the upper limit of hearing. Yet the fundamental frequency range of a bass does not rise above 440, nor those of most of other instruments, above 8000 cycles. (A-I)

Most home appliances are limited to 5000 cycles. Professional equipment seldom reaches beyond 8000 cycles in uniform response except with constant, meticulous care. This is the point of diminishing returns for all except those who are prepared to pay dearly for the ultimate. To broaden the spectrum by one octave *with the quality which it requires* is as costly again as to reach the 8000 mark. I qualify the statement because most audio tinkers can beat twelve, fourteen or sixteen thousand cycles out of their systems with an oscillator or frequency track. But what one hears in those rarefied regions is more apt to be a squeal of protest than a clean signal. And if it is fun to emulate Jericho once in awhile, it is more fun for a steady diet to hear well-proportioned and comfortable sound. That may mean a range of 100 to 5000 cycles, or a range of 30 to 16000. Whether one or the other is entirely a matter of balance and purity which the recorded test material should help to determine.

Laboratory tests indicate that a band of 40 to 15000 cycles with a dynamic range of 70 decibels at less than one-half per cent distortion will reproduce the entire scope of orchestral music with no loss which is appreciable to the hearer. The organ would require at least another octave, and theoretically it should go to about 16 cycles.

That is the ideal. Those of us who must consider space and cost will approach it gradually by the judicious substitution of better components as cost allows. It will be useful to take some guidance from the illustrations of balanced transmission.




MUSICAL PITCH RELATION CHART
SHOWING COMPASS OF FUNDAMENTALS
(HARMONICS OR PARTIALS NOT INDICATED)
AMERICAN STANDARD PITCH, $A_{49}$=440

(A-II) In general, it may be assumed that:

| If the high frequency range reaches | The Low frequency range should go to |
|---|---|
| 5000 | 100 |
| 8000 | 70 |
| 10000 or above | 40 |

The second simple rule for good listening is that more distortion is tolerable in a narrow range than in a wide range system. Which means that an excessive frequency range is objectionable to the discriminating listener in an impure system. (A-III) Good listening therefore requires 1) balance between the ends of the frequency band and 2) freedom from distortion.

## PEAKING • EQUALIZATION

Extremes of the audio spectrum impose the greatest burden on the entire chain of operations from microphone through transmission, recording, manufacture, phonograph pickup, amplifier and speaker. The nature of vibrating bodies is to function efficiently at one frequency and not at others. A piece of metal, a taut wire or a glass tumbler will give a fixed note when struck. This is the 'resonant frequency.' Musical instruments are designed and proportioned for limited musical ranges. And within those ranges the size of their moving element is adjusted for pitch, shortened or lengthened, like the violin string or the column of air in a flute. The pipes of an organ vary in size by several thousand times to reach all frequencies and volumes of sound. But a single microphone, a single recording stylus, a single pickup element and usually a single loud speaker are entrusted with this same vast acoustical span. It is therefore inevitable that some of these links in the chain will discriminate. Electrical circuits have the same tendency, but to a degree now controlable.

**Resonance** or abrupt 'peaking' of frequencies in a limited region gives an unnatural quality to the sound—shrillness, boom or twang. It may originate at any point between the studio and the music heard from the speaker. When it is in the record it may have been caused by a resonant cutting head used to engrave the original record—an especially troublesome instrument—or in the microphones or their placement in the studio, in the listener's pickup cartridge or arm or speaker or cabinet or placement of the speaker. One recording company insisted upon locating a single microphone for symphony sessions in an alcove off the studio, thus adding an unpleasant nasal quality of the alcove to the sound of the orchestra. Some directional ribbon microphones, deadened on one side by internal screens, are bad offenders. Many records of the 1940's were stamped with their peculiarities. One was objectionably wiry. Another had a peculiar timbre suggesting the veiled, mordant quality of a viola which was superimposed on all program material. Effects of this kind are usually unnoticed except by direct comparison of normal with affected program material. Reference comparisons are supplied on the record according to the frequency curves. Practice will enable the listener to diagnose some of the common frequency peaks and he may be guided to recognize and correct defects in his own equipment. (A-II)

These peaked transmissions do not appear as an accentuation of certain notes because the frequency increments are so broadly distributed. Overtones and fundamentals of all instruments sounding in the peaked region are accentuated. For example the fourth harmonic of a violin sounding middle C would be strengthened by a 1000 cycle peak. But that accented harmonic is so integrated with the others as to leave our sense of the fundamental, middle C, unchanged. The violin has therefore been altered in timbre though our sense of its pitch has not been lost. The frequency components of such instruments as may be playing within the peaked band are similarly spread beyond the intensified region. So what is added in fundamental tone to instruments playing within that area is largely balanced by the exaggerated timbre of instruments playing in other registers. The condition would, however, be recognized as irregular scale reproduction in the case of the piano, flute and other instruments of similar pure tone. A booming loud speaker or cabinet which may peak within a very narrow band—two or three notes—is the worst and commonest, and most easily corrected, example. The acoustic spectra of musical tones will illustrate how well distributed the power of a musical note is among the various harmonics.

A **peaked frequency** must be distinguished from a **rising characteristic**. A recording characteristic which rises gradually toward the useful upper or lower frequency limit can be partially equalized, at least, by the conventional tone control. But a peaked frequency is one which rises abruptly within a limited compass, and which cannot be controlled without seriously disturbing the tonal balance.

# RECORDING CHARACTERISTICS

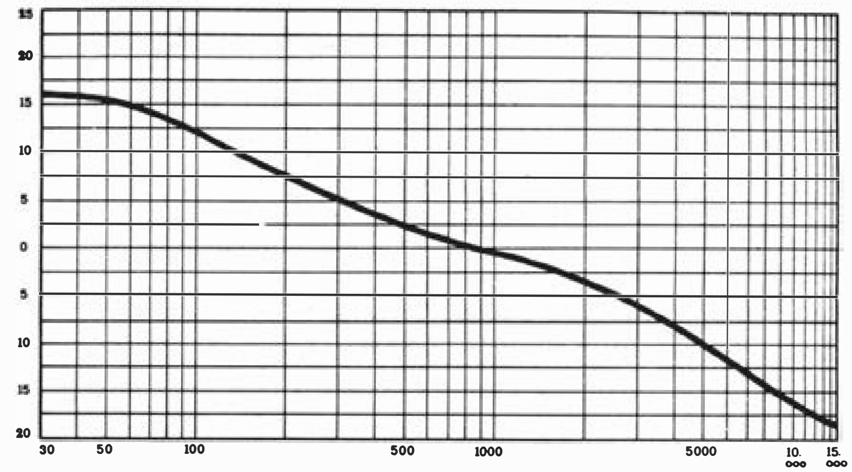
As we shall presently see, it is not possible to follow the acoustic scale of values directly in engraving a record. If one did so, a 50 cycle note would have a mechanical excursion 320 times greater than a 10000 cycle note of the same value. Strong bass notes would cause the stylus to cut from one groove into another since clearances between adjacent grooves must be limited by playing time. Soft high notes would fade to inaudibility or drop below the noise picked up from the granular structure of the record material. Signals are therefore modified according to a fixed pattern to meet these conditions before reaching the cutter heads or the recording machine. The record player must equalize, or restore the program material to its original character. When this is improperly done the sound is unbalanced or there are peaks at one or another point in the range. When a record sounds bad on first playing improper equalization is usually the cause.

All **recording characteristics** intensify the upper frequencies and minimize the low frequencies. Several recording curves are currently used, and complementary equalization curves are given for them. That endorsed by the Record Industry Association of America, RIAA, will probably become standard for all fine groove records. It represents a middle point between the other extremes and therefore a compromise single standard when variable equalization is either too costly or too confusing for the appliance buyer.

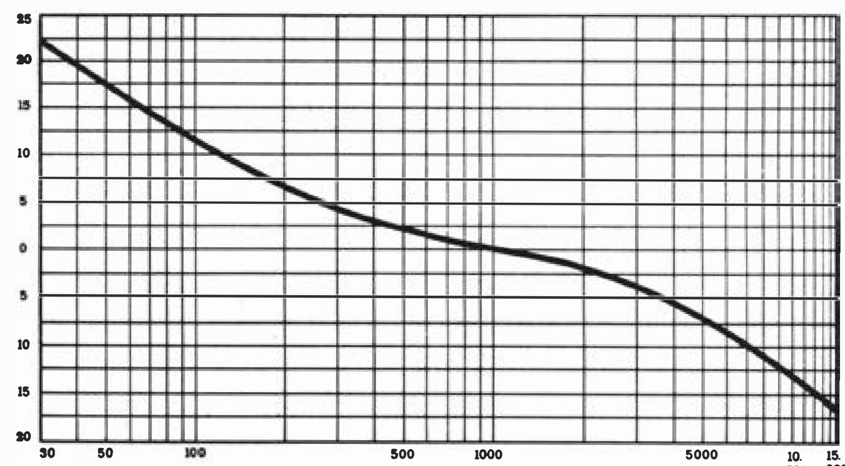
It is important to understand the function of the record characteristic: which is to pass sound energy most efficiently through the mechanical limitations of recording—and to distinguish that from the function of the tone controls: which is to trim the high or low frequencies according to the listener's taste. With a maximum groove modulation or amplitude established by physical dimensions, the amplitude of high frequency waves is augmented, and the amplitude of low frequency waves is confined, to that dimension. The exact pattern by which that is done is known as the recording characteristic.

The **playback characteristic**, then, should complement the recording characteristic, and the manner and degree to which it does so is illustrated. The **0 level** is the normal response reference, and 1000 cycles is the standard reference frequency to which the others are referred. The data are therefore related to a common level of 0 decibels at 1000 cycles, and show

