BERNARDS

AUDIO ENTHUSIASTS HANDBOOK

BY

B. B. BABANI

NO.214

85p

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BERNARDS [publishers] LTD

The Grampians
Shepherd's Bush Road
London W6 7NF
England

Although every care is taken with the preparation of this book the publishers will not be responsible for any errors that might occur.

The compiler of this book wishes to express his most sincere thanks to "ELECTRONICS AUSTRALIA" the leading radio, TV and electronics magazine published in that continent, and to Sungravure Syndication, their agents, for permission to use some of the material in this book which appeared originally as articles in that magazine.

I. S. B. N. 0 900162 56 2

First Published December 1975

Printed & Manufactured in Great Britain by
C. Nicholls & Co. Ltd, Manchester M11 4AU

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THE RECORD/PLAYBACK CURVE

On the cover of some records, appears the instruction "For proper reproduction, set equalisation to RIAA"—or something to that effect. Occasionally, there is reference to other types of equalisation; often no reference at all.

To answer the question, we don't have to break any new ground because it was very much to the fore in the immediate postwar years. The fact that so little is heard of it nowadays is really a healthy sign, because it indicates the degree to which a very confused position has been rationalised.

Terms like "equalisation" and "recording characteristic" involve the very fundamentals of lateral disc recording and that's where we must start.

Without labouring the elementary aspect too much, most readers of these columns will know that the stylus which cuts the original groove in an ordinary lateral mono disc is made to oscillate from side to side by the audio signal which it is recording. The groove is therefore not a simple spiral but deviates from side to side of the mean path.

The number of deviations or waveforms along a given length of groove is related to the signal frequency involved, a high frequency producing a lot of closely packed waveforms, a lower frequency fewer and more elongated waveforms.

The amplitude of the waveform depends to a large extent on the amplitude of the original signal, loud signals causing the stylus to oscillate with increased amplitude and therefore to produce a more devious kind of groove.

However, there is also a basic relationship between amplitude and frequency and this is the effect which ultimately makes it necessary to consider a "recording characteristic".

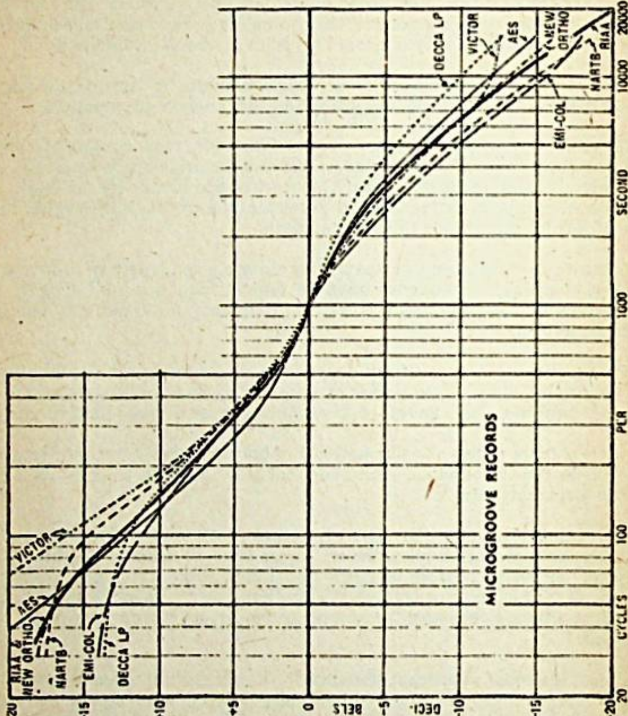
In an ordinary magnetically driven transducer, of which a cutter and loudspeaker are typical examples, the stylus (or cone, in the case of a loudspeaker) moves away from its initial position at a velocity largely governed by the driving current or voltage. Within limits set by its mechanical design, it will continue so to move for as long as the driving current or voltage is applied.

This last is a most significant sentence for it means that, when a stylus (or cone) is responding to a low frequency signal, it will have a greater time, during each half-cycle, in which to move in a given direction, than for a higher frequency signal—even assuming the same nominal voltage or current.

This is exactly the reason why a loudspeaker cone tends to vibrate over a much wider physical amplitude with a low frequency signal than for a high frequency signal of comparable apparent loudness.

And it is also true of a recording stylus. It will produce much greater deviations in a groove for low frequency signals than for high frequency signals of similar original loudness or similar amplitude in terms of driving voltage.

This produces an immediate problem in a laterally recorded disc, because wide deviations in adjacent grooves may either cause the grooves to run into one another or else force recording engineers to make the mean spiral



The audio replay characteristics for a variety of early LP recording standards compared with the present standard RIAA curve, drawn solid. Most modern amplifiers incorporate RIAA compensation only, though the other curves can be adapted closely to it by slight readjustment of the bass and treble controls.

pitch so coarse as to reduce very seriously the number of grooves and, therefore, the playing time.

While probably appreciated in the earliest days of lateral disc recording, it became a particular problem when electrical recording techniques made it possible to capture a much wider range of frequencies.

To cut a long story short, there was born the idea of deliberately restricting the natural build-up in groove amplitude in the lower register by suitably designing the cutter suspension and/or by "doctoring" the response of the recording amplifier system. The aim was to preserve enough bass for reasonable reproduction but so to limit the groove deviation that a satisfactory playing time could also be realised.

FREQUENCY LAW

This amounted to limiting the natural increase in groove deviation below a certain frequency or, to use more technical terms, to recording below this frequency to a substantially "constant amplitude" characteristic. Above the selected frequency, the groove would follow its natural pattern of deviation based on "constant velocity".

Thus, quite early, there emerged a convention of selecting a so-called "turnover" frequency and recording, below it, to a "constant amplitude" characteristic and, above it, to a "constant velocity" characteristic. Later, the "constant velocity" region of the characteristic was subjected to further manipulation by some record manufacturers.

As one might readily guess, the practice of modifying the natural recording characteristic to meet other requirements is the basis for the term "recording characteristic".

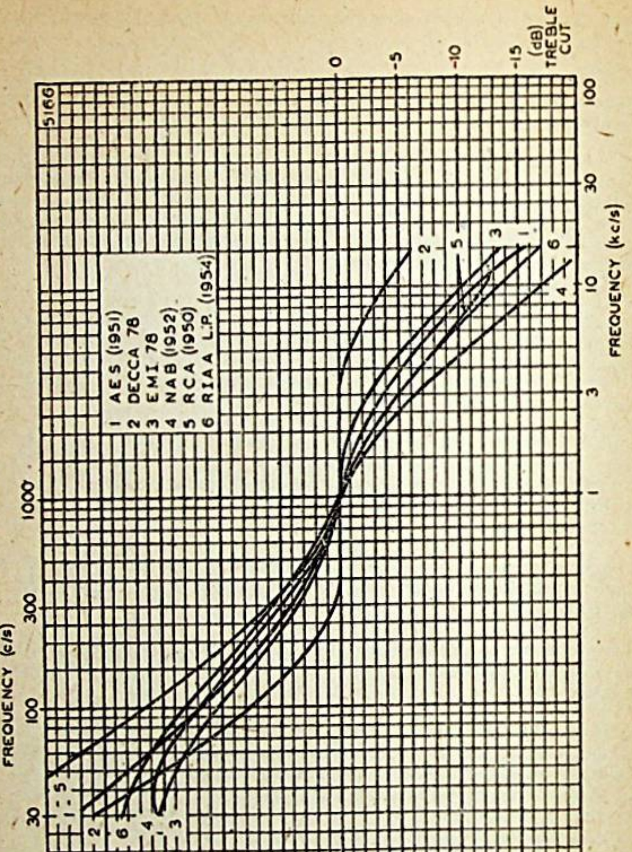
It is interesting to note in passing that for somewhat electrically equivalent reasons, certain recording characteristics have had to be adopted also for magnetic tape.

On playback, the recording characteristic adopted for the disc quite naturally has a vital effect on the balance of the electrical signal produced by the pickup. Assuming the once traditional magnetic pickup, frequencies above the turnover region were reproduced normally, while those below the turnover point commonly suffered progressive attenuation. In fact, some form of bass boosting was desirable to optimum.

The position was complicated by the fact that, while all recording companies acknowledged the reasons and need for some kind of recording characteristic, there were wide differences when it came to the details.

In England, the turnover region most favoured was between 250 and 500cps but, in America there was common preference for a somewhat higher figure.

Above 1,000cps most discs from the E.M.I. organisation (to use a current name) were recorded "flat", that is to the natural "constant velocity" characteristic of the cutter. However, there gradually emerged elsewhere the practice of boosting the treble response during record, in some cases, to quite substantial degrees.



The replay characteristics required (theoretically anyway) for optimum reproduction of 78 r.p.m. records made to various standards.

Theoretically, at least, optimum replay of these records called for different orders of bass boosting and different orders of treble cut, depending on the source of the disc being played.

Figure 1 is the reproduction of curves, which we drew out several years ago, indicating the characteristics required of an idealised amplifier system to replay accurately 78 r.p.m. records cut to the recording standards indicated.

It would appear, in retrospect, that the bass characteristics did occupy the spread more or less as indicated but we have reservations as to whether many 78 r.p.m. records effectively used the orders of treble boost needed to complement the replay curves as shown.

Incidentally, treble boost during record is commonly referred to as treble "pre-emphasis"; the cut introduced during playback is called "de-emphasis".

However, for the greater part of the 78 r.p.m. era, no very sophisticated design went into the average domestic record playing equipment. In the absence of deliberate bass boost in the amplifier, the bass end was more likely to be built up by positioning of the pole faces in the pickup, by resonance effects in the arm and stylus assembly, by resonance in the speaker and by the brute force method of lopping off the treble with a simple "top cut" tone control.

Assessment of the result was rather random, on the basis of some pickups and some records having more bass or treble than others, without much attempt to assess why this should be so or at what cost in terms of record wear, distortion, etc.

When crystal pickups first made their appearance, they achieved immediate fame for their bass response, again without a very wide appreciation of the reason behind this.

In fact, the explanation was that the crystal element produced signal output, not in proportion to the velocity of the stylus movement, but in proportion to the amplitude of movement. Therefore, for discs recorded to a constant amplitude bass characteristic, the crystal pickup produced a constant signal voltage—or level bass output.

TREBLE RESONANCE

To be sure, as a reverse bonus, its output tended to taper off over the approximate "constant velocity" portion of the characteristic but the treble response was rescued in most cases by the treble resonance of the stylus system.

And there the matter more or less rested until the immediate postwar period when the struggle began to lift disc reproduction from the backwater in which it found itself—a struggle which ultimately produced the modern LP disc.

In the process, recording characteristics, pickup design, amplifier compensation and so on came in for very critical examination and long forward strides were taken both in the professional and the home record-player field.

In setting up for LP discs, record manufacturers generally accepted the principle that the bass end would have to be recorded to an approximate constant amplitude characteristic for exactly the same reason as had applied for the 78s. Further, they were now unaminous that benefit could come from boosting the treble and producing something approaching constant amplitude above some other frequency.

To compensate these effects, "high fidelity" magnetic pickups would need to be fed into an amplifier having an inbuilt bass boost characteristic and an in-built treble cut characteristic, the latter serving both to equalise the end result and to reduce the already improved noise level of the new discs.

For mass-produced crystal pickups, the more nearly constant amplitude characteristic overall would be a boon, because they would produce a fairly level response curve without artificial compensation, creating an easy situation for the less expensive radiogram market.

But, unfortunately, for a wide variety of reasons, national and commercial, the various record manufacturers again failed to settle on a common recording characteristic.

They all used approximate constant amplitude at the bass end but, expressed in terms of frequency response, they chose slightly different turnover frequencies and limited the amount of restriction to varying degrees so that playback equipment would not have to provide an impractical degree of bass boost.

TREBLE RESPONSE

There were varying ideas, too, about how much treble boost should be incorporated into the disc and above what frequency.

Each manufacturer or group were prepared to back their opinion by releasing discs to their own "standard" and the first flush of LP's consequently appeared with different combinations of bass and treble characteristic, each one distinguished by some kind of name.

Thus, added to the already existing 78 r.p.m. standards, the new race of audiophiles was faced with the need for optimum playback curves like those shown (other than by the solid line) in figure 2.

Decca had their curve, EMI-Columbia theirs, RCA-Victor had another, while still other manufacturers settled for various attempts at standardisation such as the AES (Audio Engineering Society), the NAB, NARTB and so on.

Nothing daunted, equipment manufacturers matched this with amplifier control systems having a selection of compensation characteristics built in and selectable by a suitably calibrated switch. In fact, this sort of thing became a selling feature such that many audiophiles would not take seriously any amplifier system not so equipped.

A knowledge of recording characteristic became one mark of the expert!

HARD TO SUSTAIN

But, alas, it became increasingly difficult to sustain this ultra-purist approach. As manufacturers began to interchange program material by international agreement, the original recording curve was lost sight of behind a label which, in other circumstances, might sponsor a completely different curve.

Nor could the position always be corrected by noting the origin rather than the label. It all depended whether the program had been shipped as a metal "mother" or as a tape master, to be re-recorded in the new country! Only the "Perry Masons" could hope to keep track of the permutations and combinations.

Even apart from that, enthusiasts began to realise that discrepancies in the original recording system, the vagaries of their own pickup and, in particular, of their own speaker, often completely overshadowed the "official" differences in recording characteristics. The majority therefore gradually lost their zeal for complicated input systems.

And the recording companies, gradually losing their bias for their own viewpoint began to seek common ground. That common ground has emerged as the RIAA curve—that recommended by the Radio Industries Association of America—and embodied also as British Standard B.S.S. 1928/1955.

An important aspect of this curve was that it represented, not so much a digression from the industry's ideas, as a rationalisation of them into a sensible compromise. The equivalent playback curve is shown as a solid line in figure 2 and its median position is apparent.

Much to the gratification of RCA, the RIAA curve corresponded almost exactly to their "New Orthophonic" characteristic allowing them, perhaps unfortunately, still to assert their independence in the matter. Thus, even today, RCA's records carry the term "New Orthophonic" when, for all practical purposes, they could admit to an RIAA characteristic.