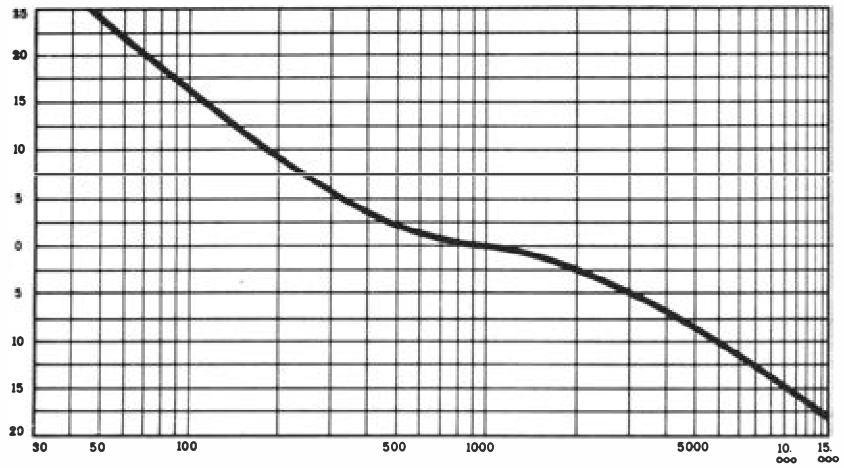

EQUALIZATION CURVE FOR NEW R I A A
(ALSO NEW A E S, N A R T B) RECORDING CHARACTERISTICS



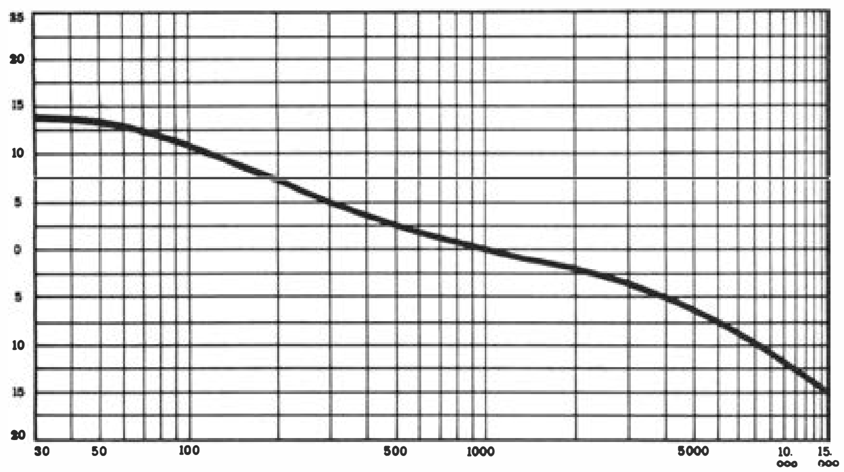

EQUALIZATION CURVE FOR ORIGINAL LP




EQUALIZATION CURVE FOR OLD AUDIO ENGINEERING SOCIETY




EQUALIZATION CURVE FOR OLD RCA

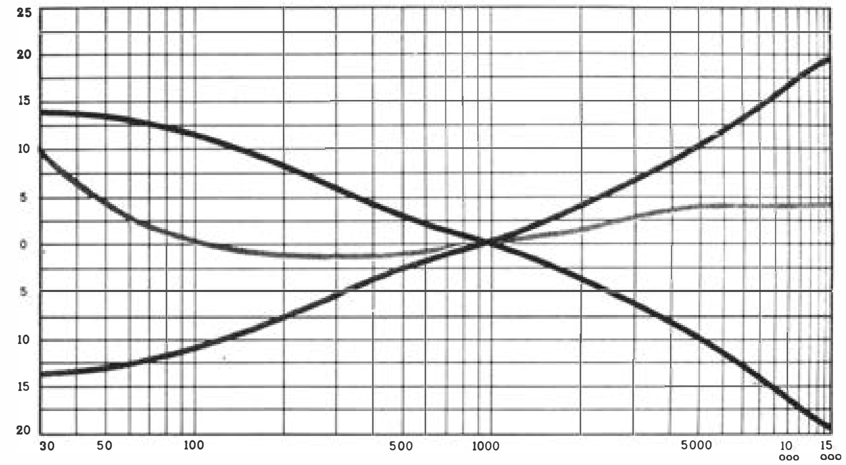



EQUALIZATION CURVE FOR LONDON ffrr LP

how far the other frequencies must depart from it to restore the signal to a normal or **'flat' response**. For example: 50 cycles on a record with the RIAA curve is almost 17 db below that of a 1000 cycle signal of the same loudness. The playback equipment must then raise this depressed level at 50 cycles by 17 db, to bring it up to its proper value as the curve indicates.

Since the recording characteristic determines the frequency pattern of the record, the equalization should theoretically be correct. But however exact it may be, the listener can have no assurance that the recording company has not added depth or brilliance artificially to the program material. Even though the recording curve is unchanged the frequency response may be altered en route to the recording characteristic network. Furthermore, the best of loud speakers have an erratic response which is acceptable only because of the ear's great tolerance of frequency irregularities.

For these and other reasons as well, the recorded sound may be improved by using a non-complementary playback characteristic. An examination of the old Audio Engineering Society (AES) curve will show that an original long playing record played back with it will be strengthened below 100 and above 1000 cycles. The resulting curve is unsatisfactory




A – THE OLD LP RECORDING CURVE
B – THE COMPLEMENTARY LP EQUALIZATION CURVE TO GIVE FLAT RESPONSE
C – THE RESPONSE OF MATERIAL RECORDED WITH CURVE A, BUT PLAYED BACK WITH THE OLD AUDIO ENGINEERING SOCIETY PLAYBACK CURVE

in appearance but compensates excellently for the weakness of much audio equipment in these two regions.

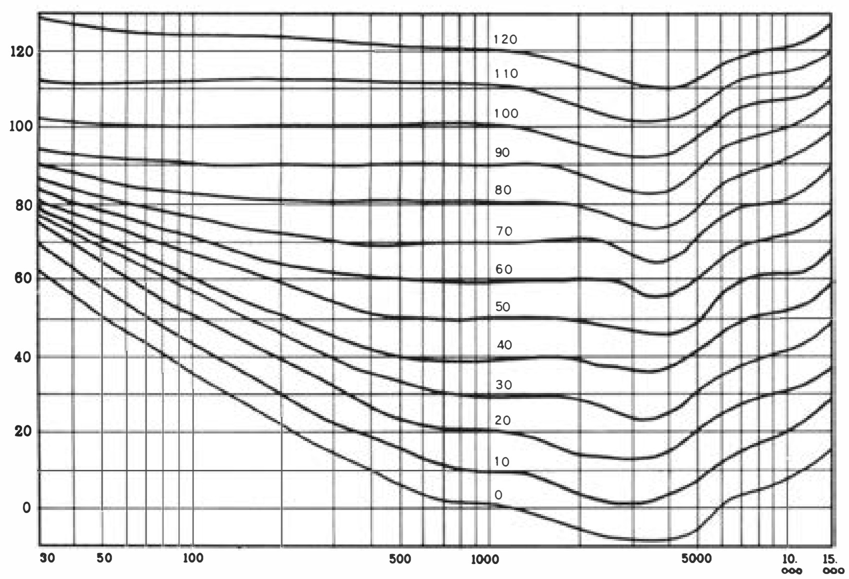
The **record equalizer** may be a fixed and invisible circuit within the chassis or it may be flexible and brought out to a control on the instrument panel. But in either case its function is to restore the electrical balance of the signal by suppressing the ascending frequencies which have been abnormally accentuated. Once that has been done, and the recorded signal flattened out, alterations in the tone balance can be made by the tone controls.

The basic equalization should not, however, be attempted by the tone controls. Many such devices have an unpleasant lump in the 100 cycle region. Loud speaker resonance and the effective limit of electrical system at about this point aggravate the condition. Timpani (which do not extend very low) monopolize the lower frequency region, blotting out 'celli basses, and low winds. Real bass is lost.

The intensity potential of kettle drums is greater than that of any other instrument except the organ (over 100 db) and as their power is concentrated mainly in the 100 cycle range they are among the most troublesome instruments to reproduce in scale, a plague to recording engineers, a curse to neighbors, but the great pride of those enthusiasts who glory in overwhelming masses of indigestible sound.

While there are some inescapable differences between an original and a recreated performance, the entire range of musical coloration is fortunately one of the things which can be preserved. If we do not hear the profound pedal notes in the Frescobaldi *Toccata* (A-III) or the tintinabulant cymbals in the *Capriccio Italien,* (A-I) we are not hearing all the original sound.

## FLETCHER-MUNSON EFFECT




CONTOURS OF EQUAL LOUDNESS IN DECIBELS AT DIFFERENT FREQUENCIES
(Fletcher–Munson Curves)

The acuity of our hearing varies with frequency, being greatest in the middle range and weakest at the extremes. If we are to hear with equal loudness the softest 32 cycle note (3 octaves below middle C) and the softest 1000 cycle note (about 2 octaves above middle C) the energy of the first must be more than 250,000 times that of the second. To maintain balance, the power must increase as the frequency decreases. (A-II) It must also increase as pitch ascends above 5000 cycles. Or to put it another way, we lose at both ends of the frequency spectrum as the volume decreases, and we hear last, at the lowest volume, the middle range from 800 to 5000 cycles. When the tremendous volume of a large orchestra is compressed for comfortable listening in a small room the *proportions* of the sound are similarly altered. The orchestra balances itself for conditions of the concert hall, not for the dimensions of the living room. While the middle instruments can whisper or roar, the basses must restrict themselves to a much narrower dynamic range. But when we reduce this scope all the way down the scale as we must for home consumption, we find that the lowest and highest frequencies suffer. To retain the musical proportions of the original it is necessary to amplify both the lows and the highs according to the contours shown. The 'loudness control' does this by combining the effect of a tone control with a volume control so synchronized as to support the low frequencies (and less often the high frequencies) as the volume is decreased. Those who do not have a loudness control will often feel that a rebalancing of tone controls is necessary for changes in the volume control setting.

## DYNAMIC RANGE

The maximum practical **dynamic range** which can be recorded on modern records is about 55 decibels. This compares very favorably with a theoretically perfect practical range of 70 db for music reproduction. Thus modern recording methods have virtually eliminated the restrictions in dynamic range as a handicap to the enjoyment of music through records. For those who may still feel the need of a wider range of volume than records contain, expanders are available which automatically augment loud passages to offset the monitoring of the recording engineers.

But there are two serious objections to be made to these devices. The first is that they must work from the amplitude (or energy) of the signal alone. Since they cannot distinguish between transient and sustained sound, ie., between a piano and strings, both are treated alike. And as each percussive note covers a wide dynamic range in itself, the expander will tend to increase the initial impact when the hammer hits and cause the tone thereafter to decay suddenly. The second objection arises from the directly proportional action of the automatic volume control as contrasted with the completely disproportional monitoring of the records. Records are not compressed evenly throughout the volume range. They are only pulled down a minimum on peaks of sound which might be impractical to engrave, and pulled up when the level sinks below that of the record surface noise. With the excellent dynamic range now possible on records and the limitations of space in which they must be heard, most listeners will probably prefer to make their own adjustments to taste.

# DISTORTION

Extended frequency response will be desirable only when it can be achieved without additions or distortion. (A-III) This is why 'response to 20000 cycles' does not mean anything unless it is supported by a tolerable distortion factor. Distortion is inevitable, and while it can be reduced, it is present to a measurable degree in all electrical music systems. If or while it cannot be avoided in a system to which improved components are gradually added, it is advisable to restrict the frequency range proportionately.

The ear will recognize only 1.25% distortion when frequencies are cut above 5000 cycles, but will recognize .7% distortion if the range extends to 15000 cycles. Percentages of distortion which are objectionable are ten or more at a range to 5000 cycles only, and only two at full range. In other words, the sensitive ear will accept a distortion factor if the frequency response is limited and yet be uncomfortable with a less distorted program at wide range.

An important effect of distortion is **listener fatigue** which materially reduces listening pleasure although the auditor will not recognize the cause. We can both hear and enjoy more music when it is undistorted. The test is not in how equipment sounds when we first hear it, but in whether it wears well for long listening. Such a test can be made by comparing the effects of a good and poor record reproducer, balanced to sound as nearly alike as possible, playing the same records, for extended periods. Unfortunately, this is seldom possible before equipment has been paid for!

In addition to limited or uneven frequency response, which is sometimes called frequency distortion, there are other conditions to which the word distortion is more properly applied.

**Harmonic distortion** occurs when the original tones subvibrate in multiples of their fundamental frequencies. This is the tendency of most vibrating mediums, as we shall see when we consider the nature of sound. It is damaging when contributed independently by the electronic apparatus. Improperly angled pickups are frequent offenders and may cause as much as 5% 2nd harmonic distortion on loud passages.

**Intermodulation distortion** is the effect which notes have on one another. The most common instance is the stuttering which an assertive bass will impose on high tones. Interaction of different frequencies also produces differential and product tones whose frequencies are the sum and the differences of the generating tones. A middle C sounded with the G above, for example, will give the effect of C an octave lower and an E an octave above. Amplifiers are capable of generating such products. Further interaction can take place when sound from the loud speaker jars the pickup arm causing it to vibrate or act as a microphone, picking up the sound a second time for reamplification. This can cause a closed electro-acoustic loop which oscillates with increasing intensity on the same note. Pickup needle assemblies which are too stiff to move freely transfer vibratory motion from the groove to the arm itself instead of the electrical generator in the arm, with disastrous results for the sound and the record.

**Phase distortion** occurs when components in the acoustic or electrical wave are out of step with one another. If two loud speakers were driven from the same signal source but with their connections reversed one would move forward while the other moved backward, and there would be a tendency at some points in the listening area for the sound of one to cancel that of the other. In this case they would be completely out of phase, or measured by the terms of angles and circles, they would be 180° out of phase. Usually phase distortion modifies the signal in subtle ways, making it soft and mushy instead of crisp and sharp. Like most distortion products, it has insidious effects on audio quality and listener reaction.

**Transient distortion** is the dilatory response of electrical or mechanical elements to sudden and rapidly changing agitation. We are usually conscious of the steady state of musical sound and less aware of its development and decay. Electrical and acoustical instruments can often deal effectively with a relatively fixed condition and yet fail to reflect almost instantaneous changes such as the impact of a piano hammer on the strings, drum rolls, cymbal crashes and other abruptly commencing and changing sounds. These are called transients because of the violence of their attack and their rapidly changing or transient character. It was transient distortion which retarded acceptable piano recording for so long, even while frequency range improved. . . . Noise is largely composed of transient sounds.

Such flexibility as will handle transients satisfactorily is closely related to frequency response. For, while a transient may not have supersonic frequency, it demands changes as rapid in the electric and mechanical systems. Abrupt sound cut-off is a simple test of transient response, revealing laggardness in the audio system, especially in the loud speaker, by persistence of the sound when the signal stops.

These and the various musical examples exploit the full capabilities of audio equipment and should disclose any tendency to distortion, intermodulation or blur. In the tumultuous climax to the 1812 overture, especially, defects should be apparent in confusion or unsteadiness of the high voices during the cannonade. The festive pandemonium should not obscure the strings, the choirs of woodwinds, of brass. Any tendency to muddle will destroy the delicate parallels and contrasts which make Ravel's Rapsodie Espagnole one of the great examples of subtle, suggestive orchestral timbre.

2

# THE NATURE OF SOUND

## From Oscillator to Orchestra (B-III)

THE FIRST attribute of sound is vibratory motion of the air, first one way and then the other. The second is the speed or frequency with which it changes its motion. There can be no sound which does not meet both of these specifications. We detect as sound the vibration of air particles at frequencies between 20 and 20000 cycles per second. When the air vibrates at other frequencies, it is not audible to man although some animals are sensitive to higher frequencies.

Audible displacement of air at any given point is infinitesimal except in explosions. One must not think of sound as a motion of air from point to point i.e., from a musical instrument to the hearer. A better simile is the motion of beads on a string. When one end is tapped the motion is conveyed to the others in order, while the beads retain their same relationship and substantially their same position: or like the ripples which rise and fall on the surface of an otherwise stationary body of water.

### SPEED OF SOUND

The properties of air are such that the motion we speak of occurs within a small radius and is transmitted to the surrounding atmosphere as variations in pressure. Since air has a certain density and elasticity it will transmit sound at a fixed speed. This is approximately 1120 feet per second, or the velocity of a pistol bullet. If you see a man drive a nail and hear the hammer blow just a second later, he is 1120 feet away because the sound will travel that distance in one second while vision is almost instantaneous. If he is farther it will take longer to hear, and if nearer, he will be heard sooner. If he were a mile away the sound would reach an observer 4.7 seconds after the hammer fell.

There can be no such thing as arrested sound, corresponding to a still picture. If the motion of air is stopped there is silence. Time is therefore an essential dimension of sound.

### GRAPHIC REPRESENTATION OF SOUND

So we can describe sound as a rate and character of vibration of air particles. This is easily diagrammed or charted. We first draw a straight horizontal line or 'axis' to represent the time element. Above and below this axis we plot the increased or diminished air pressure at time intervals. The most complex sounds can be reduced to these two dimensions of height (air pressure) and length (time interval). Such curves are described by various electrical and acoustical devices, the most familiar of which is the recording machine stylus.

Let us say that this curve has a duration of 1/440th of a second. The note, then, has a frequency of 440 cycles per second. In other words, the air goes through the changes indicated by the wave form 440 times each second. One complete cycle begins with the air at rest and continues through the increasing pressure, the decreasing and subnormal pressure and ends where the air returns to normal. The curve of a single cycle is sufficient to illustrate a simple note as it repeats without change for the duration of the sound. The maximum deviations of the wave are its **amplitude**.

All vibrating elements in the acoustical-electrical-mechanical chain follow analagous laws and conform to the same changes. So the waveform describes each stage in the metamorphosis of recorded sound: the impact of the air molecules upon the microphone element, the voltage which it generates, the flow of current in the amplifying tubes, the magnetization of the recording tape and the excursion of the cutting stylus. The record itself is an engraving of the waveform from which the pickup stylus receives vibratory motion to be converted again into electrical signals, amplified and returned to mechanical and then acoustical energy by the speaker.

If the frequency of a sound is doubled the pitch is raised an octave, and if it is halved, the pitch is lowered an octave. We instinctively recognize the same note in whatever octave it may be heard. Between one note and its octave the ear also recognizes other intervals which form the scale.

In music as in simple sound, the time element is essential. A single note has no musical meaning until it is played in rhythm, compared and contrasted with other notes. This is a tune, or melody. A further development associates notes of various intervals in chords. The strongest combination is the octave. The next, which sounds hard and open to modern ears, is the fifth. Thirds convey a more emotional feeling, and when combined or contrasted with fifths and octaves, they form our conventional system of harmony. These intervals have their basis in nature, as we shall see.

## WAVELENGTHS

Sound passes the ear at a fixed speed—about 1120 feet per second. Since its frequency may be high or low, with many or few alternations of pressure, these alternations or waves have a measurable length which equals the speed divided by the frequency. The **wavelength** of a 100 cycle note is 1120/100 or 11.20 feet. That of a 1000 cycle note is 1120/1000 or 1.120 feet. This is also the approximate length of a column of air which will resonate at that frequency, and consequently double the length of a pipe open at one end, such as flute, required to sound that frequency. A pipe stopped at both ends, such as a stooped organ pipe, will produce a wave about four times its own length.

## AMPLITUDE/VELOCITY

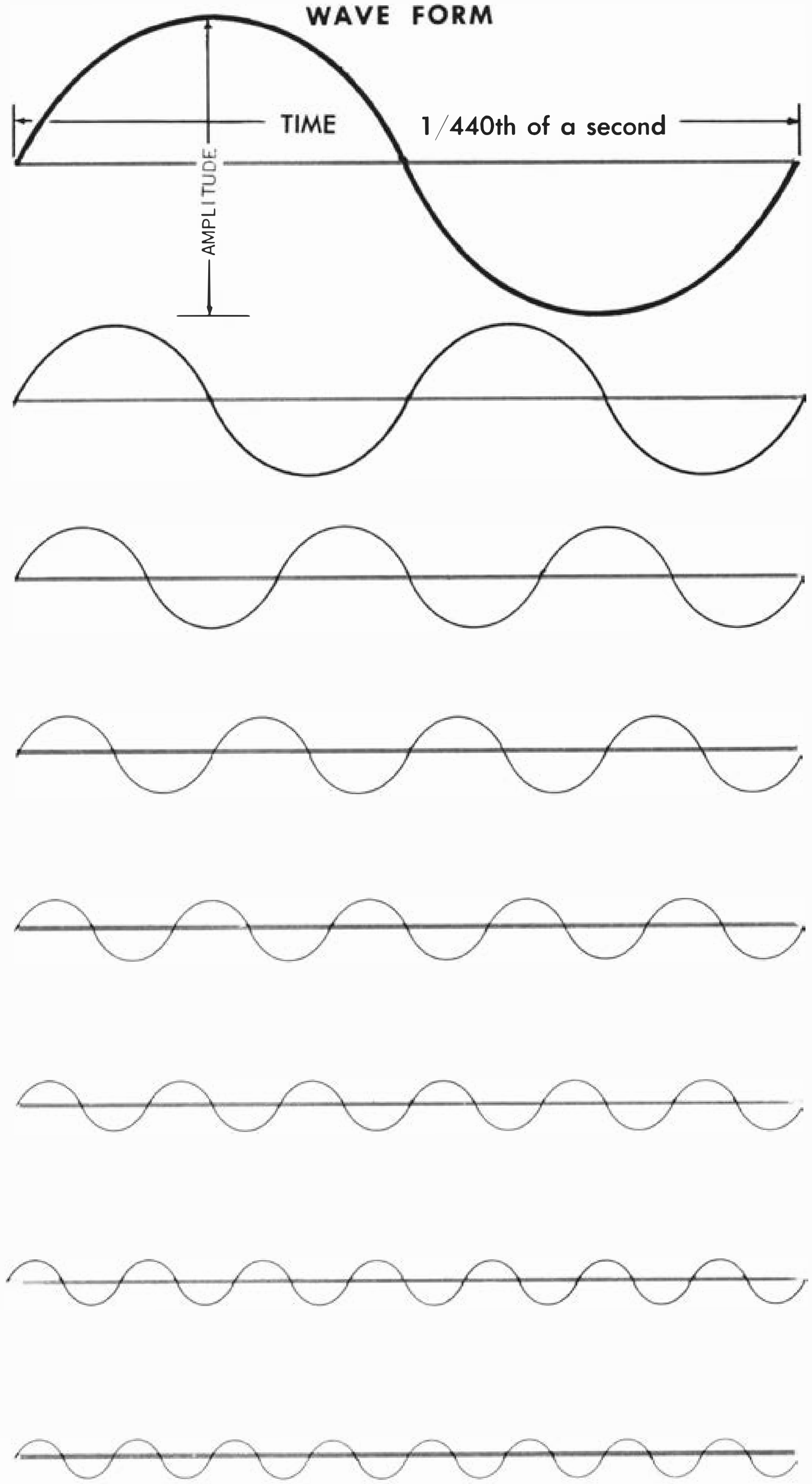
The **amplitude** of a sound wave varies inversely with its frequency assuming, of course, that its sound energy remains constant. The sound energy in turn is directly related to the velocity with which changes in pressure or other related phenomena takes place. **Velocity** must not be confused with the frequency of sound waves or the speed with which they travel. The term refers to the speed with which a loud speaker cone moves back and forth, or a pickup stylus must oscillate in synchronization with the changing air pressure.

If we raise the pitch of a note while retaining the same sound energy the amplitude of the wave decreases. The energy, which is constant, is distributed over more alternations in pressure and each wave or cycle therefore has a smaller **excursion** or departure from its normal position. But the speed with which the changes in pressure take place remains the same. This condition, which is the natural one in sound, is known as a state of 'constant velocity.'

Velocity is indicated in waveforms by the angle at which the curve crosses the 0 axis—its point of greatest speed. The same laws of pressure and velocity apply to the movement of the microphone diaphragm, the vibrating string or reed of a musical instrument, to the recording stylus, the pickup needle and the speaker cone. Their velocity or speed will remain constant at all frequencies when dealing with a signal of constant value.

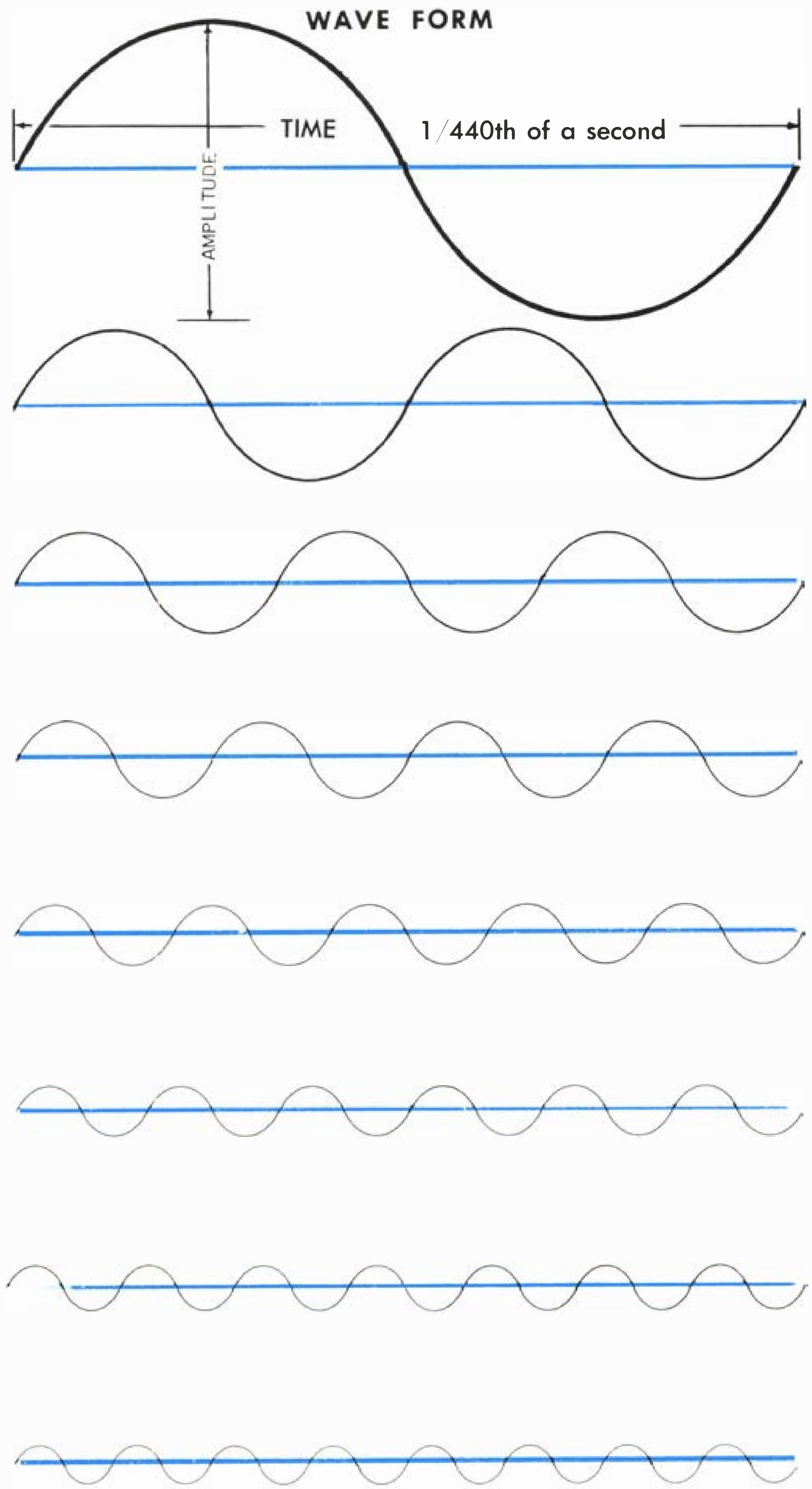
In recording, however, it is not feasible to follow this natural scale of values. The increased amplitude of bass notes would make very broad groove spacing necessary with a serious reduction in record playing time. At the other end of the spectrum, following the same proportions, very high notes would drop to inaudibility or below the record surface noise. The signal is therefore artificially intensified as its frequency increases. That means in plain language that the higher-pitched a note is, the louder it is laid down on the record, other things being equal. Several **recording characteristics** are now in use as we have seen, and all are constant amplitude in principle, slightly modified. The maximum volume of the lowest frequencies is the limiting factor and with that amplitude established, the entire spectrum is made uniform with it. In this way the high frequencies, which would normally drop