Over the past few years, the great majority of records have been made to the RIAA curve, as illustrated herewith. Where they are not so marked, it is still fairly safe to assume that the RIAA curve applies.

Thus, taking 1000cps as reference, the bass is attenuated during recording by about 13db at 100cps, making it necessary for an amplifier, operating from a flat magnetic pickup, to boost the bass by this same amount.

Below 100cps the slope of the bass attenuation curve flattens out, as also does the bass boost requirement in the amplifier. This is intended to make life just a little easier for designers, who have to cope in practice with playback motor rumble, cabinet and room acoustic feedback, floor vibration and hum pickup, all of which is made much more difficult by every extra decible of boost in the 20cps region.

At the top end, a boost of about 13db at 10Kc. calls for the same order of treble cut in the amplifier, for proper compensation.

The general adoption of the RIAA curve and its equivalent British Standard has resulted very largely in the disappearance of amplifier input switches calibrated for other recording standards. In most cases, the old 78 characteristic has disappeared along with them, the practical situation being that audiophiles are becoming less inclined to play old recordings, irrespective of what merits they might have had musically.

However, that does not mean that such recordings are unusable on equipment designed primarily for RIAA characteristic and, as often as not, simply marked "Pickup" or "Gram". In most cases old recordings, made to other standards, will sound just as good as they are likely to sound to modern ears, played to RIAA. If they don't, a touch on the variable controls to bring the bass or treble up or down, as required, should put the resultant curve either close to the "official" optimum or what sounds best, according to your point of view.

As for the old 78s, many will prefer them, these days, with treble cut to minimise the background noise. If not, the treble can be restored by using as much treble boost as seems desirable.

EQUIPMENT TO MATCH
I know full well that some will not agree with these remarks or the trend that they refer to. Some, and particularly those with a big collection of older LP's, cannot feel happy unless they have tracked down and directly compensated for each and every recording characteristic involved, and without relying on supplementary controls.

If your enthusiasm is of such an order, I guess the equipment must match it. However, I'm not sure that all of it is based on sound judgment.

For my part, I'm quite content with RIAA compensation only and the knowledge that balance could be modified, if need be, by a touch of the other controls. In fact, the need seldom seems to arise, but that part of the story can be left over till another time.

STYLUS COMPLIANCE, MASS, ETC.

What is the meaning of the term "compliance", as applied to pickup stylus assemblies? Should a good pickup exhibit high compliance or low compliance, and how is it denoted? What is its relationship to the "dynamic mass" of the stylus assembly? These questions are answered in a not-too-involved manner in the article which follows.

Such was the reception given to our article on pickup balance, that we would probably have tackled, anyway, other important pickup characteristics like this one, in due course, and in a similar fashion.

However, we were spurred on by a letter from a reader which read as follows:

Dear Sir,

To the best of my knowledge you have not yet dealt with the subject of compliance. To most of us, it is just one of a group of impressive looking specifications.

From what I can gather (and no one I asked really seemed to know) the higher the figure quoted for compliance, the more easily the stylus will be able to track the grooves, leading, in the ultimate, to lower stylus pressure and its associated benefits.

But I find the whole matter rather confusing. I say this because I have just been studying advertisements dealing with players and cartridges. Of those checked, the recent ones all seemed to confirm my theory. However, I really began to wonder when two separate advertisements appeared in earlier copies of this magazine both proudly boasting that their respective cartridges had low compliance! Both were by very well-known manufacturers, what is more.

Therefore, I respectfully submit that few people know what compliance really is, and suggest that you may be able to clear the whole matter up.

I must confess that though we have probably used the word quite freely, I can't recall an article dealing at any length with the subject of compliance.

What is more, a check through an armful of "popular" audio and hi-fi text-books on our shelves produced plenty of references to compliance but nothing much in the way of an actual definition. Far and away the most helpful reference came from that excellent book by F. Langford-Smith, "The Radiotron Designers' Handbook".

But, before we start quoting definitions, it might be better to go over a stretch of more familiar ground.

One end of a pickup arm is normally supported from a fixed base by pivots, which allow the head or stylus end to move freely in both a vertical or horizontal direction.

In the playing position, the head end is normally supported by the stylus, riding in the record groove. As we saw in the recent article on pickup balance, the stylus is not normally required to support the whole weight of the pivoted arm but only that designedly small portion of the weight which is not cancelled by counterbalancing or reverse spring pressure.

The stylus is not a rigid fixture within the pickup head, nor can it be. It must be able to vibrate from side to side, in accordance with the deviations of the normal laterally recorded monaural groove. To play stereophonic recordings the stylus must be capable of vibrating in a vertical direction as well.

These vibrations are normally conveyed to a small coil, or portion of a magnetic circuit, or a section of a "crystal" element, the movements relative to the fixed portion of the head serving to generate the relevant signal voltage.

FLEXIBLE MOUNTING

One scarcely needs to be a mechanical genius to realise that the stylus and anything fixed rigidly to it must, in turn, be attached to the fixed portion of the head by a flexible or compliant medium—a piece of rubber or soft plastic, a spring, a nylon thread or something of the kind.

It also follows, by fairly simple deduction, that the flexible medium, apart from just holding the stylus and assembly, must also serve to restore it, at all times, to a central position, about which the groove can cause it to vibrate.

But how strong should this restoring force be?

If its only job was to return the stylus to "normal" when not in contact with the groove,,it could afford to be very slight indeed. And this would be a very good thing. The more compliant or flexible the suspension, and the smaller the restoring force, the less will be the pressure which the groove has to exert on the stylus to make it vibrate laterally and/or vertically.

However, as we saw earlier, the stylus assembly has to do more than just follow the modulations of the groove. In fact, we can amplify and catalogue these extra tasks as follows:

(1) It has to support the downward vertical thrust of the pickup arm. Therefore the flexible or compliant support allowing vertical movement has to be as flexible as possible to permit the stylus easily to follow groove modulations, yet sufficiently inflexible as not to yield unduly under the constant or "static" load of the arm's downward thrust.

(2) The downward thrust is really only constant for a perfect turntable and disc. Practical turntables and discs are likely to show undulations which will cause a periodic rise and fall in the magnitude of the "static" load. The vertical flexibility has therefore to cope with this added problem.

(3) Since the groove is not a circle but a spiral, it diverts some of the turntable motor energy into moving the arm sideways, against the friction of the base pivot. In playing decks and changers additional stiffness may be introduced, over part of the travel, by the automatic trip mechanism. Thus the lateral stylus support has to be as free as possible to permit the stylus easily to follow groove modulations, but sufficiently inflexible to cope with the sideways thrust from the groove.

(4) In practical equipment, lateral thrust forces are likely to be complicated by eccentricity of the recording, vibration of the equipment as a whole and the tendency of a stylus to run inwards, due to tangential drag forces from the groove passing rapidly beneath it.

In short, the pickup designer is faced with the need to compromise. He would like to make the stylus support as flexible or as compliant as possible to allow the stylus to follow groove deviations but he must not ignore the very serious complicating factors, as listed.

In run-of-the-mill record players, these "complicating factors" may reach quite serious proportions so that the designer is obliged to make the stylus suspension much stiffer—or less compliant—than he would like. This, in turn, means that the groove has to exert higher forces on the stylus to get it to move, with consequent increase in groove and stylus wear.

GROOVE SKIP

Furthermore, the higher forces tend to make the stylus skip up and out of the groove, rather than to follow its deviations. Therefore, reduced stylus suspension compliance normally involves an increase in playing weight to keep the stylus firmly in the groove.

Again—higher wear!

Not only must the original designer face these problems but they also await the individual enthusiast who would seek to improve his otherwise run-of-the-mill playing deck.

He will discover that there is more to gaining the benefits of high compliance and low playing weight than just fitting a cartridge with the desired specifications. Unless he can effect proportionate reduction in the bearing losses, improve or remove the trip mechanism and possibly improve the balance of the arm, the delicate stylus suspension of the replacement cartridge may soon be tortured to destruction.

That, I suggest, is the reason why Decca chose to go so far but no further with their Dream "Auto Change" ceramic cartridge, reviewed some months ago in these columns; why they have now released a "Transcription" version of the Deram cartridge for use in better-quality arms including, of course, their own.

Perhaps we can sum it all up this way:

A pickup stylus mounting should exhibit the highest practical degree of flexibility or compliance because this makes it easier for the stylus to follow deviations of the groove. Immediate benefits of high compliance include improved capacity to track heavily modulated passages (resulting in lower distortion) and the possibility of doing so with low playing weight (resulting in reduced stylus and record wear.)

However, the above benefits can only be realised if the arm mechanism is sufficiently refined as not to impose impossible conditions upon the highly compliant (and often delicate) stylus support system.

ARM REQUIREMENTS

This is the precise reason why so much attention is paid to the design of playing arms for highly developed record playing systems; why enthusiasts so commonly discuss things like bearing drag, stiffness of connecting leads, tangential thrust, dynamic balance and such like; why they might pay as much money for a pickup arm as other people pay for a complete player!

Nor has the process reached any finality. Lots of excellent cartridges and arms are available today, which can be used with one another. But there is ever the tendency to strive for a combination of the two which will track heavily modulated passages better, and with lower playing weight, than any other.

Now we can quote a definition or two! G.A. Briggs puts it this way:

COMPLIANCE: The inverse of stiffness. The yielding qualities of the members which suspend the moving parts of a pickup or a loudspeaker.

F. Langford-Smith says:

COMPLIANCE: The ratio of the displacement of a body to the force applied.

You can take your pick as to which of these definitions best fits your pattern of thinking, but they add up to the same thing . . . high yielding qualities or large displacement of the body for a given applied force.

On quantitative basis, the displacement of a pickup stylus is normally expressed in terms of microcentimetres (ucm), and the applied force as one dyne. There is, of course, a regular definition for a dyne, but it is roughly equivalent to one thousandth part of a gram, considering it as a weight acting in the particular direction.

Up till fairly recently, the compliance of pickup stylus assemblies was rarely quoted and it is still not often quoted for run-of-the-mill cartridges. However, it has come to be regarded as an important specification of a cartridge making any real claim to high fidelity.

By way of example, an early General Electric variable reluctance cartridge was credited with a compliance of 1.7 microcentimetres per dyne. The Goldring variable reluctance mono cartridge claimed 5 u cm/dyne. This latter could still be regarded as a good round figure to achieve, even though the Acos mono Hi-Light cartridge is quoted as high as 12 u cm/dyne!

Vertical compliance which, of course, must be considered in stereo pickups, is commonly lower than horizontal compliance (one-half or less) probably because vertical suspension has to be designed to cope with more formidable external forces than the horizontal suspension system.

There is one more important aspect of this whole question. Compliance describes the ease with which the stylus assembly may be displaced from normal, but if it has only a remote relationship to the ease with which it may be accomplished at a high rate of speed as, for example, at high audio frequencies.

Here we become concerned, not just with the yielding qualities of the suspension, which might be considered as something of a constant, but the mass of the system, which has to be accelerated.

Two heavy children on a balanced see-saw don't call for much effort to rock them gently up and down. But try to rock them at a high speed and see how much effort is required!

In the language of pickups, it is necessary to consider the mass of the assembly which needs to be accelerated. This is far from being a simple quantity because not all sections of the assembly–like all sections of a lever – need to be accelerated through the same distance. Again, there may be a certain amount of compliance or springiness in the assembly which tends to isolate some of it from the actual stylus tip.

This necessarily complex quantity is normally resolved by relating it all as a single, effective mass at the stylus tip. This is referred to as the "effective stylus mass" or the "dynamic mass" and indicates the equivalent mass which has to be accelerated if the stylus is to track faithfully the deviations of the groove.

Ideally, the dynamic mass should be as low as possible. This, along with high compliance, allows the stylus to track the groove to best advantage, at all frequencies involved, and with the lowest practical playing weight.

As with compliance, dynamic tip mass is seldom quoted for run-of-the-mill crystal cartridges and, in fact, it was seldom quoted for the first generation of high fidelity magentics.

Initially 4 or 5 milligrams was reckoned as reasonable, but this has gradually been reduced until several pickups have now achieved the very commendable figure of one milligram or even a fraction less. In fact, this would have appeared to have become the present-day target and its achievement, along with high compliance, generally spells an ability to play all recordings, heavily modulated and otherwise, with a playing weight of about 2 grams.

Incidentally, compliance of the suspension and dynamic mass at the stylus tip are the principal quantities which determine the natural resonance of the stylus system and high compliance must be accompanied by low tip mass if the resonance to is to be kept above the normal audio range. But perhaps we've gone far enough in what we hope has been a helpful discussion.

DISC RECORDINGS – THEN AND NOW

This article relates something of the background of disc recordings as we now know them. In particular, it explains why a so-called "recording characteristic" is necessary and what it involves in terms of frequency response.

The history of "gramophone" or "phonograph" recording makes a fascinating study but it is not our purpose to dwell upon it here. It takes in the work of people like Leon Scott who, well over a hundred years ago, established a connection between sound, and wave patterns capable of being produced and inspected. It would dwell heavily on the work of Thomas Edison who, in 1877, succeeded in recording sound upon the surface of a cylinder and playing it back.

The technique of recording on flat discs is normally credited to Emil Berliner, from work done just before the turn of the century. Technical arguments aside – and there were many – Berliner's flat discs offered the overwhelming advantage that they were capable of ready duplication by a plating and pressing process. In addition, they were cheaper to market and easier to store. By the 1920s, Berliner style discs, spinning at a standard 78rpm, had virtually ousted Edison cylinders and laid the foundation for what has been a very successful industry ever since.

What we have to say virtually begins at this point.

Inscribed or pressed into the surface of a disc recording is a groove which spirals gradually inwards from the outer edge towards the centre. If this groove is examined under a powerful magnifying glass it will be seen to contain tiny, wave-like deviations, produced when audio signal currents through the cutting head caused the stylus to vibrate (figure 1).

For those with a mathematical turn of mind, it is appropriate to regard the groove as an inscribed graph, in which the instantaneous value of the audio signal (side-to-side deviation) is plotted against elapsed time (along the spiral).

When the groove is subsequently traced by a stylus in an electrical replay head, the replay stylus must substantially duplicate the deviations of the recording stylus. As a result, the replay head will deliver an electrical output signal closely equivalent to that which produced the original recording and this can be amplified to drive a loudspeaker.