# HARMONIC SERIES OF




| | MULTIPLES OF FUNDAMENTAL |
|---|---|
| FUNDAMENTAL FIRST HARMONIC<br>One cycle 440 cycles per second | **FX1=440 cps** |
| OCTAVE — SECOND HARMONIC<br>Two cycles 880 cycles per sec. | **FX2=880 cps** |
| THIRD HARMONIC — TWELFTH<br>Three cycles 1320 cycles per sec. | **FX3=1320 cps** |
| FOURTH HARMONIC — FIFTEENTH<br>Four cycles 1760 cycles per sec. | **FX4=1720 cps** |
| FIFTH HARMONIC — SEVENTEENTH<br>Five cycles 2200 cycles per sec. | **FX5=2200 cps** |
| SIXTH HARMONIC — NINETEENTH<br>Six cycles 2640 cycles per sec. | **FX6=2640 cps** |
| SEVENTH HARMONIC — 21st FLAT<br>Seven cycles 3080 cycles per sec. | **FX7=3080 cps** |
| EIGHTH HARMONIC — 22nd<br>Eight cycles 3520 cycles per sec. | **FX8=3520 cps** |

# HARMONIC SERIES OF




| | MULTIPLES OF FUNDAMENTAL |
|---|---|
| FUNDAMENTAL FIRST HARMONIC<br>One cycle 440 cycles per second | **FX1=440 cps** |
| OCTAVE — SECOND HARMONIC<br>Two cycles 880 cycles per sec. | **FX2=880 cps** |
| THIRD HARMONIC — TWELFTH<br>Three cycles 1320 cycles per sec. | **FX3=1320 cps** |
| FOURTH HARMONIC — FIFTEENTH<br>Four cycles 1760 cycles per sec. | **FX4=1720 cps** |
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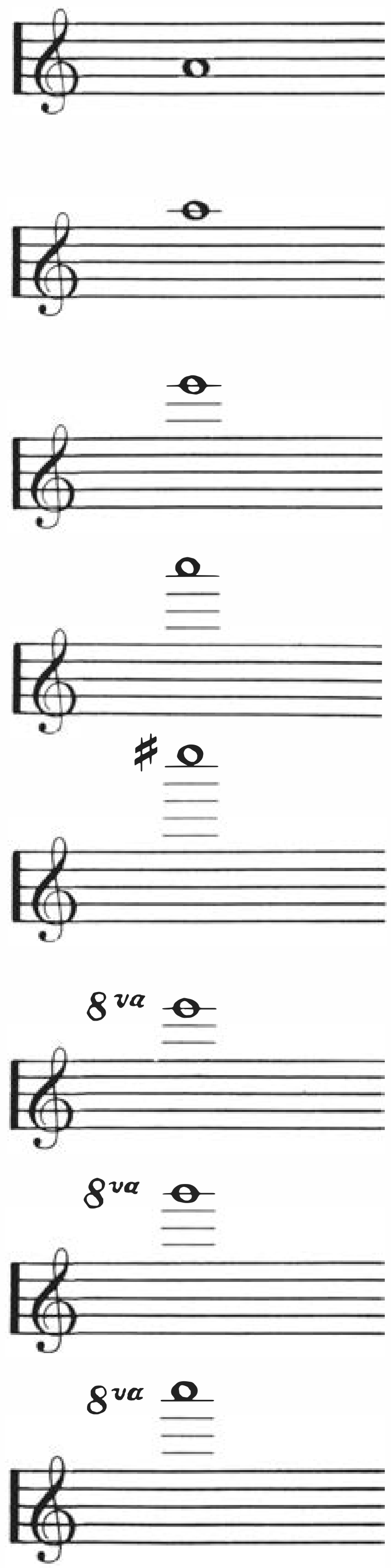
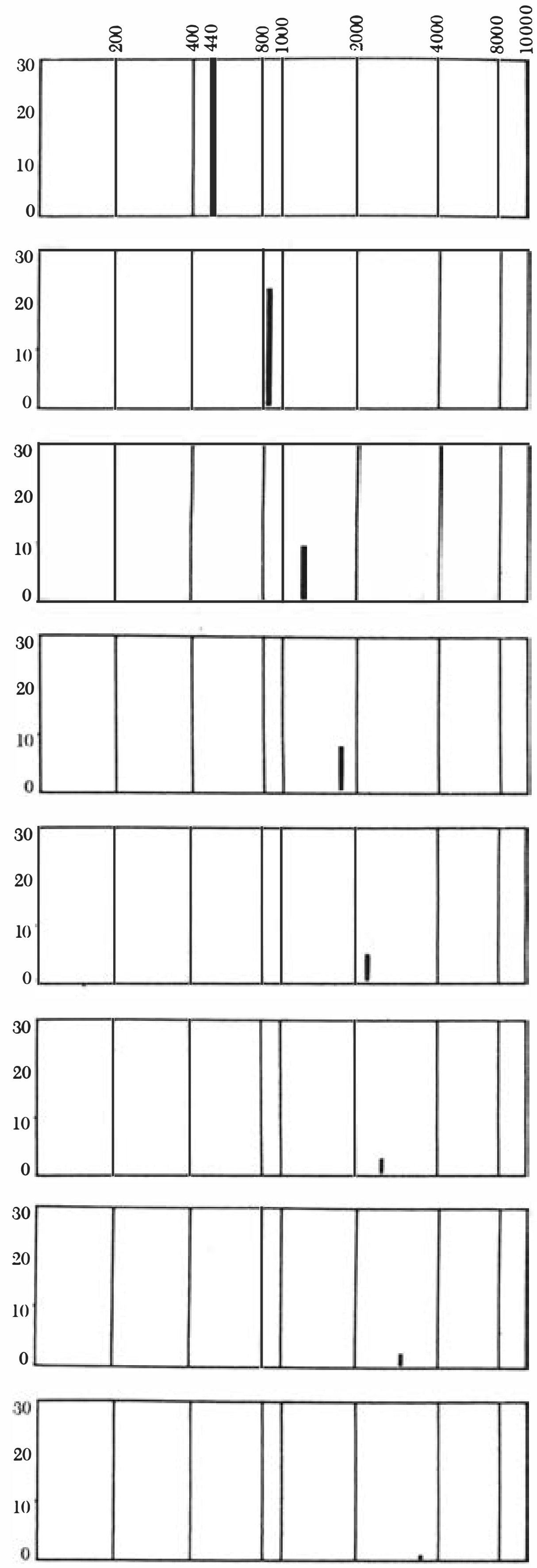
| SIXTH HARMONIC — NINETEENTH<br>Six cycles 2640 cycles per sec. | **FX6=2640 cps** |
| SEVENTH HARMONIC — 21st FLAT<br>Seven cycles 3080 cycles per sec. | **FX7=3080 cps** |
| EIGHTH HARMONIC — 22nd<br>Eight cycles 3520 cycles per sec. | **FX8=3520 cps** |

# 440 CYCLE NOTE

| MUSICAL INTERVAL | MUSICAL NOTATION | |
|---|---|---|
| UNISON | | A |
| OCTAVE | | A |
| TWELFTH | | E |
| FIFTEENTH | | A |
| SEVENTEENTH | | C♯ |
| NINETEENTH | | E |
| FLAT TWENTY-FIRST | | G |
| TWENTY-SECOND | | A |

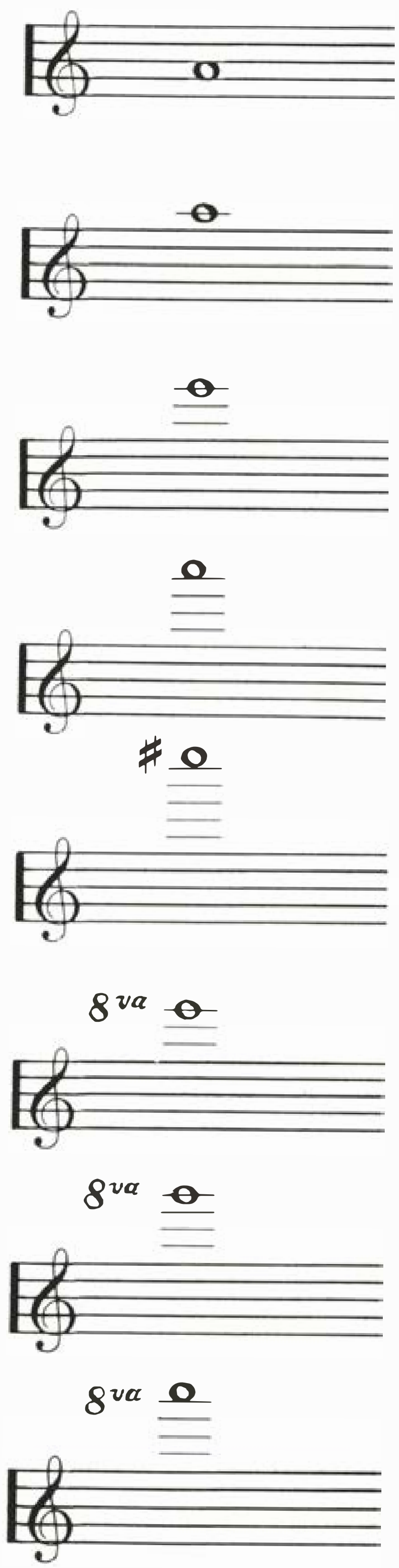



ACOUSTIC SPECTRUM

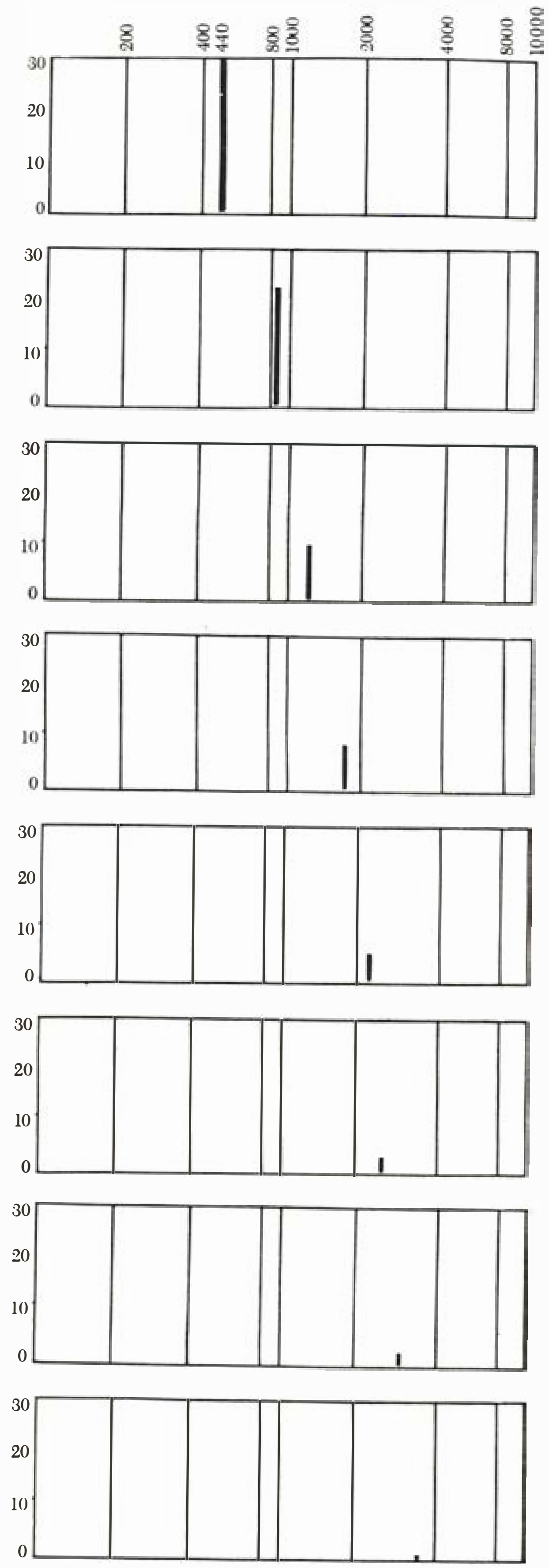


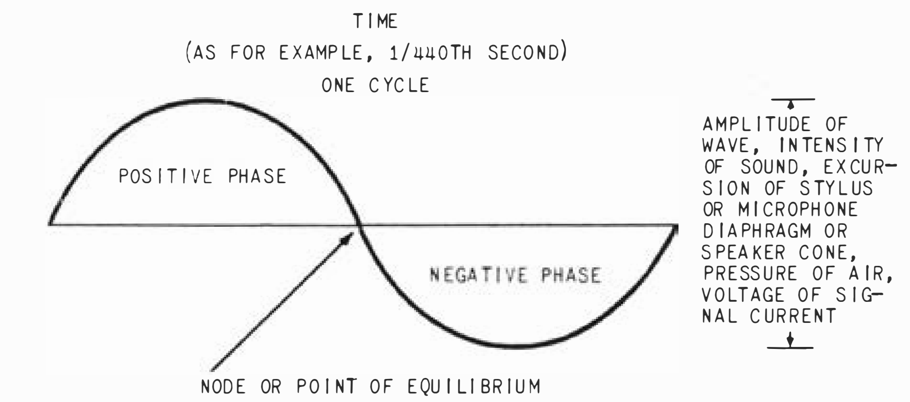
# 440 CYCLE NOTE

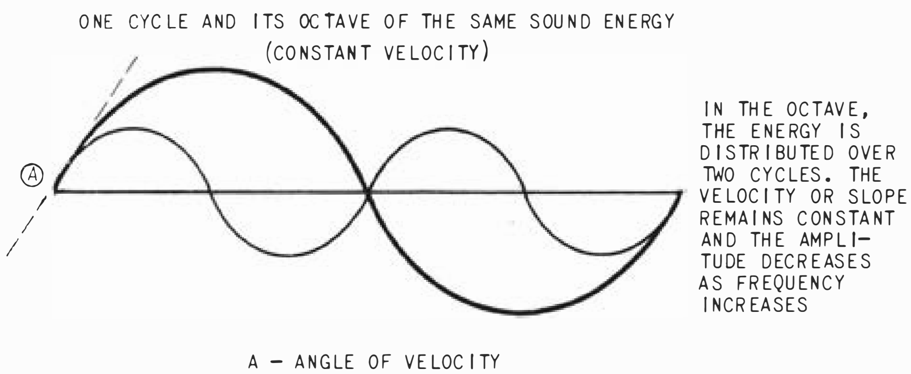
| MUSICAL INTERVAL | MUSICAL NOTATION |
|---|---|
| UNISON | A |
| OCTAVE | A |
| TWELFTH | E |
| FIFTEENTH | A |
| SEVENTEENTH | C# |
| NINETEENTH | E |
| FLAT TWENTY-FIRST | G |
| TWENTY-SECOND | A |

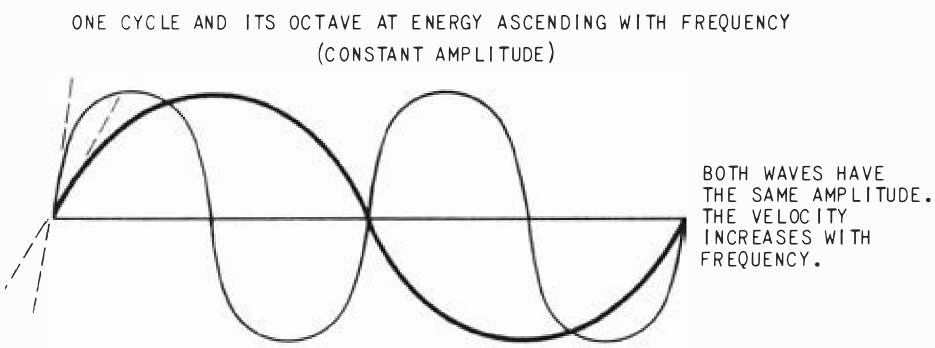



ACOUSTIC SPECTRUM

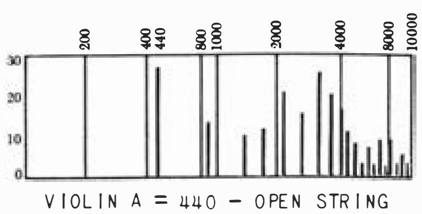


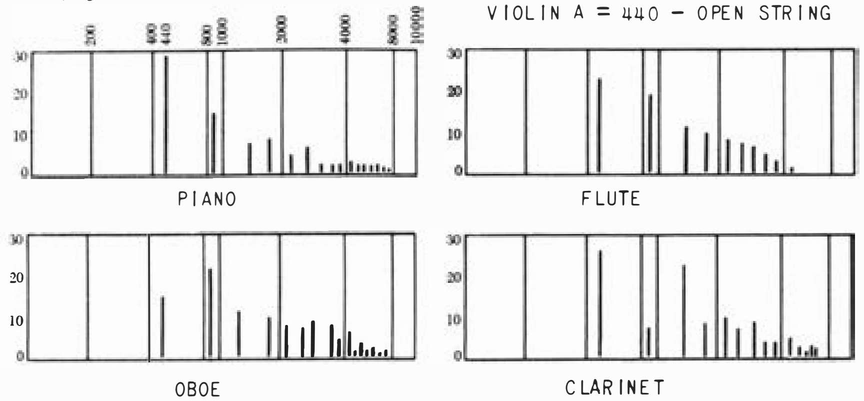



Acoustic spectra for five characteristic musical instruments playing the note A (440 cycles per second) showing the relative intensities of the fundamental, which is the lowest harmonic, and the upper partials. (After Olson) See pages 15-16.




down to insignificance, are brought up to the same strength as the bass. In correcting this, the playback equalizer attenuates the high frequencies and with them the surface noise which exists in that frequency region.

The velocity therefore increases with frequency in a **constant amplitude** system. When the **velocity** is **constant,** the amplitude is inversely proportional to the frequency. The terms constant amplitude and constant velocity must be understood to apply to signals of a given value. The louder a note is, the greater will be its amplitude and its velocity in either system. But if we keep its loudness and move up or down the scale, it will follow a pattern of either constant velocity or constant amplitude as the case may be.

78 rpm records were recorded with a constant velocity characteristic above a turnover frequency. Below that—200 to 800 cycles—the pattern was of constant amplitude. That is why the equalization necessary for 78 rpm records is radically different from that needed for fine-groove 33 rpm recordings.

The direct correspondence between sound level, velocity and angle of the recorded waveform make it possible to judge the recorded level of a record by the width of the light pattern which it reflects. This is most useful in checking the performance of cutter heads. First a 1000 cycle note is recorded at reference level, then again 2 db higher, and 2 db lower. The other frequencies are then recorded at the original reference level. The pattern should then be illuminated by a single light about 8′ distant and as nearly as possible in the same plane as the record surface. When viewed from above the width of each frequency band will correspond directly with the level which has been recorded. If the recording is flat, the bands will be of uniform width. If some frequencies are higher or lower in level, their light patterns will be wider or narrower, and by checking their deviation against that of the two variant 1000 cycle bands it will be possible to estimate their deviation. Examine the test frequencies given on the accompanying record and you can see what the recording characteristic does to the signal. (B-I)

# COMPLEX TONES

Vibrating bodies—strings, reeds, metal bars or columns of air—tend to vibrate both as a whole and also in sections. As they subvibrate in half they generate a frequency double that of the fundamental. As they subvibrate in thirds they produce a sound three times the fundamental frequency, and so on through many multiplies of the fundamental frequency which gives the note its pitch. The ear hears all of these as one sound, and not as several separate notes. Together, they are known as the **harmonic series** and are composed of the **fundamental** and the **upper partials** or **overtones**. It is the number, prominence and phase relationships of these harmonics which give distinctive character to various sounds.

The fundamental contributes body and carrying power to the sound while its vividness, brilliance and richness are the result of its upper partials. As a rule the richer a sound is, the better it will blend with other sounds. The piano and the harpsichord are signal antitheses of substantial fundamental body tone with little harmonic development, and of rich harmonic development with little or no fundamental component. The latter qualities make it possible to multiply harpsichords with increased brilliance and color and with no aural distress, as Bach demonstrated in his concertos for several harpsichords.

Among the purest sounds are those of the tuning fork, the upper register of the piano and flute. Strings, muted brass and organ reed stops are extremely rich. The upper partials extend far beyond the compass of the score, and it is they which make a clarinet, a violin and an oboe sound different from one another. It is also why these instruments sound alike when reproduced by a system of limited frequency range. The first eight harmonics are identified by the table, and others follow at increasingly narrow intervals. As many as fifty components have been discovered in a single musical note. Our modern tempered scale departs slightly from the natural scale, in which the harmonics are intervals so as to divide an octave into twelve equal tones. F♯ becomes the same as G♭, greatly simplifying transposition, notation and instrument manufacture. But those who play the instruments of continuously variable pitch, i.e., the strings, often follow the natural intervals instinctively.

Our waveform is becoming more complex. From the simple sine wave of a pure note it assumes squarish or saw-toothed or triangular outlines according to its harmonic components. The richer the tone is, the more nearly its wave will approach a saw-tooth form. The brasses have especially sharp peaks. Waves which have the odd numbered harmonics alone or predominantly, as the clarinet and stopped organ pipes, have outlines which are more square.

A saw-tooth wave can be generated electrically, and contains the complete range of musical sounds at its frequency. By filtering out the unwanted components, it can be reduced to the contours of any desired tonal pattern. Some electronic musical instruments produce tone in this way. It would obviously be impossible to translate an absolutely square or saw-tooth wave into acoustical or mechanical terms for the beginning and end of each cycle involve an instantaneous change of values without any time interval whatever. That would mean that a loud speaker cone would move from point to point without taking any time to do it, or that a recording or pickup stylus would make a right-angle turn!

When we add to the complex sound wave of a single note all the others of an orchestral score, some at intervals which comprise chords and others almost at unison but always slightly detuned, the waveform becomes more intricate. Another factor enters as the sound is reflected from walls, ceiling and floor, either combining or conflicting with the directly projected sound. There is the fundamental of each note of each instrument with many upper partials, all interacting with all the other sounds, reflected, recombining with the original and other reflected sounds. The staggering complexity of the total product results from the great number of components, all simple in themselves, reinforcing or opposing each other at various intervals. But the result is still a single wave form, expressed in two factors of time and amplitude. It can be made visible by an oscilloscope and scientifically analyzed by classifying its repeated patterns.