While these broad principles are easy enough to grasp, their application requires a decision about many of the working parameters. Most obvious, perhaps, are the diameter of the disc, the pitch of the spiral and the speed of rotation. A large disc, a fine spiral and a slow speed all contribute to a longer playing time but other considerations intervene to set limits on these various quantities.

Then there are details of the groove itself – its shape, depth, width, the nature and extent of the deviation with cutter modulation and the nature of the material which makes up its walls; in short, the material of which the record itself is made.

Closely dependent on groove characteristics is the nature of the replay needle or stylus – the shape and dimensions of its tip, the way in which it is supposed to ride in the groove, the material of which the stylus is made and the weight of the pickup head which it can reasonably be expected to support.

As might be expected, there have been numerous arguments through the years within the record industry regarding appropriate dimensions and conventions but various "standards" have emerged out of the sheer commercial necessity that records from any source be playable on equipment from any other source.

The earliest conventions within the disc recording industry were established to cope with mechano-acoustic methods of recording and playback. In the recording studios, sound waves, collected by a large horn, were concentrated on to a diaphragm directly connected to the recording stylus. Many compromises had to be accepted in order to satisfy the prime requirement that there should be adequate modulation of the groove.

Reproducing sound from a disc was essentially the reverse of the recording process. The modulated groove was traced by a "needle" directly coupled to a thin diaphragm located at the throat of a horn. Air pressure waves created at the surface of the diaphragm were propagated into the listening room. Since the system was devoid of electrical amplification, the listener was entirely dependent for adequate sound on the efforts of recording engineers to secure adequate deviation of the track.

Because the playback needle had to perform significant "work" in driving the linkage, the diaphragm and the air in the horn, its movement was quite stiff. To hold the needle in the groove, the playback head had to bear down heavily upon it, with a playing or tracking weight which was, in fact, measured in ounces.

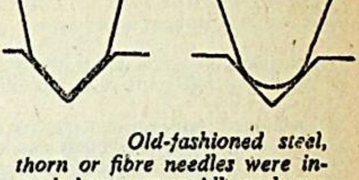
On the slender needle point, this represented a tremendous pressure per unit area, and very rapid wear had to be expected of the needle point, or the groove walls, or both.

Early practice was to regard the needle as the more logically expendable item. The formulation from which discs were pressed contained a certain amount of abrasive filler, the effect of which was to rapidly grind the steel needle to conform to the shape of the particular groove: in so doing, it distributed the downward thrust over a greater surface area. The granular nature of the shellac-plus-filler formulation added a considerable amount of "crackle and pop" noise to the reproduced sound but this came to be accepted as almost an intrinsic part of reproduction from disc.

It certainly had the designed effect on hard steel needles. Older readers will remember the days when one purchased them by the hundred, each small tin carrying the advice "replace needle after every playing".

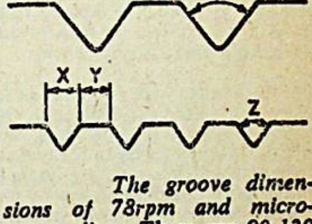
Around 1925 microphones, amplifiers and electrical recording heads appeared, giving engineers greater flexibility in the kind of material they could record, more precise control over amplitude, smoother and wider frequency response and lower distortion. Then, in the late twenties, electrical playback heads began to appear in quantity, delivering an electrical output signal which could be passed through an amplifier and loudspeaker system for reproduction.

Under a microscope the tracks of a conventional mono disc will be seen to deviate from side to side, giving rise to the term "lateral" recording. The deviations are exaggerated in the drawing for the sake of clarity.



Old-fashioned steel, thorn or fibre needles were intended to wear rapidly and conform to the shape of the groove, as in (a). Jewelled styli were radiused accurately to be gripped between the walls, as in (b).

In normal two-channel stereo disc recording, independent modulation on the two 45-degree groove walls produces the respective output signals (upper sketch). By suitable phasing of the drive to the cutter, a centre-image sound will produce a substantially lateral resultant, similar to a mono signal (lower sketch).



The groove dimensions of 78rpm and microgroove discs. There are 90-120 grooves to the inch on standard 78rpm records (top), the dimensions being X 0.006 inch, Y 0.004 inch. Microgroove records have 200-300 grooves per inch with dimensions X 0.002 inch to 0.003 inch, Y 0.0017 inch. The included angle Z is approximately 90 degrees in each case, and the maximum bottom radii are 0.001 inch and 0.00015 inch respectively.

Since the needle in these new heads did not have to drive a diaphragm directly, they held the promise of a less stiff – or more compliant – needle system, reduced playing weight and therefore reduced needle and record wear.

In fact, designers seemed to be rather slow in exploiting this possibility and the majority of early electrical pickups were cumbersome and used exactly the same needles and the same heavy chuck system as acoustic phonographs of the day. It was well into the thirties before the idea of higher compliance, ligher weight pickups began to catch on.

Conservatism notwithstanding, the thirties saw steady progress in the quality of signal impressed on to discs and subsequently recaptured during playback.

Unfortunately, however, the industry of the day was soundly wedded to the idea of compatibility. Engineers could improve the records if they wanted to but only within the framework of existing practice. Anything they produced still had to be fully compatible with the requirements of acoustic phonographs – even if most of those still in use were in underprivileged countries!

One important development which did come near the end of the 78rpm era was that of truly light-weight, high-compliance pickups fitted with long-wearing (wrongly called "permanent") styli. The sapphire tips were ground and radiused, more or less accurately so that the stylus would ride part way down the V-shaped groove, gripped between the two walls (figure 26). So held, it could be expected to trace the groove more accurately and be more responsive to very fine deviations corresponding to the higher modulation frequencies. Ability to respond to the higher frequencies was further improved by a reduction in the mass of the stylus and elimination of the traditional and cumbersome needle chuck.

The standards which had more or less emerged by the end of the 78rpm era are set out in the upper diagram, figure 3 and under the heading 78rpm in the table of figure 4. It is important to stress that these were praotices ultimately adopted by a majority in the industry; earlier records in particular may show discernable differences from these figures. More subtly, the grooves in some earlier records differed from the shape depicted and, if played with a jewelled stylus, needed one with a radius as great as .0035in., as compared with the normal standards of .0025in.

After the war, renewed efforts were made to upgrade the performance of 78rpm equipment, particularly by those who believed that the established standards had to be preserved. However it was becoming increasingly clear that the disc record could not offer better fidelity, lower surface noise and longer playing time – all highly desirable qualities – until industry threw off the schackles of the acoustic era and adopted a system designed expressly – and only – for electrical reproduction

Accompanied by a great deal of argument and rivalry, a new set of standards emerged soon after the war or, rather, a new set of practices which gradually emerged as standards. They can be summarised as follows:

RECORDS: Shellac and fillers formulation gave place to plastic materials such as vinyl and polyethylene. The substitution of an homogenous material for one which was essentially granular in nature, made possible a dramatic reduction in record surface noise. It became possible to record, and play back successfully from, a track exhibiting much smaller deviation.

PICKUPS: The use of plastics for commercial record pressings presupposed that they would always be played with lightweight electrical pickups. In fact, the situation posed a fresh challenge to designers to produce pickups which would track successfully at a playing weight of a few grams at the most. This would necessitate using a needle or stylus system exhibiting very high compliance (a minimum of stiffness) and very low effective tip mass (low inertia or a minimum of weight for the groove to accelerate to and fro).

STYLUS: While new, very light-weight pickups could conceivably use new, lightweight steel needles (in fact, a few did) it was evident that jewelled styli would be preferred by the public. Needing to be changed only occasionally, they obviated the earlier tedium of chainging needles after every side.

GROOVES: With the assumption that only lightweight pickups would be used, and a further assumption that the public could be trained to handle less massive equipment, the industry opted for grooves of approximately one third the width and depth of those which had for decades been used with 78rpm records. Furthermore, they were to be recorded much closer together, averaging from 200 to 300 to the inch instead of the earlier figure of 90-120. In fact, it was evident that the groove spacing could be varied in the more refined type of recording lathe, the grooves being packed tightly together during the quiet passages and opened up in anticipation of loud or heavily modulated passages.

The new type of record actually took its name from the new groove dimensions and "microgroove" became the term which distinguished the new generation from the older "coarse groove" 78s.

SPEED: With the reduction in groove and therefore stylus tip dimensions, it became possible to record and retrace deviations of much shorter wavelength, equivalent to sounds of much higher frequency. The industry decided to trade some of those potential advantages for longer playing time and, after a great deal of technical and commercial argument, two new speeds emerged: 45rpm, mainly for 7-inch pressings of shorter items and 33rpm for 10-inch or 12-inch longer playing pressings for collections or major works. Both are "microgroove", however, with the same standards for groove and stylus.

PLAYING TIME: Lower rotational speed and closer groove spacing provided what the disc record industry desperately needed – a longer playing time per side. The new standards made possible recordings of major works with a playing time of 30 minutes or more on a single 12-inch side.

MUSIC SOURCE: In the recording studios, a vitally important development made it possible to take full advantage of the potential of microgroove records – the use of magnetic tape for master recordings. Prior to this, original recordings had to be made directly and without interruption on to a master disc, and the problems of obtaining 30 minutes of flawless recording were very great. When it became possible to make the master recordings on tape, studios had at their disposal the means to re-record faulty passages, to assemble one optimum performance from a number of takes and assemble the material for collections by stringing together individual recordings in appropriate order.

	Coarse groove Mono		Microgroove			
			Mono			Stereo
Nominal Speed (rpm)	78		33-1/3		45	33-1/3
Nominal Size (in.)	10	12	10	12	7	12
Outer Diameter of recorded surface (in.)	9.52	11.52	9.52	11.52	6.625	11.52
Inner Diameter of recorded surface (in.)	3¾		4¾	4¾	3¾	4¾
Centre hole (in.)	0.285		0.285	0.285	0.285 or 1.5	0.285
Average pitch (grooves per in.)	100		250			250
Stylus tip radius (in.)	0.002-0.003		0.0007-0.001			0.0005-0.0007
Stylus included angle	40 deg.-55 deg.		40 deg.-55 deg.			40 deg.-55 deg.

Figure 4: This table compares the main features of the 78rpm, microgroove mono, and microgroove stereo disc record systems. A few 7-inch mono records have been released, intended for playing at 33rpm, while a few 7-inch 45rpm records and 10-inch 33rpm records have been released in stereo. Such information is normally given on the jacket and/or label.

With the new standards and the impetus to re-equip after the war, microgroove recordings and playing equipment found rapid and wide acceptance both by high fidelity fans and at general listener level. Supporters of the older system brought forward a variety of arguments to support its retention but it rapidly lost ground in the face of obvious advantages of the new: Lighter, "unbreakable" pressings, informative and decorative cover sleeves, longer playing time, lower noise, lower distortion, wider frequency response.

At the same time, however, the new system was threatened by the rising popularity of domestic tape equipment which offered one major advantage over disc. It could present multiple signal tracks, making possible stereophonic reproduction. Instead of having to listen to sound concentrated through a single channel, multi-track recording and playback allowed the sound to be spread across an area fronting the listener, creating a more natural and more pleasing effect of dimension and space.

In fact, the industry already possessed the potential answer to this challenge, dating back to work done around 1930, by A.D. Blumlein.

This involved the use of a special recording head, capable of modulating the stylus tip both horizontally and vertically. One early scheme was, in fact, to record one track in the horizontal mode (lateral recording) and the other in vertical mode (hill and dale recording). This and other such ideas were ultimately put aside in favour of using the respective signals to drive the cutting stylus tip at 45 degrees, as illustrated in figure 5. In effect, the information is modulated into the respective groove walls, one signal into each wall.

For playback a suitably designed pickup is necessary, with a stylus system capable of tracing deviations having vertical as well as horizontal components. Put another way, the stylus has to be capable of moving up and down as well as sideways. In addition, the internal transducer system has to be capable of resolving 45-degree vibrations into separate output signals, equivalent to those which were used originally to drive the cutter.

Investigation of the 45-45 stereo system indicated that a further reduction in the tip radius of the replay stylus would be desirable – and also practicable in view of the dramatic way in which designers had succeeded in reducing the tracking weight of ordinary pickups.

However, the industry faced the problem of introducing a new standard hard on the heels of the microgroove system. Would the listening public take it up?

An obvious prerequisite seemed to be that the system would have to be compatible with the existing monophonic (i.e. single-channel) microgroove, at least to the extent that purchasers of new stereophonic equipment must be able to play their existing microgroove records without prejudice.

Guided by this requirement, the industry nominated 0.5mil (.0005in) as a desirable figure for the tip radius of a stereo stylus but settled upon 0.7mil (.0007in) as an acceptable compromise for playing both stereo and mono discs.

It was reasoned that such a stylus would play stereo discs reasonably well, offering the additional advantage that it would wear less rapidly than the finer tip. At the same time, it could be expected to play the vast majority of mono discs rather better than .001in styli, the sole exception being those mono discs which had been recorded (in many cases inadvertently) with a groove having a heavily radiused bottom. All future mono discs could obviously be recorded with the finer replay styli in mind.

A second measure was to ensure that the cutting system was so arranged that signals arriving in phase at the microphones would interact to produce a substantially lateral modulations of the groove. Thus a performer standing centre-stage would produce a substantially laterally modulated groove, similar to a mono recording. The pickups, of course, would be wired to correspond, to produce a centre-image sound from a centre-stage performer.

In these circumstances, a stereo system could be used to reproduce mono records directly, the lateral track producing a centre-image sound just as it, in fact, an actual mono loudspeaker were operating at that point.

As things worked out, purchasers of stereo equipment showed a clear preference for stereo records when adding to their collections and, apart from the odd disc as already mentioned, experienced no difficulty in playing mono records without change to the system. For all practical purposes, stereophonic systems proved to be completely compatible with existing and new mono records.

The reverse is not necessarily true, however and, for many years, listeners were warned against playing stereophonic records on monophonic equipment.