## DYNAMICS AND TIMBRE

As an orchestra score calls for increasing loudness and intensity of sound it will add new voices not only in unison with the skeletal harmony, but also at intervals which augment the entire tonal fabric. To the strings are added the wood-winds and brass, playing at higher and lower pitches, and the percussions. This relationship between intensity and harmonic richness is a widespread principle in nature. Single instruments become more brilliant when played loudly than softly. In this respect, brasses have the greatest, and the piano the least, scope of coloration.

As a result of this increasing harmonic complexity, the waveform will become more intricate without reaching unmanageable amplitudes. The increased power is distributed across the time dimension, rather than concentrated within a narrow compass. Sounds in which a narrow concentration does occur, such as loud piano, horn or tympani passages, present the most difficult recording and reproducing problems.

## ACOUSTIC SPECTRA

While the waveform can be made visible and analyzed, another form of graphic description is more clear and useful for many purposes. It plots the components of a sound according to their relative strength and frequency. The acoustic spectrum therefore reveals the exact position and value of the harmonic components at a glance. It does not indicate the phase relationships, but these seldom affect the tone greatly although they may modify the waveform.

The timbre of any instrument will vary with the register in which it is played. Sometimes the fundamental frequency is weaker than the upper partials. That is the case with the piano and bowed instruments when these instruments play in a low register. The lowest G on the violin sounds the second to tenth harmonics all stronger than the G itself. The predominant partial of its lowest E played by a bass is the twelfth, or B in the next octave above. Second in strength is octave E, then the fifteenth, or E two octaves above, the seventeenth, which is G# above that, and only then the fundamental frequency itself, followed by the higher partials.

Nasal quality usually means a well developed third harmonic or twelfth note. Symmetrical brilliance of sound comes from the octaves—second, fourth, eighth harmonics. The snap and crack of brass comes from the strength of its dissonant partials above the seventh. High thirds and fifths, such as the third and fifth partials, give piquancy.

What the musician sees as a note on a staff, the scientist now sees as a waveform or as a chart of the various component frequencies in their proper relationships. It is possible, with repeated listening and study of the spectra, to recognize the presence of the partials which produce certain tone colors.

## ACOUSTICS

(B-IV) When sound energy impinges upon a surface it is either reflected, absorbed and/or transmitted. It can be reflected either as an echo—which is undesirable; or as reverberation—which is an essential component of good musical sound. An echo is a reflected sound which is recognizable apart from the original, and which follows at an interval of more than 1/15th of a second. When reflections follow at a shorter interval the ear does not distinguish between them and the original, and all are heard as a single, persistent sound.

The percentage of reflected to total sound in a studio of agreeable acoustics is far higher than one might suppose, and in even an average room it is around ninety to about ten per cent of directly radiated sound.

Insufficient reverberation leaves music parched and harsh, while too much will muddle its melodic line and changing harmonies. The reverberant properties of almost any room will vary with frequency. That is, a studio may reflect more high frequency sound than it does low frequency sound, or vice versa. Normally, reverberation time is expressed for 1000 cycles per second. At very low frequencies it should be twice as long, and at very high frequencies (10000 up) it should be one and one-half times as long as at 1000. This means that the middle frequencies will decay most rapidly and the extremes last, leaving the aural rate of decay symmetrical. An unbalanced reverberation pattern such as one with excessive high frequency persistence can be very unpleasant, especially when it is exaggerated by a distant microphone pickup.

The optimum reverberation period varies, of course, with the music, the instrumentation, the size of the playing group, the size of the studio, and with taste. Solo piano, next to speech, takes the least, then in ascending order string quartets, large chamber groups, small concert orchestras, dance bands, symphony orchestras, with organ usually taking maximum reverberation—up to several seconds. Large orchestras need speaking room as well as acoustico-esthetic framework. A small group would be lost in as much space. Wood panelling favors the qualities of string instruments, especially for chamber music, while the ideal framework for a brassy, percussive score such as Tchaikovsky's *1812* would have harder surfaces and a longer reverberation period. The decay of the sound should always be smooth, without abrupt changes.

Synthetic reverberation can easily be added to existing recordings by several devices. The simplest is the reverberation chamber which, as its name indicates, is a room lined with hard surfaced material into which a loud speaker delivers the sound which is reverberated and picked up by a microphone. One company uses its stair well for this purpose and many a record surfeited with critical encomiums emerged dry as a bone from an austere concert hall, to be glamourized later on the back stairs!

Another device is a tape recorder with several pickup heads spaced out from the recording head and so wired that the program is first recorded and then picked up at intervals as the tape travels from the recording to the pickup heads. The highest fidelity is not essential to this synthetic reverberation, and so the components are not critical. The tape is a closed loop, passing from the last pickup to the eraser and returning to the recording circuit.

Reverberation period and character can thus be controlled. Another mechanical reverberator consists of a voice coil like that of a loud speaker, driving one end of a coil spring damped in oil, along the length of which are mounted mechanical pickups similar to those used in phonographs. It operates on the same principle as the tape reverberator. Such devices are useful for the improvement of poor program material but they are always substitutes. Unfortunately, they are seldom equipped with enough repeat stations and it is possible on some staccato sounds to detect the individual signals. While reverberation continues beyond the intervals when the ear would distinguish a repeated signal (1/15th second) it consists less of many reflections following each other than of a continuous wave of sound in which there are (or should be) no actual gaps. This will be clear if one thinks of sound emerging from a source not in single lines but as a continuous spherical wave which meets the reflecting walls continuously along their surfaces. As all these surfaces are continuous and as the sound meets the walls at all angles and distances from its source, the reflection is similarly uninterrupted and continuous.

The great proportion of the reflected component in what we hear would seem to leave little to be said for the value of dinaural sound reproduction. Except in chamber music of modest dimensions played in small quarters, we do not normally hear each instrument in its place. We hear a totality of sound—original and reflected from all sides and at all angles, leaving us with only a sense of general direction and little idea of precise location except that which is supplied visually.

# 3

# HOW RECORDS ARE MADE

A GOOD record begins with a good studio. Studios which are too dull or too reverberant, or have unpleasant resonant peaks, must be tamed unless the recorded sound is to have the same defects. This may mean curtaining off certain areas where hard surfaces will return the sound at undesirable intervals. Or it may mean adding reflective screens to strengthen and brighten the tone.

A rule of thumb procedure has developed whereby studios are built with a 'live' and a 'dead' end, the first to focus and project the sound into the open space, the second to sponge it up before it can echo. Accordingly, many studios have been designed on the principal that no more sound reflection should be tolerated than will direct and reinforce the original sound.

If a studio is deadened in the interests of excessive definition the effect is very much like a piano without a soundboard. One could hear it, but it would lose tremendously in sonority and volume, and even more in tone quality. This state of affairs was illustrated by a scandalously bad and shamelessly publicized radio studio from which emanated with unprecedented razzle-dazzle the coarsest and hardest sounds ever generated by well-intentioned musicians.

A studio, then, may be good or bad. It may have good spots and bad spots. If it is strange to the recordist, he will sound it out by walking around, speaking or clapping and listening for reverberation and echoes. Between a source of sound and a close reflective wall, a weird acoustic condition can develop whereby the direct sound meets the reflected sound with periods of reinforcement and cancellation which are known as standing waves. These can be avoided by changing the positions of the players in relation to the wall.

The instrumentation and the nature of the music determine the studio set up. In general, the smaller the group of instruments, the closer, more intimate, must be the microphone pickup. The more 'presence,' the less reverberant, and the more vivid will be the tone colors. The vividness and definition of any recorded instrument decreases as it is a) distant from the microphone, b) played softly and c) played rapidly. To compensate for this, instruments of elusive tone such as the oboe are sometimes given special microphones to augment what the main microphone receives.

The orchestra may be sectionalized and picked up on several microphones, both to control the balance and to obtain great impact and presence. The cost is perspective. One does not hear the instruments that way in a concert hall, unless he is sitting in the midst of the orchestra. The forceful impact and presence of which the microphone is capable has made some people wonder whether concert hall sound is a paragon or only a necessity. Whatever the answer, the problem resolves itself into getting the microphones far enough away for good sound while preserving whatever presence one believes desirable together with the tenuous voices which can easily be lost in large spaces. The microphone always magnifies distances and the optimum microphone distance is always closer than a corresponding listening distance. Sometimes a secondary main microphone is set up at a more remote distance and its signal mixed with the primary channel to augment its reverberation.

The large romantic orchestra such as that of Berlioz and Wagner is probably the least efficient recording entity, adding more and denser sounds to a murky tonal composition as a vast army of players blow and saw and drum at each others' backs, the sound emerging fat and asthmatic from an acre of absorbent clothing. At best it reaches a massive dignity which must be frosted with high percussions and fortissimo brass to brighten its sombre texture. When a chorus is added, this condition becomes even worse. More than a few records which essayed such ambitious tonal feats would have been greatly improved by sending half of the musicians home.

In comparison with such dubious heroics, the baroque ensemble shines with a luminosity which touches of cymbals add, but do not impart, to romantic scores. If the problem of recording a large orchestra lies in avoiding the opposite dangers of letting it get sticky on the one hand, or degenerate into a distant chatter on the other, the problem of recording chamber music is to confine the sound without stultifying it. In too large a space strings evaporate in sort of a floating whine, due in part to their own strongly developed third harmonic (twelfth) and partly to their relatively delicate sonal texture. (There is a reason why an orchestra has so many strings as compared with its winds.) It is therefore good practice to equip small string and chamber music groups with contributory surfaces and well proportioned, ample, but not excessive, studio areas as in the Mozart Quartet K 80 (A-II).

The voice is losing its terrors for recording men although sopranos can lay waste to most microphones unless they are backed off on peaks or turn their heads. This can often be heard on records. The piano with the tremendous initial impact of its percussive tone and its transient character is almost the supreme test of microphone endurance and adaptability. But the day of the recorded piano which sounds like a marimba, a celesta or a banjo is happily over. Indeed it is now possible to recognize the maker's name by such sound as we hear in the Liszt Second Concerto. (A-III). This was one of the last advances to come in the art.

Microphones vary in the same respects as other audio equipment: frequency response, transient response, power capacity and also in their polar response or directionality. Some are adjustable for uniform pickup in all directions or for sensitivity on one side only, or on two sides. Some have provisions for a low frequency cut off to filter the unpleasant boom which the singing or speaking voice often causes. Like lenses in photography, they have their applications, their strong and weak points. Several different ones may be used for large orchestras and groups and even for dance bands for high definition of individual instruments.

The live performance and the recorded recreation of it have different perspectives on at least two counts. One is the visibility which enables those present to supply relative value to sounds—especially similar sounds—which they may not actually possess acoustically. This deception can betray the novice into thinking that a solo string, for example, is sufficiently prominent while accompanied by other strings because he can hear and see and follow the player. Experience has taught recording men the value of closing their eyes from time to time if they are in a control room from which the players are visible. And even this will not prevent the confusion which processing and replaying visit upon individual voices. A little exaggeration of relative values is sometimes desirable. Too much can be fatal!

Another difference in perspective results in decreased reverberation time since the lower threshold of audibility is fixed, and comes quicker in the decay period if the original sound level is lowered. This and the Fletcher-Munson effect act to contract the sound in two dimensions—space and frequency.

In negotiating all these problems the engineer can hope for a studio of the right size and aural properties; an ample assortment of microphones which can be used together or separately as needed; facilities for mixing the microphone inputs, or combining them in various proportions; tone compensators to facilitate tailoring bass-heavy or shrill input channels; and possibly a reverberation mechanism with access from the console. The latter may be useful either for the entire ensemble or special groups—frequently first violins—to recapture the life which may escape in open space—and give them the sheen beloved by a public which prefers metal violin strings to gut.

All these devices afford ample scope for ingenuity and even for artistry. From them have evolved new types of light concert music and popular song and dance productions with their palpitating celestial choirs, their chairside saxophones and other novelties which I shall leave to the mercies of the reader.

Granted a good studio and the best of equipment, the art still has ample challenge for the recording man, as the steadily improving output of records demonstrates year by year. Every recording session is different in some respect from every other. There are always new problems of balance, elusive parts to be captured and brought into relief, difficult groupings of instruments to manage without affecting the tutti, solos which need presence, intense voices to be controlled, timid ones to be encouraged, ensembles which need perspective and blend.

Placement of the instruments will account for much. It will seldom achieve everything with a large orchestra, especially if there are important soloists or a chorus. This would involve several problems which call for divergent solutions and the recordist must take the ensemble apart and cure each ill separately. Listen to the contrasting pickups of Hershy Kay's *Western Symphony* (B-IV). The balanced pickup was done with a main Telefunken condensor microphone 15′ above the floor, 25′ from the first stands, a directional ribbon microphone picking up the first violins which were raised on tiers for special reverberation chamber treatment, a directional Photophone microphone for woodwinds, and another 44 for piano. The other examples were done by very close microphones with exaggerated presence and inadequate perspective.

Studio recording is now invariably done on magnetic tape which has many advantages including durability, ease of editing and adaptability to nervous exegencies of a recording session. The recording machine can, if necessary, run almost continuously thus making one less variable to be synchronized with the many other men and instruments which must act precisely in concert. Mistakes can be immediately corrected after a short pause without going back to the beginning of the record, parts can be announced and directly identified and any number of snips and fragments—each the best available performance—strung together into a composite whole. There is a divergence of opinion on the effect all this has on art. Spontaneity can suffer when a group which is repeatedly starting and stopping and retracing its steps. It may place each note with commendable precision. But the spirit, the soul, the spontaneity of an unfettered performance will often excel in the things which make music effective.

As musicians play and tape flows, the engineer and recording director keep attentive vigil in the control room. They hear the music under similar conditions to those in which records are heard. It is not their job to supervise the musicians although the most successful recordings are always made by musicians who will accept the advice of those who know what happens after the sound leaves the original tape. That will be reheard and checked by many ears. But the little changes of perspective which make a disc recreation just a bit different from that heard on the original tape under laboratory conditions—these are the details of finish which set a fine recording apart from a mere transcript of happenings in the studio.

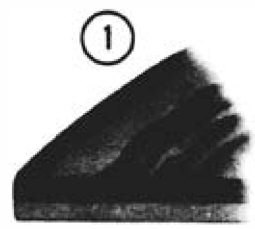
Any reduction in scale will minimize the effect of dimensions and dimensional relationships on an unchanged aural consciousness. The Fletcher-Munson effect is only the most noticeable instance of this. A performer or conductor who can, with the assistance of the recording men, project his interpretation into the new dimensions of the record listener will serve his own objectives better than one who is enthralled by the immediate beauty of his performance. This affects the dynamic range, the instrumentation, and even—less frequently—the tempo. One can play for the moment, for the sound in the studio, or play for the record. And occasionally what might sound best on the latter will sound very peculiar indeed in the former!

The manufacturing process begins with an original lacquer disc (the 'wax' of an earlier day) which is recorded from the tape original. At this point the latter has been edited, corrected for unsatisfactory balance and improved with any other refinements such as additional reverberation. Tape editing is becoming a new craft in the industry and calls for both musical and technical skills of a unique order.

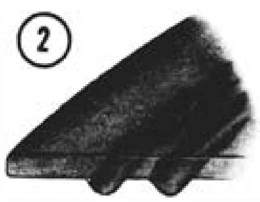
The recording machine has a heavy duty turntable carefully balanced and driven through special transmission which carries a minimum vibration to the disc. The cutting head, unlike the pivoted and swinging pickup arm, is held rigid and moves in a direct line from the edge to the center of the record blank under the power of a motor-driven feed screw. The engraving stylus does not indent, but routs a channel in the disc surface from which the excavated material is drawn off under suction.

When the original lacquer disc is finished it receives a molecular deposit of gold or other metal which will be sufficient to carry current for an electroplating bath. This is strengthened by a heavier, structural deposit of copper and forms a negative copy which can be stripped from the lacquer disc. This is plated in turn and stripped to yield a positive two stages removed from the original, and playable. By such repeated platings and strippings durable metal positive and negative copies are duplicated. The stamper which is used in the record press to produce copies from soft plastic is, of course, a negative metal copy.

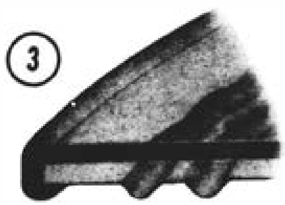
## A BRIEF RESUME OF THE STAGES, ACOUSTICAL, ELECTRICAL AND MECHANICAL, THROUGH WHICH THE SOUND PASSES ON ITS WAY FROM THE MUSICIAN TO THE RECORD LISTENER:



LACQUER ORIGINAL
(POSITIVE)

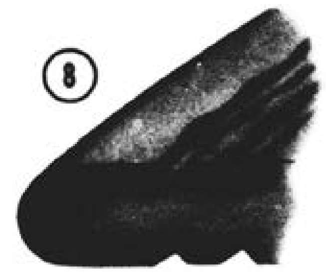


MASTER
ELECTROPLATE COPY
(NEGATIVE)



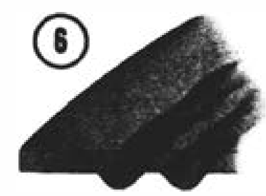
MASTER
(STRENGTHENED)

Original sound
Acoustics of Studio
Microphones
Lines to Control Rooms
Controls and Amplifiers
(Modifications such as artificial reverberation
or equalization)
(Line amplifiers and lines
if recorded at a distance)
Tape Recorders
Tape Editing and Copying
Recording Equalization
(The 'Recording Characteristic')
Recording of Disc
Molecular preplating
(Gold sputtering or graphite dusting)
Heavy plating
Stripping of negative plate
(The 'master')
Plating of master
Stripping of positive from master
(The 'mother')
Plating of mother
Stripping of negative copy from mother
(The 'stamper' which goes into the press)
Commercial plastic copy stamped from above
Pickup stylus, cartridge
(Preamplifier stages for low level magnetic cartridges)
Equalization circuits
Voltage amplifier stages
Power amplifier stages
Loud speakers
Acoustics of audition room

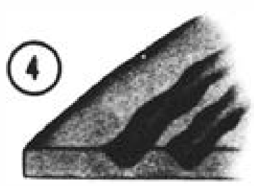


FINISHED RECORD
(POSITIVE)

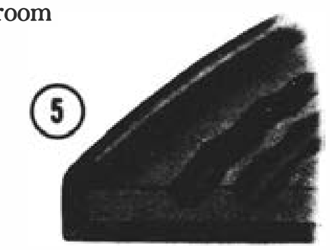
STAMPER
(STRENGTHENED)



STAMPER
ELECTROPLATE COPY
(NEGATIVE)



MOTHER
ELECTROPLATE COPY
(POSITIVE)



MOTHER
(STRENGTHENED)

Each stage of manufacturing has its own contributory operations and quality controls. If we assume that the individual links in this chain perform exactly as intended we still have the following variables: recording characteristic/user's equalization, acoustic editing by the recording company, tone controls in the playback, loudness equalizer. Perfect recreation of the original presupposes the perfect coordination of all these, insofar as that is possible, by the record user.

# 4

# A WORD ABOUT EQUIPMENT

RICH as the rewards and modest as the cost may be, the career of audio experimenter has its man-traps. Many an enthusiast has learned bitterly that an attempt to improve his rig at the most conspicuous and seemingly important point—the loudspeaker—can be an expensive disappointment. He may also read an excellent voltage response from the frequency run and wonder why he can not hear what he sees. Perhaps the same speaker sounded good in the shop but sounds poor at home.

These and other pitfalls cannot be charted precisely, but a few pointers may reduce their hazard.

Do not buy on price. There is no other commodity in which prices range so widely and bear so little direct relation to results. Remember that every link in the chain must be adequate for a job. Neither a pickup nor an amplifier nor a loudspeaker manufacture high or low frequencies. They simply pass or reject what is on the record.

Listen to equipment in conditions as near as possible to your own before deciding to buy. Take ***This is High Fidelity*** with you and hear it on the component you are considering, associated with the other components which you already have. That is—if you are investigating amplifiers, have it played from the same model pickup you now have, and driving the same speaker. That will give you a better comparison with your own norm. Unless you are positive of better results don't buy—or postpone your purchase until you can devise a better test. Perhaps some of the dealer's components are not operating properly. That happens surprisingly often!

Consider each new component as to what it will do now, and as to how it will fit into your plans for progress. Novelty is a big part of the fun, but it can be disturbing to find that one component becomes obsolete when another is replaced. In each component, look for a) permanence of value, b) flexibility and c) adaptability.

A good starter for the modest budget would be an amplifier in the \$50-\$60 class with provisions for several inputs, not more than 2% intermodulation or harmonic distortion at an output of ten watts and a frequency response of 30-15000 cycles $\pm$ 2 db. It should have input jacks for crystal and magnetic pickup cartridges, radio and TV tuner and possibly tape. Output impedances should be at least 4-8-16 ohms. Listen, listen, listen. Don't conclude that an amplifier or anything else is good because the manufacturer talks big. Hear him, but also hear his product as well.

One can start on a very satisfactory single record player with crystal cartridge. The speeds and stylus points will depend upon what will be played. If only 33 rpm records are to be used a single speed and a single point crystal cartridge player is available at around \$20 which will serve well. If 45 and 78 rpm records will also be played provisions must, of course, be made for them, and in the latter case there must be a .003″ point for shellac records as well as the .001″ radius fine groove point. Be sure that the cartridge mounts in the pickup arm by machine screws on the standard ½″ mounting centers. This will allow for replacements later.

While diamond points are both better and ultimately cheaper for anyone who plays more than a very few records, their cost is high and it is questionable whether the investment of their cost in a temporary cartridge is justified. Needle points now only fit the cartridges for which they are made. They cannot be interchanged like the 'needles' of a former day. A good sapphire point will give perfect reproduction when new. It should be changed occasionally as wear quickly progresses undetected, and can ruin vinylite records rapidly. The pickup arm and cartridge should operate satisfactorily with a point pressure of not more than 8 grams.

The turntable should be accurate and constant in speed. The first can be checked by the stroboscope on the label of ***This is High Fidelity.*** Under a fluorescent or argon light its dots should appear to stand still. The light must be from alternating current at 60 cycles, and it may be necessary to exclude other light—daylight or incandescent—while checking the speed. Constant rotation can be tested by listening to one of the frequency bands—3000 cycles—for any variation in pitch. In choosing a professional turntable for use with a magnetic cartridge it is also desirable to check the magnetic isolation between the latter and the motor. With amplifier on, and turned up as high as possible, and the motor on, move the cartridge over the turntable without allowing the point to touch its surface. Any hum which develops shows magnetic leakage at the points were it is detected.

Loud speakers require very careful listening tests for obvious reasons. There are splendid 8 and 10″ and 12″ speakers available for between \$8 and \$25. There are others, much inferior to them, which cost in the hundreds. Size is no more an index of value in speakers than price, and there are several good 8″ models which, in a good baffle, will outperform many larger speakers. There are, obviously, better larger models. But the point is that small speakers are made which surpass the *average* larger one, and they often cost less. A good small speaker is preferable to a mediocre larger one and a single unit is usually better than a poor duplex or triplex.

Tell the dealer how much you are willing to spend, and then ask him to play all speakers within that price without identifying them. Keep track of them as A, B, C. Go from one to another and back again. Some will have better high or low or middle frequency response. They may sound well at low volume levels and break up under power. Others will have a dull leaden sound even though you can clearly hear the things you may listen for.

Try each speaker and every component under all sonic conditions and carefully avoid judging on noisy percussion examples alone. Give soft strings and woodwinds a chance. Listen passively as well as critically. Straining attention will supply missing or deficient elements which a more relaxed ear will miss. Hear piano as well as instruments of sustained tone. Play soft passages loud and listen to the pauses as well as the notes. They can tell much about audio equipment.

A speaker and its baffle are mutually dependent, and neither can fully compensate for deficiencies in the other. A speaker which reaches down to 30 cycles cannot deliver such a frequency to the air without an adequate baffle to prevent an acoustic short circuit of the pressure on one side with the rarefaction of air on the other. The effective baffle is the distance from one side of the cone to the other measured around the box, board or horn which separates the two. It must be ¼ the wavelength of the lowest note to be reproduced. Thus if we wish a baffle adequate down to 50 cycles:

$$\text{the wavelength is } \frac{1120 \text{ (the speed of sound)}}{50 \text{ cycles}} = 22.4'$$

An effective baffle must be ¼ this wavelength, or have a path from front to back of $\frac{22.4}{4} = 5.6'$. In a flat board, this would mean that 2.8′ would be needed around the speaker on all sides, or that a 12″ speaker mounted in a board with its minimum dimension of $1' + (2\text{x}2.8')$ or 6.6′ would be adequate. Since boards 6.6′ square are not usually welcome in homes, the next best thing is to use a wall or to enclose the speaker in a closet, box, or large cabinet to confine the back wave.