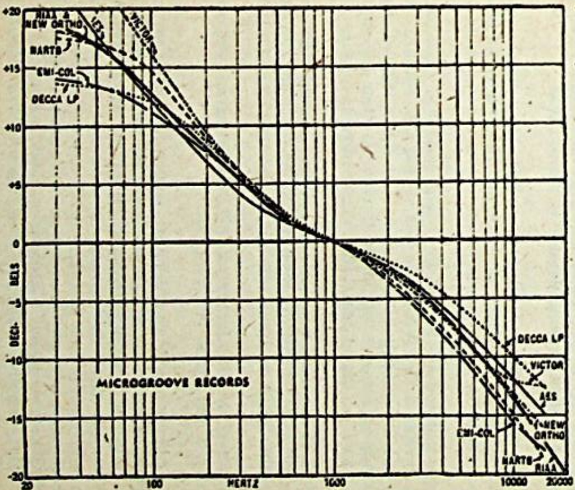
Superficially, stereo records will, in fact, play and sound normal on mono equipment. The speeds are the same and, because of the way the signals on the stereo disc are phased, a mono pickup will derive from it a fairly normal sounding mono signal.

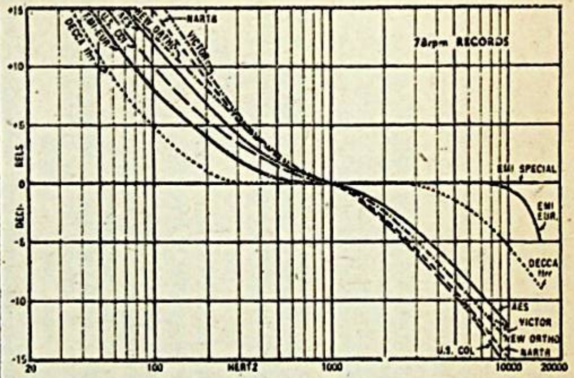
The big problem is that the stylus system in many mono pickups has very little vertical compliance; it will move from side to side quite freely but is resistant to vertical movement. When a stereo record is played with such a pickup, it may damage the groove in respect to vertical components of modulation and therefore ruin it or compromise its quality for subsequent stereo playing. Hence the early warnings.

During the last few years, the stereo system has gained tremendous impetus to the point where stereo equipment completely dominates the quality market and the call for 12-inch mono pressings has fallen to an uneconomic level. This has led to the introduction of so-called "compatible stereo" records, which are supposed to be stereo records capable of being played normally and without damage on existing mono players.

From the purist viewpoint, the concept is unacceptable, since a mono pickup which would damage stereo records 10 years ago, will still damage stereo records at the present time. Or again, if a record has a genuine stereo content, then it will have vertical components of modulation and these will be just as vulnerable now as they would have been 10 years ago!

The difference in the situation is more one of degree. Faced with the problem of reverse compatibility, a record manufacturer may choose to control the placements of instruments and/or the stereo mix so as to avoid excessive vertical modulation, not vitally necessary to the stereo effect.





These two diagrams depict the optimum frequency compensation characteristics for playing back various brands of 78rpm record (figure 7, above) and of mono LP records (figure 8, left).

His "compatible stereo" record may, therefore, contain less exacting vertical modulation than a stereo record made without this precaution in mind.

At the same time, care may be taken with the microphone placement, the stereo mix and even the musical arrangement itself to ensure that the mono listener obtains a satisfying version of the multi-track sound.

As far as pickups are concerned, the passing years have seen a progressive improvement in the tracing characteristics of mono units and the odds are rather more in favour of stereo records escaping calamitous damage if played thereby. And there are fewer mono pickups to be reckoned with anyway!

Lastly – and rather grimly – if a listener buys compatible stereo records and damages their stereo content by repeated playings with a totally unsuitable pickup, he may not be actively aware of the fact until some time in the vague future.

If a particular listener is wedded to a mono system for any reason, and is dismayed by the diminishing supply of mono discs, two courses are available:

1. Re-fit the pickup with a mono cartridge of suitable type, which is known to have adequate vertical compliance. Manufacturers are now offering compatible mono cartridges for this very purpose.
2. Re-fit the pickup with a stereo cartridge, which can be as budget-priced or as expensive as desired, and connect the output terminals to produce a mono signal suitable for a mono amplifier system.

Either course will render a mono system truly compatible, without relying on the element of luck to avoid damage to stereo discs.

One of the aspects of disc recording which frequently puzzles enthusiasts has to do with talk of frequency response and frequency compensation. What is it all about?

Earlier, it was pointed out that, in the lateral system of recording, the signal is used to vibrate the stylus from side to side. The amount of deviation which can be tolerated depends a good deal on the spacing of the tracks. If the deviation is excessive in relation to track separation, the grooves may intersect on adjacent signal peaks, with rather obvious results.

The deviation of the track is directly related to the amplitude of the signal being recorded and it is the responsibility of the recording engineer to see that permissible limits on amplitude are not exceeded, even momentarily.

There is, however, another factor, less obvious but no less important, namely the frequency of the signals being recorded.

When sound energy of a given loudness is applied to a diaphragm, it causes the diaphragm to accelerate to a proportional velocity. If the sound is of a very high frequency, the diaphragm will not have traversed a very great distance before the sound wave reverses in pressure phase, causing the diaphragm to reverse its direction. At a much lower frequency, the diaphragm will tend to move through a greater distance before it reverses direction. In short, for a given sound energy level, the amplitude of movement of a vibrating body, either producing the sound or responding to it, will increase for diminishing frequency.

A familiar example can be found in the strings of a piano or, again by comparing the high-note string of a violin with the low-note string of a double bass. The amplitude of movement of a string producing a very high frequency is quite small; that of a string producing a very low frequency, visibly apparent.

A similar comparison can be made between the diaphragm movement of a high frequency "tweeter" loudspeaker and a low frequency "woofer". Both may be radiating sounds of designedly comparable loudness but, whereas the tweeter loudspeaker cone will not be moving perceptibly, that of the woofer may be oscillating by plus and minus a quarter-inch or so.

In the case of a disc recording cutter, either driven directly or per medium of an amplifier system, the same tendency is evident. If it is allowed to move naturally with a velocity proportional to the loudness of the original signal, and irrespective of the frequency of that signal, it will oscillate through a very large distance for the lower signal frequencies.

And here the recording engineer faces a number of conflicting factors. If the stylus is allowed to follow its natural inclination at very low frequencies, the deviation will be so great as to dictate a very wide groove spacing. This must reduce playing time. In addition, the deviation may be of such an order as to make it difficult for practical pickups subsequently to retrace the groove.

On the other hand, if the low frequency waveforms are restricted to an acceptable amplitude by simply reducing the level of the recorded signal as a whole, this level may turn out to be so far down at middle and high frequencies, that it will be comparable with or below the inherent noise of the disc and reproducing system.

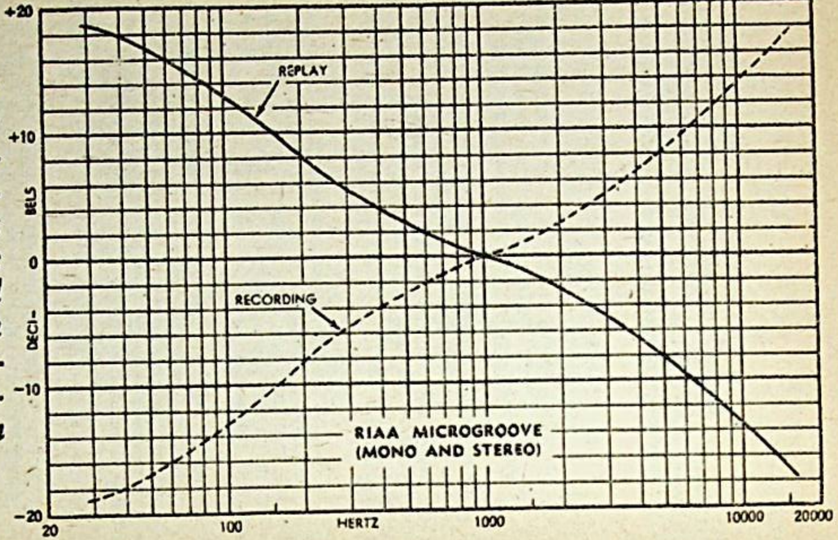
Quite early in the history of recording, and certainly with the introuction of electrical recording, engineers realised that a deliberate compromise had to be arrived at. Clearly enough, they had deliberately to restrict the tendency for groove deviation to increase with diminishing frequency and it seemed logical that matters be arranged so that, below a certain frequency, for a given sound energy level, the recorded groove should remain at a constant deviation amplitude.

To introduce here a couple of technical terms, this meant that below a certain frequency (conveniently referred to as the "turnover" or "corner" frequency) the groove should be given a "constant amplitude" characteristic. Above the turnover frequency, it would revert to its natural "constant velocity" or unmodified velocity characteristic.

With electrical components and circuitry, it is not unduly difficult to achieve such an effect. A selected value of coupling capacitor at a strategic point in the circuit can introduce just the right kind of roll-off at the low frequencies to produce a constant amplitude characteristic below a selected turnover region. Inductance can be used in the same way.

The vital point, however, is that, in imposing a constant amplitude characteristic on the bass region, the recording engineer is effectively imposing a bass cut – and a recording characteristic is born.

The RIAA recording characteristic (dotted) and the equivalent playback charac t e r i s t i c (solid line). As will be apparent from figure 8, it occupies a position intermediate between the older curves.



If the recording is to sound properly balanced on playback, the bass needs to be restored or boosted by the same amount as it was attenuated earlier – and th need for playback compensation becomes evident. Hence a playback characteristic.

In the twenties and early thirties listeners were dependent on playback devices which gave an output nominally proportional to velocity – acoustic phonographs or magnetic pickups – and the niceties of applying bass compensation were either not widely appreciated or difficult of achievement.

Use was typically made of such things as pickup arm resonance, or magnet/ armature spacing but the end result generally left a good deal to be desired.

If engineers cut the bass too heavily, in the interest of playing time, their products were likely to be criticised as sounding "too thin". If they opted for more bass (and wider track spacing) the playing time was too short! If they sought both, by limiting total recorded amplitude, the records would be "too weak". Later in the 30s, as the quality of playback equipment improved, some reassessment of the position became possible and companies adopted revised practices, which seemed best to suit the current market.

Interestingly enough, the "revised practices" affected more than the bass. Toward the end of the 78rpm era, many of the major companies, particularly American, began to move away from the European convention of a flat treble response to a policy of deliberately boosting the treble. Implicit in this was a desire to meet the needs of electrical rather than mechanical reproduction and to provide recordings which would sound brighter (particularly with crystal type pickups) and in the knowledge that overbright reproduction could be corrected readily by the ordinary amplifier tone control. In fact, some of the practice established with the last generation of 78rpm records carried over into the microgroove era; this much will be evident from an examination of figures 7 and 8.

Not surprisingly, out of all this change and compromise came many different recording characteristics and, by inference, the same number of complementary replay characteristics. Figure 7 shows a collection of replay characteristics and emphasises the problems of enthusiasts who, for one reason or another, have since sought to play back an assortment of 78rpm records to best advantage.

With the introduction of microgroove records, engineers had to re-think the position. Having no longer to worry about acoustic phonographs and with industry well attuned to the need for playback compensation, they were free to impose any limitation on the bass recording characteristic that seemed appropriate.

There was also fairly common agreement that it should be possible to apply treble boost during recording. This would necessitate playing back with amplifier systems having an equivalent amount of treble cut – a provision that would tend very usefully to reduce any residual surface noise from the disc.

Unfortunately, and despite the lessons from the 78rpm era, technical disagreement and commercial rivalry prevented the immediate adoption of an industry standard and early microgroove records were recorded to a variety of nominal standards, as indicated in figure 8. Amplifiers of the day reflected this situation in that many of them were provided with switches allowing particular playback compensation characteristics to be selected.

It took several years for the industry to adopt anything like a common standard but this is now more or less universal. Known as the R.I.A.A. characteristic, it is illustrated in figure 9. The solid curve shows the recording characteristics and the dotted curve the complementary characteristics which should be provided by the playback amplifier.

Fortunately, the R.I.A.A. characteristic represents something of an average of earlier practice and can reasonably be used as a playback characteristic for all microgroove records. The purist may object that there are significant differences between the R.I.A.A. characteristic and certain of the others – but it does not follow that practical pressings were ever as precise as the curves might suggest. There is a lot to be said for being content with the R.I.A.A. characteristic (which is all that is available on modern amplifiers anyway) and nudging the bass and/or treble controls slightly if further correction seems to be desirable.

Fortunately, the R.I.A.A. characteristic was adopted forthwith and almost universally for stereo records and, in fact, has become as universal as the stereo record itself.

This, then, is the background to disc records, the various standards to which they have been recorded and their frequency characteristics. What all this means in terms of playback equipment must be the subject of another chapter.

HOW DO WE EVALUATE LOUDNESS?

This article, condensed from the "Hewlett-Packard Journal", was written as a preamble to a discussion on sound level meters and loudness evaluation. Although not strictly "audio" it covers such important subjects as the ear's response to frequency, units in which sound and loudness are evaluated, subjective pitch, and similar subjects.

Prolonged loud noise damages hearing, makes sleep difficult, makes us irritable, and interferes with our ability to think. Very loud noises can cause pain, nausea, fainting, fits, psychosis, or death.

The sad truth is that our environment is getting noisier all the time. If this bothers you, you aren't alone. There is a great deal of effort being expended these days to reduce the amount of objectionable sound that bombards us. In these noise abatement efforts, the measurement of loudness plays a critical role.

Not all sounds are noise, of course. Many sounds carry information which is useful or essential for our life. Speech and music are two examples; the sound of a motor car horn is another.

Sometimes it isn't easy to decide whether a sound is information or noise. Often it is both. For example, the sound of a machine can be considered an information-carrying sound because it tells the machinist whether or not his machine is functioning properly. But for his neighbour who is operating another machine, this sound is noise – it carries no useful information.

Most of the everyday sounds we hear are noise to us; yet many of them carry information for someone else. It is a function of society to establish limits to keep noise to a minimum while insuring that information-carrying sounds are audible to those who need to hear them.

If we want to define such limits we have to be able to measure them. This turns out to be a difficult task because the yardstick that must be applied is the subjective sensation of loudness, that is, loudness as it is experienced by people. This sensation seems to involve complicated physiological and psychological mechanisms.

A good loudness meter would have to imitate many unique properties of the human ear. These properties have been extensively investigated by a great number of scientists. However, we still do not have a very good understanding of the physiological processes underlying many of them; our knowledge of these properties is only empirical. We still can't make a complete model of the ear. Nevertheless we have learned to make fairly accurate models of the loudness-sensing function of the ear.

A number of approximations have been formulated for computing a quantity proportional to loudness, using the results of more-or-less detailed analyses of the noise to be evaluated. We shall discuss three of these methods in this article. Two of these are the calculation methods of Zwicker and Stevens; the third is the comparatively simple sound-level meter. The methods of Zwicker and Stevens have been internationally accepted in Recommendation 532 of the ISO (International Organisation for Standardisation). Except for some recent refinements, the sound level meter is described in ISO Recommendations 123 and 179.

Some scientists believe, with good reason, that loudness is not a completely satisfactory measure of how much a sound will disturb a person. Attempts have been made to define a better measure, called annoyance. So far, these attempts have not met with much success, chiefly because of the large number of unknown psychological factors that contribute to the effect of any sound on any individual at any time. These factors include such things as a person's past history, his present state of mind, what he is trying to do at the moment, and so on. One definition of annoyance that has found some acceptance is Kryter's "perceived noise" concept (ISO R 507, later modified), which uses a method similar to Stevens' loudness-computing method to arrive at annoyance in PNdB.

Kryter's method is designed primarily for the type of noise produced by jet aircraft. At present, this method is in a state of flux, and no one is certain what its final form will be. Some experts feel that a modification of the simple sound level meter should give adequate results for jet aircraft. It appears, therefore, that until our understanding of the psychological effects of sound improves greatly, the only reasonably objective measure of the disturbing power of a sound is its loudness.

Sound at a particular point is a rapid variation in pressure at that point around a steady-state value. In air, the steady-state pressure is atmospheric pressure (which actually changes, but slowly enough to be considered constant compared to the rapid pressure variations of sound). Sound pressure is measured in the same units as atmospheric pressure. It is an alternating quantity, and usually the term "sound pressure" refers to its RMS value.

At a frequency of 1KHz, a sound with an RMS pressure of 2×10^{-4} μbar*, or about 2×10^{-10} atmosphere, is just below the threshold of hearing for good ears; that is, a sound of this magnitude is inaudible, but slightly larger sound pressures can barely be heard. This demonstrates the amazing sensitivity of the human ear – it can detect variations in atmospheric pressure as small as a few parts in 10^{10}.

Another of the remarkable properties of the human ear is its large dynamic range. At 1KHz, it can hear sounds as small as about 2×10^{-4} μbar, and at the other end of the sound-pressure scale, it can accommodate sound pressures as high as 200 μbar without becoming overloaded. Bigger sounds, say 2,000 μbar, are physically painful.

Because the dynamic range of the ear is so large it is common practice to use a logarithmic scale for sound pressure. A reference value of 2×10^{-4} μbar, approximately the threshold of hearing at 1KHz, has been agreed upon. RMS sound pressure is commonly expressed in dB above 2×10^{-4} μbar and referred to as sound pressure level. Mathematically, if p is RMS sound pressure and P is sound pressure level, then

$$P = 20\log_{10} \frac{p}{po} \text{ dB}$$

where $po = 2 \times 10^{-4}$ μbar.

In terms of sound pressure level, then, the ear's dynamic range is about 120dB. Not many electronic instruments can match this.

Sound in its environment can be thought of as a field, just as electromagnetic waves are fields. Three common types of sound fields are the plane sound field, the spherical sound field, and the diffuse sound field.

Sound in a homogeneous space propagates outward from a source in all directions and consequently forms a spherical field. In a spherical field, the sound pressure decreases with the square of the distance from the source. When a microphone is relatively far away from a source the sound field may appear to be a plane field, in which the sound pressure is constant in any plane perpendicular to the direction of propagation.

If sound is generated in a room, sound waves are reflected from the walls, and a directional sound field can only be found very close to the source. Furthersfrom the source, sound approaches any point uniformly and randomly from all directions. Thus the sound field is diffuse. Such a field would be found in a factory if the nearest machine were not too close.

It is often important to know whether the sound field in an area is approximately plane or diffuse. If it is plane, directional microphones can be used with advantage to measure it; it it is diffuse, omnidirectional microphones are needed. Often a field will be partly plane and partly diffuse. In a factory, for example, a machinist is in the directional sound field of his own machine but in the diffuse sound field of noisy machines in the distance.

The transition from a directional sound field to a diffuse sound field in a room is characterised by a critical radius, which can be estimated as follows:

$$r_G = 0.14\sqrt{aA},$$

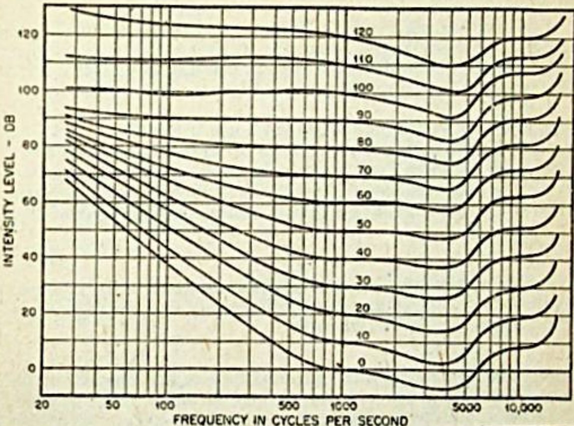


Figure 1. Curves of equal loudness level for pure tones in frontal sound field, according to ISO Recommendation 226. These curves show how frequency response of the human ear varies with loudness.

where a' is the absorption coefficient of the walls and A is the surface area of the walls, floor, and ceiling. In an average factory a' is between 0.05 and 0.2. In normal rooms a' is between 0.1 and 0.3. The change from a directional or plane field to a diffuse field can be considered to occur at a distance Rg from the sound source.

In loudness measurements two types of field are usually considered. One is the diffuse field. The other is a plane sound field which approaches the hearer from the front, head on; this field is called a frontal sound field.

Since loudness is a subjective quantity the primary instrument for measuring it can only be a human observer. To determine whether one sound is louder, equally loud, or less loud than another, we would have to let a statistically significant number of people compare the sounds and then average their opinions. Similarly, to determine how loud a sound is, we would have to choose a standard sound and have a significant number of people compare the unknown with the standard.

In acoustics the accepted standard is a pure 1KHz tone or narrow-band noise centred at 1KHz. The loudness level of any sound is defined as the sound pressure level of a standard sound which appears to a significant number of observers to be as loud as the unknown. Loudness level is measured in phons, the loudness level of any sound in phons being equal to the sound pressure level in dB of an equally loud standard sound. Thus a sound which is judged to be as loud as a 40dB 1KHz tone has a loudness level L=40 phons.

Although the logarithmic phon scale covers the large dynamic range of the ear (120dB) conveniently, it does not fit a subjective loudness scale. A factor of two in loudness does not correspond to double the number of phons. Over most of the audible range, that is, for loudness levels of 40 phons and greater, the corresponding increment is 10 phons. This is an empirical fact; why loudness should be different from physical quantities like voltage, for which a factor of two corresponds to 6dB, is not fully understood.

It is also difficult to add loudnesses in phons. If, for instance, we produce one tone at 200Hz with a loudness level of 70 phons, and another at 4KHz with the same loudness level, it would be convenient if both tones together would yield a loudness level of 140 phons. Unfortunately, this doesn't happen. The two tones actually are perceived as a loudness level of 80 phons.

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* One μbar equals one dyne per square centimetre or 0.1 newtons per square meter. Human speech at a distance of one metre generates a sound pressure of about one μbar.

In an effort to obtain a quantity proportional to the intensity of the loudness sensation, a loudness scale was defined in which the unit of loudness is called a sone. One sone corresponds to a loudness level of 40 phons. For loudness levels of 40 phons or greater, the relationship between the numerical values of loudness level L (in phons) and loudness S (in sones) is given by

$$S = 2^{(L-40)/10} \qquad (1)$$

(ISO Recommendation R 131).

Table 1 compares the loudnesses (sones) and loudness levels (phons) of several common sounds. Notice that the loudness scale in sones corresponds fairly closely to our subjective sensation of loudness. We feel, as a matter of experience, that a speaker in an auditorium speaks about four times as loudly as someone who talks quietly with us in normal conversation. It is more meaningful to state that a jet aircraft at takeoff is about 50 times as loud as our conversation than to state that the jet aircraft generates 120 phons in contrast to 60 phons generated in ordinary conversation

TABLE 1

Loudness Level (phons)		Loudness (sones)
140	Threshold of pain	1024
120	Jet aircraft	256
100	Truck	64
80	Orator	16
60	Low conversation	4
40	Quiet room	1
20	Rustling of leaves	
3	Hearing threshold	

FREQUENCY RESPONSE RANGE OF SOUND REPRODUCTION

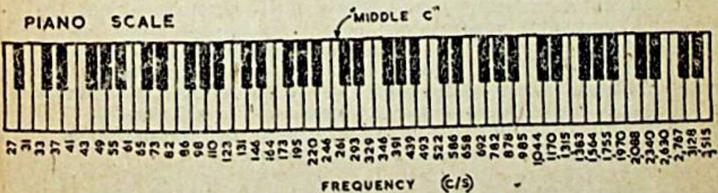
Average Quality Hi-Fi	30 to 16000 C/s
Bass speaker Average Quality in enclosure	35 to 4000 C/s
Bass speaker Finest Quality in Suitable enclosure	18 to 3000 C/s
Domestic Quality Record Player	180 to 7500 C/s
Domestic Quality Tape Recorder	160 to 7000 C/s
Earliest Gramophone with sound box	400 to 4500 C/s
Electrostatic Wide Range Speaker	300 to 20000 C/s
Finest Hi-Fi Equipment	15 to 22000 C/s
High Quality Magnetic Recording Tape	40 to 16000 C/s
High Quality Tape Recorder	40 to 16000 C/s
Human voice adult speech	90 to 1300 C/s
Inexpensive Transistor Set	350 to 6000 C/s
LP $33\frac{1}{3}$ r.p.m. record	42 to 11000 C/s
Middle Range Speaker	300 to 6000 C/s
Normal AM Table Radio	100 to 8000 C/s
Percussion Instruments	40 to 180 C/s
Portable Transistor Set High Quality	100 to 10000 C/s
Range of Human Hearing	15 to 18000 C/s
String Instruments	40 to 3200 C/s
Telephone	375 to 2500 C/s
Tone Control Bass Range	20 to 200 C/s
Tone Control Presence Range	1000 to 6000 C/s
Tone Control for Rumble Filter	18 to 60 C/s
Tone Control Treble Range	5000 to 20000 C/s
Tweeter High Quality	2000 to 18000 C/s
Tweeter Inexpensive Grade	1600 to 15000 C/s
Wind Instruments	45 to 4500 C/s

VIBRATIONS AND THE MUSICAL SCALE

Ratio of vibrations of 1 octave in any part of the musical scale

Note	C	D	E	F	G	A	B	C
Ratio	1	9/8	5/4	4/3	3/2	5/3	15/8	2
Decimal Ratio	1	1.125	1.25	1.33	1.5	1.66	1.875	2
Tonic Sol Fa Scale	Doh	Ray	Me	Fah	Soh	Lah	Te	Doh

Piano scale showing the frequencies to which the keys are usually tuned, which is to a slightly different pitch from that used by physicists, based on Middle C = 256 c/s., and such scales are apt to be misleading. Frequencies of black keys can be obtained by multiplying the frequency of the white key below it by 1,05946. This scale is useful for the approximate calibration of oscillators and rough determination of resonant frequencies etc.



The loudness level of a 1KHz tone is the same as its sound pressure level. This would also be true of pure tones of other frequencies if perception were constant with frequency. However, it is not. The loudness level of any other sound (in phons) is not, in general, equal to its sound pressure level (in dB). For example, if a large number of observers compare a 100Hz tone with a 1KHz tone, they will judge the two to be equally loud only when the 100Hz tone has a higher sound pressure level than the 1KHz tone. The frequency response of the ear is not flat.

Although the subjective sensation of loudness differs from person to person, normal ears seem to agree within a few dB, at least for the young male subjects who have participated in most subjective tests. Hence it is possible to draw curves or contours of equal loudness level for normal ears, as shown in figure 1.

Equal loudness level contours were first published in 1933 by Fletcher and Munson. The slightly modified form of their curves shown in figure 1 is now universally accepted as reference data (ISO Recommendation 226). The curves of figure are for pure tones in a frontal sound field. They show, for example, that a 40 phon 100Hz tone has a sound pressure level of 50dB, but an equally loud 40 phon 1KHz tone has a sound pressure level of only 40dB. The 3 phon curve is just above the threshold of hearing for normal ears.

Notice that the curves converge at low frequencies, but are approximately parallel between 1 and 10KHz. This means that the ear's frequency response is a function not only of frequency but also of level. Therefore it can be simulated only with networks which are nonlinear with respect to both frequency and amplitude.

Curves of equal loudness level for a diffuse sound field can't be measured using pure tones, because it is difficult to set up a diffuse field using pure tones. Pure tones are likely to bounce off walls and nearby objects and produce standing-wave pattersn, whereas sound in a diffuse field is supposed to be uniform in all directions. However, diffuse-field loudness comparison can be carried out with consistent results using frequency-modulated tones or narrow noise band.

Differences in sound pressure levels necessary to give the same sensation of loudness in a diffuse field as in a plane field were standardised in ISO Recommendation 454. Using these differences (see figure 2), curves of equal loudness level for the diffuse sound field can be calculated from those for the plane field.

To human ears, broad band sounds, like those of jet aircraft, seem much louder than pure tones or narrow band noise having the same sound pressure level. Figure 3 illustrates this effect for band-limited noise having a centre frequency of 1KHz. Figure 3(a) is a series of sound intensity density spectra for bandwidths of 100Hz, 160Hz, and 200Hz. All three spectra have the same area, so all three noises have the same sound intensity (sound power per unit area). This means that all three noises have the same sound pressure level. But all three noises are not equally loud.