Total enclosures are economical and relatively compact. But they are acoustically inefficient like the flat baffle, and they have a pronounced resonance at some point below 150 cycles. A second opening in the front called a 'bass reflex port' will smooth out and lower the bass response somewhat, and give two resonance peaks instead of one. These peaks should be of equal height and that which is contributed by the cabinet resonance can be adjusted by the size of the port to match that of the loud speaker.

The most efficient form of baffle is the exponential horn which best couples the very different acoustical properties of the vibrating cone and the open air. It is available in numerous models most of which are built for corner installation and use the walls of the room to extend the enclosed horn. The cone is usually mounted to speak directly out with the backwave passing through the horn and out at the end of its path.

Building horn baffles is not recommended to an amateur cabinet maker, but total enclosures and bass reflex cabinets are easily assembled from new stock or from old boxes. The largest enclosure is always to be recommended. It should be as solid as possible and well padded inside with sound absorbent material and it should be mounted high so that the music speaks out and does not stifle in a shroud of draperies and carpeting. Since most speakers are strongly directional in their radiation it is advisable to sit directly in front. When this is not possible, or broader radiation patterns are necessary, a few pieces of sheet metal can be placed at an angle over the opening to deflect the high frequencies to the listening area.

A useful test of a speaker's high frequency increment and coverage is to stand very near, but at one side of, or above a speaker opening and to use some hard surfaced object—a shellac record or plate—to reflect the beam from the direct axis of the speaker to one's ear. Use a test track with bright upper frequencies such as cymbals and tambourines. Unless the speaker is equipped with some form of high frequency diffusing cells, new octaves will be apparent when the reflector is held at exactly the right angle. While it is obviously possible to hear the same frequency range by sitting directly in the path of the narrow high frequency beam, a little tinsmithing will make it possible to distribute the component usefully throughout a room.

By these two devices—proper baffling and diffusion of the narrow high frequency beam—it is possible to even exceed with a *good* inexpensive speaker, the results of many indifferent, but expensive installations. Audio distributors have manufacturers' recommended dimensions which should be taken as minimum. At least 10 cubic feet should be provided for a 12″ cone. A closet will, of course, provide far more than that.

**A PROGRESSIVE PROJECT PLAN FOR THE MOST RESTRICTED BUDGET, DESIGNED TO MAKE EACH STEP MOST PRODUCTIVE FOR THE INVESTMENT.**

**1**

**Turntable and pickup as above**

**2** New magnetic pickup cartridge with diamond point. Change amplifier input. Counterbalance arm if necessary to lower point pressure.

**5** Professional turntable which can be mounted in either a new cabinet or a utility box. The old pickup arm and cartridge can be remounted. Observe exactly the same mounting distance between turntable spindle and pivot of pickup arm, and the same height from the surface of the turntable to the vertical mounting pivot of the arm. It is usually necessary to block up the arm since the better turntables are very thick, consequently higher, than inexpensive commercial models. It is well to install a new stylus point at this time.

**8** Professional pickup arm. A new point will be in order now since the mounting center will be farther from the spindle and the stylus will have a different angle than in the shorter arm.

**Amplifier as above**

**3** Preamplifier-control equalizer unit to connect to the high level input of the amplifier. This should be of the best quality to form part of the permanent installation. Or, this step can be combined with the next.

**4** A complete new amplifier including preamplifier, equalization and control functions if step 3 is omitted. This should be the best the owner expects to have, and not an intermediate step to something better. It is preferable to wait for the best rather than to make expensive amplifier replacements. If the preamplifier control unit has been step 3 this step will only include the power amplifier.

**Speaker as above in inexpensive baffle**

**6** A new speaker, either a very good single radiator or the best multiple system contemplated, or such parts of a complete system as will be available for the budget. Perhaps a better single speaker which can be extended by woofer and tweeter later, and handle the middle range only.

**7** A loaded horn or baffle with provision for the speaker as above or to follow.

**9** Replace or add to the all purpose cone a new heavy duty woofer, tweeter and cross-over network.

These stages need not follow in this order but can be varied in any convenient sequence. The order given will prove about as practical and economical as any other. One might wish the bass response which only a horn-loaded cone can give before having a permanent turntable. But strong bass sensitivity might pass too much rumble from an inferior turntable motor —which is why it might be better to buy the motor first. On the other hand it would be quite practical to buy a first-class amplifier at the outset and save the loss involved in replacing a temporary model.

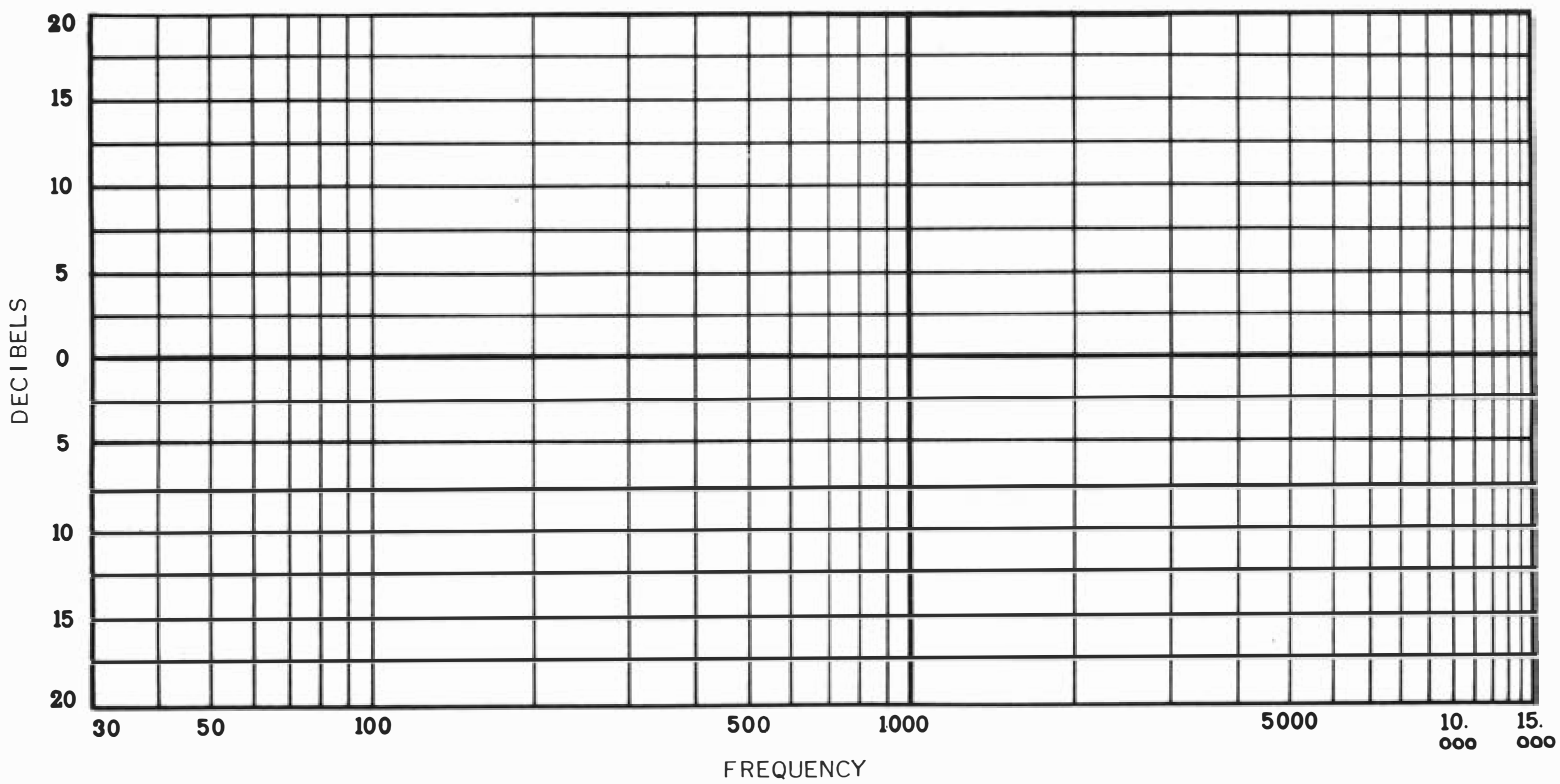
Other accessories such as radio tuners, tape recorders and TV sound can be added as required. But it is well to follow the same rules in regard to the relationship of the parts and permanence of value. Do not duplicate such functions as pickup pre-amplifier and tone controls in a radio tuner which are or will be incorporated in the audio amplifier.

# HOW TO MEASURE THE RESPONSE OF YOUR AMPLIFIER AND PICKUP

Detach the loudspeaker leads from the output terminals of the amplifier and substitute for them a wire wound resistance of the same value, 4, 8 or 16 ohms. Attach to the same terminals an alternating current voltmeter which is calibrated in decibels for output measurements. Set the selector switch to the lowest alternating voltage range—usually to 2.5 volts. Play the test frequency band (B-I) of *This is High Fidelity.* Adjust the volume so that the 1000 cycle note reads 0 decibel on the meter. Then plot the value of each frequency on the logarithmic graph paper supplied for the purpose. Be sure that the record playback equalizer (if this is variable) is adjusted for the RIAA curve. Do not change any setting after beginning the test. Connect the dots entered at each frequency and you will have a response curve for the pickup, record equalizer and amplifier. The same test can be made with the speaker connected but there will be a slight discrepancy due to the changing impedances of the voice coil at different frequencies. If this is done it is well to remove the speaker from the cabinet, or so open the cabinet that there will be no resonances.

# HOW TO TUNE THE BASS REFLEX PORT

Leave one lead from the speaker attached to the amplifier but insert between the other lead and the other terminal a 100 ohm wire would resistor. Connect meter to read output of amplifier beyond the resistor. Put the frequency track on the machine and read the results on the meter as above. Two peaks will be detected below 150 cycles. They should be as near as possible to the same height. If unequal, partially close the port and take another reading. Experiment with different port openings until the peaks are most nearly the same height. Maximum effect from a bass reflex cabinet is obtained when the peaks are separated as much as possible and when they are of equal amplitude.



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BACH: Three Harpsichord Concerto No. 2 in C major (With Concerto No. 1 & Concerto for Four Harpsichords in A minor) — Helma Elsner — Renate Noll — Franzpeter Goebels, harpsichords — Pro Musica Orchestra, Stutgart — Rolf Reinhardt, conductor ...............................................PL 8670

**BAND 4:**

TCHAIKOVSKY: Violin Concerto in D major Op. 35 (With Mendelssohn: Violin Concerto in E minor Op. 64) — Ivry Gitlis, violin — Pro Musica Orchestra, Vienna — Heinrich Hollreiser & Hans Swarowsky, conductors.......................PL 8840

BEETHOVEN: Piano Concerto No. 1 in C major Op. 15 (With Rondo in B flat major Op. Posth.) — Friedrich Wührer, piano — Vienna Pro Musica Orchestra — Hans Swarowsky, conductor ................................................PL 8400

BEETHOVEN: Symphony No. 8 in F major Op. 93 (With Symphony No. 4) — Vienna State Philharmonia — Jonel Perlea, conductor.............................PL 8740

MOZART: Symphony No. 25 in G minor K. 183 (With Symphonies No. 29 & 33) — Vienna State Philharmonia — Jonel Perlea, conductor........................PL 8750

BACH: Brandenberg Concerto No. 4 in G major (With other five Brandenberg Concerti) — Vienna Chamber Group — Jascha Horenstein, conductor..............DL 122

R. STRAUSS: Don Juan (With Death and Transfiguration and Till Eulenspiegel's Merry Pranks) — Bamberg Symphony — Jascha Horenstein, conductor..........PL 9060

KAY: Western Symphony (With Virgil Thomson: Filling Station) N. Y. City Ballet Orchestra — Leon Barzin, conductor.......................................PL 9050

**The scales on Side A Band 1 and the Orchestral chords on Side B Band 2 are played by the N. Y. City Ballet Orchestra, Leon Barzin, conductor.**