If the loudness of the noise which has 100Hz bandwidth is So, then the loudness of the noise which has 160Hz bandwidth is also So. But the loudness of the noise which has 200Hz bandwidth is greater than So.

Fig. 3(b) shows what increasing bandwidth does to the loudness of noise having a centre frequency of 1KHz and a constant sound pressure level of 600dB. Up to a critical bandwidth of 160Hz, the subjective loudness is constant. Beyond that point, however, there is a marked increase in loudness. At a bandwidth of 2KHz the loudness level L has increased from 60 phons to 74 phons. Loudness S has increased by a factor of 2.5.

Similar investigations, using different centre frequencies, yield different critical bandwidths. At a centre frequency of 200Hz the critical bandwidth is approximately 100Hz. At 5KHz it is about 1KHz.

We account for the effect of bandwidth on loudness with any broadband measurement. Accurate loudness measurements can be made only by taking into account the spectral distributions of sounds being analysed. The necessary degree of resolution in the spectrum analysis is clear from figure 3(b).

The human ear's critical bands seem to be related to another property of the ear, namely, subjective pitch. Subjective pitch tells us how our ears compare the frequencies of different sounds.

If an average untrained observer – not a musician or a piano tuner – were first allowed to listen to two tones, say a very-low-frequency tone and a 4KHz tone, and then were asked to tune an oscillator until he heard a tone that fell exactly half way between the first two tones, he would not pick something around 2KHz. Instead, he would pick a tone having a frequency of about 1KHz. In subjective pitch, then, 1KHz is halfway between 0 and 4KHz. The unit of subjective pitch is the mel; 0 to 2400 mel span the frequency range 0 to 16KHz.

Remarkably enough, it turns out that a subjective pitch interval of approximately 100 mel located anywhere in the audio range corresponds to the width of a critical band at that point! Probably, the same mechanism in the ear is responsible both for critical bands and for subjective pitch. However, our understanding of the ear is still not good enough to allow us to identify this mechanism.

In loudness measurements, the frequency scale most commonly used is linear in subjective pitch z. However, the mel is not used. Instead, the width of a critical band is defined as one Bark. Accordingly, the audio range comprises 24 Bark. Figure 4 shows how subjective pitch, in Bark, is related to frequency.

Two sounds presented to the ear simultaneously produce a sensation of loudness which is larger than that produced by either of them alone. Take for example, a 200KHz tone having a loudness level of 70 phons and a 4KHz tone, also having a loudness of 70 phons. If two sounds are as widely separated in frequency as these two, their partial loudnesses simply add to form the total loudness. The loudness corresponding to a loudness level of 70 phons is 8 sones. If two partial loudnesses of 8 sones each occur simultanously the total loudness is 16 sones, and the loudness level is 80 phons.

This simple summation of partial loudness can only be carried out if the individual sounds are separated widely in frequency. The closer they are in frequency the more they influence each other, and total loudness may not be quite as large as the sum of the partial loudnesses. This effect is called partial masking. In the extreme case, partial masking becomes total masking, wherein a strong sound renders a lower-level sound completely inaudible. When total masking occurs, low-level sound components cannot be heard at all and do not contribute to loudness.

The partial masking of tones cannot be understood in terms of level and frequency because pure tones represented by spectral lines cannot influence each other. Investigations on the ear have shown, however, that even pure tones or narrow-band noise excite nerves in the ear that correspond to a wide range of frequencies. Masking occurs because the ear treats sounds in an "OR" fashion – when two sounds excite the same nerve, the ear hears only the larger sound in that frequency range.

HOW COMPATIBLE IS COMPATIBLE?

Having for years operated on the basis of separate discs for stereo and mono, the record industry is now faced with the apparent enigma of "compatible" disc. What are these discs and how is it that manufacturers are now offering for sale what they said was impossible in the early days of stereo?

To cut the preamble and get back to fundamentals, the original mono LP record carried a groove which was intended to be played with a stylus having a nominal tip radius of 1mil, or one-thousandth part of an inch.

With modulation, the groove was displaced from side to side of its normal unmodulated position so that, when played, it imparted a side-to-side motion to the stylus tip.

As a basic requirement, mono pickups to play these records were designed primarily to exhibit high compliance and low tip mass in relation to side-to-side (i.e. horizontal) tip movement. Any compliance in terms of vertical movement was largely a by-product of their design and the ideas of their designer. Some mono pickups exhibited a fair amount of vertical tip compliance, others very little.

The 45/45 system of stereo recording–the system which was universally adopted–makes use of a vertical as well as a horizontal component of groove modulation and, as a basic requirement, stereo cartridges have to be so designed that the stylus exhibits high compliance and low tip mass in respect to both horizontal and vertical modulation. And naturally, to be of any use, the cartridge has to translate these movements into proper left and right channel signals.

As a refinement, groove shape was more closely controlled, especially near the bottom, to allow the use of stylus tips having a smaller radius. While 0.5mil was regarded as a desirable figure manufacturers tended to prefer a slightly larger figure – usually between 0.6 and 0.7mil. This larger "compromise" stylus was less liable to skate in old mono grooves with a heavily rounded bottom, and also presented a larger surface area to support the playing weight of the pickup.

With the appearance of stereo discs on the market, two questions have become more or less routine. The first is:

"Can I play existing mono records with my new stereo cartridge?"

In practical terms the answer is in the affirmative. Provided the cartridge is fitted with the compromise stylus (0.6 to 0.7mil) there is every chance that it will sit in the groove without touching the bottom and play the record well. In fact, because of the more ambitious design criteria for stereo cartridges, there is a good chance that a new cartridge will have better characteristics than the older mono cartridge which it replaces.

Some enthusiasts, with large collections of early mono discs did strike a certain amount of trouble with the finer styli "skating" on the bottom of "blunt" grooves but, by and large, most enthusiasts play mixtures of mono and stereo discs on stereo equipment without further thought.

The other question, of course is:

"Can I play stereo discs with my existing mono cartridge?"

To this question, two answers have commonly been given. One is "Yes, provided you fit it with a stereo stylus–0.7mil instead of 1mil".

This answer has normally been deprecated as worthy only of non-technical salesmen, who could not be expected to know that it was wrong!

And, in terms of accepted theory it is wrong! While it observes the difference in stylus dimension, it completely ignores the requirement that the cartridge must be able to track vertical groove modulation, even if it does not put the vertical "information" to any practical use.

The approved answer has long been:

"Don't take the risk. Unless your mono cartridge just happens to have adequate vertical compliance, playing stereo discs with it will scour the vertical deviations and ruin the disc for future stereo playing.

Record companies have long supported this stand by carrying warnings on their stereo discs that they must be played only with stereo pickups.

This has meant that dealers throughout the world have had to handle mono and stereo versions of the same albums, a situation which has been unwelcome but seemingly inevitable.

Initially, stereo versions were in minority demand, but the position has gradually been changing so that, in the realm of long-playing albums, mono has become the poor relation.

There is more to this than a simple reversal of priorities. A few years ago, the urge to keep a secondary stock of stereo pressings was strong, because stereo was new and up-coming. But a secondary stock of mono pressings lacks this advantage; mono is old and seemingly obsolescent!

It is of little wonder that, in the face of heavy competition, some manufacturers, distributors and retailers have been looking hard at the economies which might be effected by eliminating the duplication of titles and producing discs which can be played in either mode.

Overseas comment on the proposition suggests that three major producer areas reflect different lines of thinking.

The British attitude, as expressed by such notables as Arthur Haddy and Dr G.F. Dutton is politely but firmly against the idea of compatibility. If mono is wanted, let it be the best mono; if stereo, let it be the best stereo–each without the compromises that compatibility would mean.

The American attitude, as summed up by Norman Eisenberg in "High Fidelity" is: "Egh! which translates to why bother? or who needs it? or why make life easy for the dealers; we all have problems!"

The present stronghold of compatibility is Europe where many new covers are displaying the lines: "Stereo–auch mono abspielbar" or "Stereo–peut etre joue en mono".

For the English market the endorsement is more likely to read: Compatible stereo/mono. This is a full stereo recording, but it has been specially engineered so that it can also be played on a mono system, without suffering damage.

It all leads to the vital question: Is there such a thing as truly compatible, no-compromise stereo record, which can be played without damage by any existing mono microgroove cartridge?

To this might well be added another question: "Is it possible to produce a record containing stereo information, but for mono playing, which will sound just as good as a purely mono equivalent?"

On completely purist grounds, the answers to both questions are probably in the negative but the purist answer is not necessarily the most practical one, commercially. One should at least consider and listen to some of the records which are being sold, right now in this country.

What then might be involved in producing compatible records?

To begin with, groove depth and dimensions would have to be kept large enough to hold mono styli (together with any likely "plus" tolerance) between the groove shoulders. While not a major consideration, the requirement would limit the degree which grooves could be packed together to secure longer than usual playing time.

Within the groove itself, it would seem obvious that compatibility would require a limitation on the amount of vertical modulation, to ease the problem of tracking, without catastrophic damage, with a mono cartridge.

This might, for example, be achieved to a degree by limiting either the overall amount of groove modulation or simply the amplitude of loud passages. In the one case, it would be at the expense of signal-to-noise ratio; in the other of dynamic range.

A further possibility would be a deliberate manipulation of the content to relieve either channel of the responsibility of handling heavy passages, particularly heavy brass passages, on its own, By making sure that such signal is shared substantially by both channels, in phase, the weight of modulation would assume a horizontal (mono) character, with the lighter instruments and overtones exploiting the vertical (stereo) to lend a sense of direction.

Commenting on this technique, Norman Eisenberg (mentioned earlier) has this to say:

"This approach ties in nicely with the so-called 'MS' method of stereo microphoning long in favour on the Continent. 'MS' stands for 'middle-side'– a term that derives from the typical pickup patterns of the mikes employed. This setup senses left and right groupings, but with a strong emphasis on the centre-mix signal.

"The ambience of the place – the 'room effects' or reverberation – is suffused subtly with the sound itself to further tonedown any 'extreme' stereo effects. An appreciable amount of the sound to be recorded is about evenly split between the two channels and, if due attention is paid to phase relationships – which, we are assured, it is – the channels can be combined for an acceptable mono version.

"The deepest bass, in such a setup, is handled in one of two ways. If the heavy bass choirs (string bass, tuba, and perhaps the heavy end of the percussion battery) are seated predominantly to one side of the orchestra their sonic output becomes mixed with the hall's reverberant sound (mentioned earlier) so that a great deal of its directionality is suffused with the over-all ambience. As a result, much of the sound appears as a mono signal that will not make great demands on a pickup's vertical response.

"If the heavy brass happens to be centred at the rear (as it sometimes is in the seating plan of European ensembles), so much the easier for the compatible approach: the most demanding sonic passages are then 'naturally' split between left and right sides and can be recorded as is.

"The result, in either case, is bass virtually in mono–that is, on stereo playback, it will be just about centred between the two speakers.

"At its best, this recording technique can make for a thoroughly enjoyable sound – warm bass, full midrange, and clean "well-rounded" highs. According to the opposition, however, it is not suited for full-impact sound, for the 'sonic spectacular', for the most dramatic kind of stereo spread that a no-compromise recording setup can yield.

"Be that as it may, it is essentially the technique that many European recording outfits have been using all along."

It would appear that this dependence on microphone choice and placement is behind many of the compatible records which are currently on sale or planned for sale in Australia. Reportedly, the masters are being cut locally on standard stereo recorders, fed from master tapes "which already contain the compatible signal."

To what extent this may differ from the original stereo tape is a moot point.

On the other hand, the emergence of techniques for sensing and limiting waveform conditions which would embarrass playback styli would suggest that electronic circuits could assume much of the responsibility for limiting vertical modulation to any predetermined parameters.

In fact, it is possible to visualise a system which would produce a "compatible" track with normal stereo characteristics, except for individual wave-trains where the limiting provision would modify the recording towards mono characteristics.

It would be a kind of "now stereo, now mono" recording, which exhibited good separation most of the time but cheated on critical passages. Since this would be achieved in the cutter amplifier system, such a recording could be contrived from any master tape.

On the practical side, the writer reviewed two "synchro-stereo" discs from the Concert Hall Record Club in the April issue, last (page 119). Judged as stereo records, these were given high marks for quality and stereo characteristics, with a special mention of long playing time! The one observation was that the heavier bass components seemed to be concentrated towards the centre, which is in line with this discussion.

It also lines up with remarks on such discs made in "High Fidelity Magazine".

"Meanwhile, compatible discs from several companies are on the market and a few general comments may be in order. The stereo effect of these records is on the whole more subtle than that of typical releases from the big British or American companies. It is more a matter of suggesting air and space than of fully documenting left-to-right spread of the ensemble.

"One excellent way to judge left-right-and-centre signal "weight" is to listen via headphones, switching the amplifier from mono to stereo. While on compatible recordings the sounds of instruments whose outputs are predominantly in the midrange (for instance, trumpets and violins) take on a stereo spread, the bass choirs (and on pops, most of the rhythm section) remain dead-centre.

"The difference between mono and stereo is even less apparent when one listens over loudspeakers."

Against this, other compatible records which have been observed by the writer and fellow reviewer Harry Tyrer have been seemingly quite normal in their stereo emphasis. They must obviously have contained a generous helping of vertical component in the signal, to sound that way.

It is difficult, in the face of this, to be sure how much of the so-called compatibility is attributable to technical achievement, how much to deliberate compromise or how much to mere words on the jacket.

According to reports, the German record companies found that, out of some 4.5 million pickups, now in use, only 400,000 were so "strictly mono" that they could not track normal stereo discs; further, that all the mono pickups manufactured in the country since 1960 had been designed with inbuilt vertical compliance and .7mil styli.

On these figures, only 8 per cent of all German pickups could not play stereo discs and of these:

*Some would ride the shoulders of grooves leaving the lower walls in reasonably good shape for later playing with a new stereo cartridge;
*Some would be accommodated by a modest compromise in the recording practice;
*Some might damage the stereo information on records which would never be played critically on stereo anyway.

In short, the odds for taking the risk look pretty favourable.

Apparently, the major record companies outside continental Europe are unwilling to adopt techniques which would limit their exploitation of the full stereo potential, nor are they convinced that the so-called compatible pressings are as proof against damage by mono pickups as their sponsors claim.

Their present attitude is to let mono die at its own rate and to encourage its residual adherents to re-equip with good quality stereo cartridges, connected for mono output. This way, the full stereo facility will be preserved both for present-day users and for mono supporters themselves, when they re-equip fully to play their rising collection of stereo records.

This thinking is fine from the viewpoint of the stereo purist and those with a vested interest in re-equipment.

It is not so good for the enthusiast limited to mono reproduction, because at least some of the current "no holds barred" type of stereo discs can lose far more than mere direction by being played in mono mode.

It is, in fact, an interesting exercise to switch a system to mono mode, from time to time, when playing stereo discs. In some cases there is little change in the general balance–just a concentration of the sound origin to the zone between the stereo loudspeakers. In other cases, the whole "body" seems to drop out of the reproduction.

Presumably, in a genuine mono pressing of the same performance, the signals from the multi-track tape master would have been mixed and balanced differently to obtain maximum sonic impact.

How all this will work out in the next couple of years is hard to say. To stereo users, it may not amount to a great deal because, overall, they will probably continue to get much the same range of records and recording techniques as they have in the past, endorsements notwithstanding.

The real rub is to collectors of mono LP albums, who emerge as the poor relations, facing three somewhat disagreeable propositions:

*A diminishing supply of genuine mono records.
*Compatible records which some claim are still prone to damage.
*Discarding what might have been an expensive mono pickup to replace it with a stereo counterpart, used in mono mode.

MORE ABOUT ACOUSTIC FEEDBACK

Last month we introduced the subject of acoustic feedback but only got so far as to talk about the problem in relation to public address amplifiers. It left untouched three other aspects which we mentioned in the introduction – acoustic feedback in electric guitar amplifiers, in short-wave receivers and in home record playing equipment.

The problem of acoustic feedback in electric guitars is one which has been raised quite frequently of late in readers' correspondence and, in particular, since we featured a series of electric guitar amplifiers.

While many readers have constructed these amplifiers successfully, a few have run into trouble: The amplifier systems rumble or roar, whenever they are turned up to the level needed for onstage entertainment. In most cases, the constructors have jumped to the conclusion that the amplifier designs have been faulty in some way.

In fact, the trouble has turned out, almost invariably to be–you've guessed it–acoustic feedback, closely paralleling the situation discussed last month in relation to public address amplifiers.

It generally transpires that the guitar which they are trying to use is an acoustic model which also carries a pickup unit to feed an electronic amplifier. The instrument can be played with or without amplifier, depending on whether the additional volume and/or control effects are required.

Being basically an acoustic instrument, it has the normal thin, hollow body specifically designed to radiate into the surrounding air, as direct sound, the vibrations of the strings.

Unfortunately, from the present viewpoint, the process works all too well in reverse, the thin hollow body picking up sound from the surrounding air and feeding it back into the strings! When these vibrate, as a result, they induce signal voltage and current in the electronic pickup unit, the guitar therefore behaving as a kind of microphone.

The more efficient it is as an acoustic instrument, the more efficient is it likely to be in the reverse role!

When such an instrument is coupled to an amplifier and loudspeaker system, trouble is just around the corner.

The guitarist turns up the amplifier gain and strikes an introductory chord. An amplified version of the chord radiates from the loudspeaker as sound waves. These strike the body of the guitar, causing it to vibrate; this vibrates the strings, inducing a signal in the pickup units.

This drives the amplifier, which drives the loudspeaker, which excites the guitar body, which vibrates the strings, which generates a signal, which drives the amplifier . . . and so on.

Before the guitarist knows what's happened, the whole system has begun to sing or rumble or roar, depending on the frequency where the highest gain is evident. And, in a guitar, with deliberately tuned strings, there is no lack of resonance peaks.

The fundamental way around the problem of acoustic feedback in a guitar is to use a solid-body instrument which is incapable of radiating significant sound except through an associated amplifier. Conversely it will be equally unresponsive to incoming sound and therefore to feedback effects.

If circumstances dictate the use of an acoustic instrument, the guitarist is in much the same position as a public address operator. He must accept some order of feedback as inevitable but strive to limit it to a level where its effects are not noticeable.

*He may have to keep the loudspeaker well away from the guitar, even though this may be contrary to the desire, from a musical standpoint, to have it close by.
*He may have to limit the gain of the system, making up volume by vigorous fingering, when he would like to reverse the relationship.
*He may have to be content with a much lower output, in terms of watts, than the amplifier is capable of delivering.

So, he finds himself using just as much gain and power as he can, from a loudspeaker as closely placed as he can, coaxing the controls and tilting the guitar to try to minimise the howling.

The whole point which we are trying to make here is that his problem is not basically one of amplifier design, even though differences in amplifier response may have a second-order effect on the trouble.

His real and basic problem is acoustic feedback—sound waves from the loudspeaker striking the body of an acoustic instrument and feeding them as electrical signals back into the amplifier.

There is an element of chance about whether or not acoustic feedback will cause trouble in an individual case but, while ever the basic problem is present, there can be no real grounds for surprise if trouble does arise as a result.

To be sure, the lot of the electric guitar specialist would be easier if he did not have to worry about acoustic feedback.

The reference to acoustic feedback in short-wave receivers is admittedly out of place in an audio column but we decided to retain it because its effect will certainly not be unknown to many readers, whether or not they recognise the cause.

It works this way:

A shortwave receiver appears to be perfectly normal and stable in its behaviour, until it is tuned to an incoming signal. Then, when the volume is advanced to bring the signal to a good level from the loudspeaker, the set begins to howl, ceasing to do so immediately it is detuned or the volume is retarded.

This kind of trouble can have a purely electrical origin, when current drawn by the power output stage modulates the high tension supply and ultimately affects the receiver's oscillator supply. This can produce the rather ludicrous but very serious situation that, each time an incoming signal is reproduced through the audio system, feedback through the power supply to the oscillater detunes the carrier by which it is arriving.

When the audio signal disappears as a result of the carrier being detuned, the carrier immediately reappears in the IF system and detector, producing a new segment of audio.

This now-you're-here now-you're-not action of the signal, translated into more technical terms, produces a sound that can be anything from a "motor boating" plop to a rumble or a roar.

The normal cure for this kind of trouble is to decouple the oscillator supply voltage more effectively or, in an extreme case, to operate the oscillator from a regulated voltage network.

Acoustic feedback in a shortwave receiver produces much the same audible effect, though it is more likely to be higher in frequency, in the nature of a howl.

COMMON CABINET

Normally, it happens only when the loudspeaker is mounted in the same cabinet as the rest of the set though in an extreme case, it can happen when the loudspeaker is merely close to the set.

The culprit is usually one or other of the plate assemblies of the oscillator tuning capacitor. If these are loose, or under some kind of compression, they may show a tendency to vibrate at a natural resonance of a hundred or few hundred cycles per second. In vibrating, they may modulate or detune the oscillator and therefore any signal which it is heterodyning down to the intermediate frequency.

If the receiver's loudspeaker is mounted sufficiently close to the suspect capacitor, the latter can be caused to vibrate continuously at its own natural mechanical resonance. This produces a frequency modulation of the receiver's oscillator, and of the signal fed through to the IF channel and detector. The end result of this interaction or feedback between the audio output and the local oscillator is a tendency to howl or rumble whenever a signal is tuned in.

The magnitude of the trouble increases with the order of change from signal to intermediate frequency, which is the reason why it is worse in short-wave receivers than in broadcast receivers.

It also becomes more serious as the selectivity of the IF channel is increased, because frequency shift of the oscillator and resulting IF signal produces proportionately more effect at the detector and a greater spurious audio output in the way of a rumble or howl.

While vibration of the oscillator tuning capacitor has been singled out as the most likely cause of acoustic feedback in a shortwave receiver, any other component in the oscillator circuit capable of affecting the oscillator frequency, can have the same effect.

"Microphonic" oscillator valves are a quite classic source of trouble, for example.

Acoustic feedback can be picked from electrical feedback (via the high tension line, etc.) by the simple expedient of operating the receiver into a loudspeaker placed well away from the chassis.

In practice, to cure the trouble is often a lot harder than to find it.

*A capacitor with thick plates and a heavy frame will usually show less sensitivity to acoustic feedback than one with thin plates or split rotor plates, or a flimsy frame.
*Mounting the capacitor on rubber or spring supports is another useful trick.
*Again it may be possible to mount the speaker flexibly to the chassis or cabinet rather than rigidly, thus at least minimising direct vibration.

Once again, however, the problem is clear—to prevent acoustic or mechanical vibration from the speaker from affecting the receiver's local oscillator.

AUDIO SYSTEMS
Coming to the last major section of the story, acoustic and mechanical feedback are twin and often inseparable villains in domestic record playing equipment. They are responsible for many of the incipient—and active—rumbles and roars about which record enthusiasts complain from time to time.

Even in these days of stereo the majority of record playing equipments are housed in a single cabinet. In a convenient position near the front or top is the record playing deck or changer; elsewhere are one or more speakers, while the amplifier, radio tuner, etc., are stowed wherever vacant space remains.

When such a unit is in operation, sound waves from the speaker tend to set up vibrations in the various sections of the cabinet, aided and abetted by direct mechanical vibration from the speaker frame.

These vibrations must inevitably be communicated to the playing deck and there is every chance that they will vibrate the tip of the stylus, in relation to the pickup cartridge and arm, when the stylus is resting in a record groove. Such vibrationary motion must generate a corresponding signal voltage, which is fed to the amplifier and thence back to the speaker, where it originated.

In short, and once again, we have a complete feedback loop.

The degree to which the feedback may be evident is dependent on many factors – the gain of the amplifier for normal playing, the power and frequency response available from the amplifier and loudspeaker, the rigidity or otherwise, and mass of the cabinet, the structural lines along which vibration might travel, the isolation of the playing deck from the cabinet structure and the overall characteristics of the pickup arm and cartridge.

Acoustic and mechanical feedback effects in a record player are usually most evident at the low frequency end of the range; hence the earlier reference to rumble and roar.

If the feedback is evident but only to a minor degree, its usual effect is to aggravate any low frequency imperfections in the system. One could explain this by suggesting that the player is "regenerative" at the low frequency end.

Thus, rumble in the playing deck or in a particular record is made to sound louder than it would otherwise be. Footsteps on the floor or bumps on the cabinet build up into annoying magnitude. And the higher the level at which the system is operated, the worse the effects become.

If the feedback gets beyond this stage, it can build up into a sustained roar whenever the gain control is advanced, or if the system is shocked into active oscillation by a heavy bass passage in the music. Often the rumble can build to quite alarming proportions.

The widespread swing to stereo has, in general, increased the seriousness of the problem although, fortunately, it is also better understood than it was. One basic reason for the increased difficulty is that a stereo pickup has to be made sensitive to vertical movements of the stylus as well as lateral, so that there is an additional plane in which vibration effects can produce feedback.

In addition, there are two speakers and two amplifiers to be reckoned with, instead of one.

Feedback effects can be very troublesome in portable and similar small player cabinets because of their restricted size, the proximity of pickup and speaker and the necessary lightness of cabinet construction.

One of the important measures towards stability is having the whole playing deck mounted, not rigidly to the cabinet, but on springs or active rubber bushes. This helps to block feedback due to vibration, while a certain amount of isolation and damping for sound waves can be provided within the cabinet.

However, one of the most significant reasons why the average small player is stable is in the deliberate and accidental restriction of the bass response. This starts with the pickup cartridge and its loading, includes the rather diminutive output transformer usually fitted and concludes with a loudspeaker system of strictly limited capabilities.

If steps were taken to alter this situation and greatly increase the unit's performance at the bass end, mechanical and acoustic feedback would very quickly become evident.

In fact, the feedback problem follows, almost automatically, hard on the heels of increased cabinet size, from the small portable player, through the table model to the floor-standing console.

As the cabinet size is increased, there is a natural tendency to boost the power and performance of the amplifier; even if rather modestly, and to use a rather better speaker system – with two inbuilt speakers in the case of a stereogram.

The improved bass performance increases the feedback potential and largely offsets the greater isolation available in the large cabinet. A manufacturer or an individual constructor has to be only slightly unfortunate for a feedback howl to appear in a particular player, when the gain control is advanced towards maximum playing level.

How small the margin can be is often discovered if a deliberate attempt is made to achieve something like true hi-fi standards in a single cabinet installation. This would normally involve selection and compensation of a pickup for full bass response down to 30 or 35 cps; the provision of modestly powerful amplifiers, complete with wide range output transformers; tone control facilities including bass boost; the use of medium to high-grade loudspeakers and the provision of more adequate baffling – all within a single unit cabinet.

In such circumstances acoustic and mechanical feedback, leading to rumbling and howling is not the exception, it is the rule. It is positively difficult to build a fully stable, high performance, single-unit stereogram.

If success is to be achieved, it is likely to involve measures such as the following:

Elaborate spring mounting of the player deck. What is normally required is a much higher degree of resilience in the spring system than is normally provided, coupled with extra mass in the suspended system. The mass may occur automatically by reason of a heavy bass plate and turntable fitted to some high quality players; otherwise the mass may have to be added, by way of lead blocks distributed under the playing deck. The elaborate springing would have to be devised by the individual constructor or manufacturer but the aim would be to have the whole playing deck and pickup system suspended quite "sloppily" within the cabinet.

Dynamically balanced pickup arm. As we pointed out in an earlier article, a pickup arm which is fully dynamically balanced is much more proof against external excitation that one which is not. Unless the user is prepared to tackle the job of reconstructing a pickup arm, this requirement involves buying the right type to begin with.

Some compromise in low frequency performance. It is nice to be able to boast full output to 20cps but very little music gets into that region. More practically, and with little sacrifice in the sound as heard, deliberate attenuation might be allowed of frequencies below about 45cps.

Semi isolation of the cabinet sections. It is likely to be most helpful if the cabinet can be built of separate sections stood together to look like a single unit, rather than built as a single unit. Thus the speaker enclosures may merely stand on either side of the player section or the player may rest on top of the horizontally disposed speaker system, but separated by plastic foam or springs. Note that heavy construction or bracing is not, of itself, any guarantee of freedom from feedback effects; isolation is the best measure.

One might assume, from all this, that feedback effects will disappear entirely if separate cabinets are used in a room.

Unfortunately, this is not the case, for feedback effects are not uncommon in very wide range, very powerful player systems disposed in the one room. Energy still seems to get back from speaker to pickup, causing aggravation of rumbling, footfalls, suface plops and so on – right through to active oscillation.

LIKELY CAUSES

If you strike this trouble, things to look for include:

*A sympathetic resonance between the loudspeaker system and the cabinet structure housing the player.
*A sub-audio peak in the amplifier/speaker system, due to rotation of the amplifier's negative feedback system.
*A springy floor providing direct coupling between the speaker and player cabinets.
*A sub-audio peak in the pickup system, due to resonance involving arm mass and cartidge compliance.

Whether or not these problems can be dealt with depends on circumstances and the individual, but preventative measures can sometimes get the individual out of trouble, as distinct from "fundamental" cures.

*If the system has a rumble filter, or one can be added, to provide attenuation below 30cps, the trouble may disappear without perceptible loss of musical quality.
*The player cabinet or shelf may be hung from a wall rather than stood on the floor.
*Elaborate spring suspension may be provided for the player deck inside its cabinet, as already discussed for a single cabinet installation.

One important point should be made in this latter conclusion: The pickup base should be rigidly attached to the playing deck and both spring mounted as a single unit. They should not be mounted separately, since movement of the turntable relative to the pickup base can only aggravate mechanical and acoustic feedback troubles.

Yes, life would be much easier for the hi-fi enthusiast, if he didn't have to cope with these twin villains. At least, if you've had to do so, you shouldn't feel too badly about it; you're in good company.

WHAT'S ALL THIS ABOUT HI-FI?

What is high fidelity and how does it relate to stereo? In this article, the author seeks to answer such fundamental questions – not so much for established readers of this journal, but for their friends who may well come seeking such information.

The quest for improved quality in sound reproduction is virtually as old as the art itself. Early "wireless" magazines were liberally sprinkled with advertisements for valves, transformers, loudspeakers and other components, stressing their ability to produce "louder signals", "clearer signals", "purer tone", "more natural speech", "lower distortion", and so on.

It might be added that such objectives were both understandable and commendable because radio receivers of the day, with their included amplifying circuitry, were so lacking in these basic qualities that the reproduced music was often positively unpleasant, and reproduced speech barely intelligible. Better quality of reproduction was not an option, but a necessity.

Around about 1930, the word "fidelity" began to find increasing use as the appropriate and collective term to describe the faithfulness with which receiving and amplifying equipment could reproduce the original sound – at least to the satisfaction of those concerned at the time!

Furthermore, it became fairly commonplace for manufacturers to include in their range special "high fidelity" products, which could be expected to contribute to a standard of sound reproduction better than from the then average product. A loudspeaker manufacturer would typically offer a range of ordinary loudspeakers, of various shapes and sizes, intended for use in ordinary radio and amplifier systems; but at the top of the range would be one or more premium quality units–high fidelity loudspeakers boasting better all-round performance, and priced a good deal higher than the rest.

The same applied to gramophone pickups, interstage and output transformers and other key components, as well as to complete receivers and amplifiers built around these components.

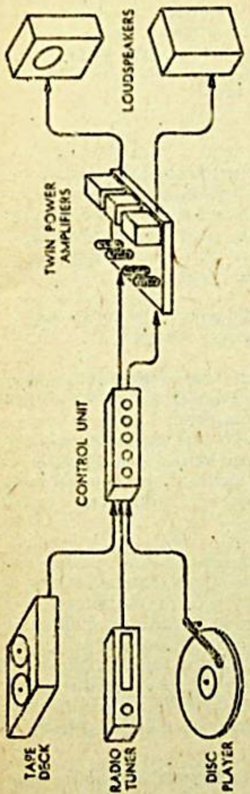
Initially the term "high fidelity" was fairly widely respected throughout the industry and one could reasonably assume that anything so branded would be a premium quality item, capable of better performance than the then average.

Circumstances have changed, however. Over the past 10 or 15 years, in their efforts to attract buyers, sales and publicity people have pressed into service every possible eulogistic phrase. The term "high fidelity" has been applied to quite ordinary radio and amplifier equipment so freely that it is close to becoming a noun meaning nothing more than "domestic sound reproducing equipment".

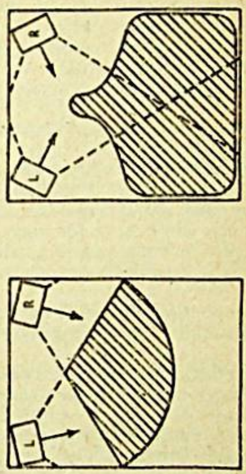
Hence the not-uncommon statement "I am thinking of buying myself a new hi-fi!"

In short, the term "high fidelity" has now lost any firm significance and the words, printed on a record label or scrolled in gold letters across the front of a reproducer, mean nothing in particular. The record, the tape, the component or the player may be very good, merely average or, in odd cases, quite poor–the description notwithstanding.

If abuse of the term "high fidelity" can be excused at all, it would be on the grounds that we are now able to obtain from quite ordinary tape, records and players a quality of reproduction that, all round, is more acceptable than was once available from expensive, high fidelity equipment. Intrusive background noise, unpleasant harshness and obvious lack of balance are—or should be—things of the past.



The majority of true high fidelity domestic amplifier systems are built up from separate units selected by the customer on the basis of performance and price.



Fundamental to the proper operation of a stereo system in the home is the ability to locate the loudspeakers for the best stereo effect.

Confusion about terminology, however, does not mean that there is any lessening of interest in what one might call true high fidelity equipment. On the contrary, any amount of equipment is available to the enthusiast of sound reproduction, which significantly outperforms ordinary production quality radio receivers, disc players and tape players.

As we have indicated, the real problem is to know what to call it.

Vendors of such equipment have to face the fact that the superlatives have already been appropriated and, in their publicity, they have to rely heavily on the fact that they are specialists in the field, that they offer a range of specialised and independent reproducing equipment and that they can explain and demonstrate advantages over everyday radiograms offered in the "furniture" stores.

Equally, enthusiasts and would-be enthusiasts have had to learn to discount the "high fidelity" label—and related claims—applied by non-technical salesmen to very ordinary receivers and amplifiers. By study, inquiry and observation, they have to gain some appreciation of the better quality sound reproducing equipment that is on offer from specialist suppliers and ultimately to choose equipment which represents the best compromise between their desires and their means.

Now what about this word "stereo"?

All early forms of sound reproduction used the so-called single-channel technique. Irrespective of the number of microphones involved in picking up the original sound, their electrical outputs were ultimately combined into a single audio signal for conveyance to the point of reproduction. The signal might be impressed on a radio carrier, in the groove of a disc or preserved as an optical or magnetic pattern on film or tape.

For reproduction, the signal was recovered and passed through a single-channel amplifier to a loudspeaker or loudspeaker system.

It was appreciated, very early in the history of sound reproduction, that while a single-channel signal could contain a composite of all the available sound impulses at any given instant, it was completely unable to communicate to the remote listener any information as to the relative location of these original impulses. This constituted a basic barrier to realism, but for many years, the problems of providing good quality single-channel sound were such as to discourage all but a few exploratory ventures into multi-channel reproduction.

As far as the man in the street was concerned, the real break-through in multi-channel reproduction came when disc manufacturers took up and successfully exploited a means of recording two distinct audio signals within the one record groove. Some variations in recording standards were involved but, superficially, the new "stereo" discs looked and played like the ordinary single-channel micro-groove recordings. The vital difference was that the groove now contained two slightly different versions of the original sound, in some ways comparable with the different versions one might unconsciously receive into individual ears, when listening to the original performance.

By recovering these two signals with a suitably designed pickup, passing them through separate amplifiers and into separate loudspeaker systems spaced a few feet apart, the missing sense of direction was impaired to the reproduced sound, adding tremendously to its ultimate realism.

And here a word about terminology: "Stereo" is a contraction of "stereophonic" which, in this context, indicates multi-channel sound reproduction. Virtually all present-day stereo equipment for the home relies on two channels of basic signal information but theatre and other commercial sound systems frequently provide more than two signal channels. The complementary term "mono" has been widely adopted to describe the older and conventional single-channel reproduction, being a contraction of "monophonic".

it is wise to avoid the words "monaural" and "binaural", which are sometimes used, quite incorrectly, to describe ordinary single and twin-channel reproduction. The two words really signify "one ear" and "two ears"— a quite different concept.

From the outset, acceptance of stereo reproduction in Australian homes was quite high. The position has now been reached where all but the least pretentious disc-playing equipment currently being offered for sale is twin-channel stereo, while twin-channel tape equipment is also freely available. The vast majority of new disc releases are stereo and the same is true of pre-recorded tapes.

Particularly during the early stages of the transition, there was a fair amount of discussion among enthusiasts about the relative merits of mono and stereo equipment. Particularly involved were those who had expended a good deal of money and effort installing high quality mono equipment and who, for various reasons, were disinclined to duplicate it to provide an extra channel. Out of this came arguments about the merits of hi-fi and stereo— as if they were rival systems.

As has already been explained, the basic idea behind the term high fidelity is—or was—reproduction of better-than-average quality. It is possible to have mono equipment of high fidelity or very mediocre fidelity. It is equally possible to have high, medium or low fidelity stereo.

Particularly confusing is the question that used to be asked fairly frequently: "What is best—good quality mono or poorer quality stereo?"

The question may have been vital enough for enthusiasts faced with the need to re-equip during the transition stage, but neither alternative can really be regarded as acceptable in the longer term.

No matter how well a mono signal is reproduced, it lacks certain vital information which is now available; therefore, in terms of present-day technology, it offers less than the available realism. On the other hand, the enthusiast can hardly regard as adequate a system which does provide dimensional information, but which is lamentably lacking in other respects.

At the present time, the objective of a true enthusiast of sound reproduction can only be a stereo system having the highest fidelity characteristics which he can afford and accommodate. If this means sacrifice and the need to reorganise domestic living space, it was ever thus—even in the days of high fidelity mono.

Well then, what attributes should one look for in a true high-fidelity sound-reproduction system?

Without at this stage becoming involved in figures, which could deter the non-technical reader, let us set down a few qualities.

FREQUENCY RESPONSE: A good high fidelity system should be capable of reproducing the full pitch range of musical sound from the deepest organ notes to harmonics on the upper fringe of audibility; this without noticeable, or at least intrusive, accentuation or attenuation of frequencies in any part of the spectrum.

DISTORTION: Used in a general sense, the word "distortion" could be made to include every minute detail in which the re-created sound, as heard, differs from that in the original environment. However, as normally applied to sound-reproducing equipment, and unless otherwise indicated, the word normally refers to spurious signal components which are generated within the equipment as a by-product of its operation, and added to the re-created sound. High-fiedelity equipment should not introduce audible distortion.

POWER OUTPUT: The level at which sound is to be re-created varies widely with the size of the listening room, the amount of noise created by the listening audience, the ideas of the audience in regard to the sound level required, and the nature of the sound itself. For an equipment to qualify as high fidelity, it must be capable of re-creating the highest level of sound likely to be required in the particular environment without generating audible distortion in its efforts to do so.

NOISE: All reproducing equipment introduces some noise background of its own into the ultimate sound – rumble from the signal source mechanism, hum from the power mains or supply, hiss from the amplifier circuitry. It is reasonable to expect this noise contribution to be inaudible to listeners in the normal environment, with the equipment set for normal playing volume but with the disc or tape stationary.

AMPLIFICATION or GAIN: These two words, which mean much the same thing, express a rather practical requirement. The equipment must have the ability to amplify the smallest signal likely to be fed to it from the signal source (radio tuner, disc or tape player) to a level necessary to produce the required power output. The ability is frequently and alternatively stated in terms of the equipment's SENSITIVITY or the minimum level of signal with which it must be fed to produce a cetain stated level of power output.

Without being backed up by figures, the above statements look rather vague but this is not entirely inappropriate, because reactions to the quality of reproduced sound vary greatly with individuals,with their background and their natural acuity. There is no such thing as "perfect fidelity" equipment and all concepts as to what can be accepted as high fidelity or adequate fidelity necessarily include some subjective compromise.

From a practical viewpoint, the person who is seeking a maximum degree of satisfaction from reproduced sound has to find his own compromise between what his senses demand, what his household can accommodate and what his means can provide. Some enthusiasts are relatively easy to please, some extraordinarily difficult.

Still keeping things on a general plane, how does typical present-day sound-reproducing equipment rate in terms of fidelitv?

Starting at the bottom, the Australian market is generously supplied with small portable transistor receivers, small portable disc players and small portable tape players. These appeal strongly to teenagers and casual listene but, despite labels and despite advertising propaganda, it is axiomatic that none of this equipment can lay claim to high-fidelity performance on even the most tolerant basis. Here are some of the basic reasons for saying this:
*Most such equipment is capable of mono reproduction only and it therefore ignores essential signal information which is now available to enthusia on both disc and tape.

*Because of limited space and the limited amount of energy available economically from internal batteries, the amplifying circuitry is unable to deliver the kind of distortion-free power required for high-quality listening.
*The small physical size of the equipment dictates the use of a diminutive loudspeaker and this, along with the very limited baffle area provided by a small cabinet, eliminates any chance of reasonable bass response.
*Most such equipment is built down to a savagely competitive price and the need to economise aggravates matters by forcing compromises additional to those listed above.

As distinct from the very small portable equipment just mentioned, the market is also well supplied with record players and tape players which are a good deal larger—more transportable than portable. Overall, these equipments more commonly provide stereo reproduction, use larger loudspeakers in larger cabinets, reflect a somewhat more generous design approach and, in the case of mains-operated units, avoid the power limitations imposed by batteries.

As a result, the standard of reproduction available from such equipment is normally a good deal better than from the kind of unit first mentioned and they are to be preferred where diminutive size and portability are not absolute prerequisites.

However, virtually all such equipment can only lay claim to an "ordinary" standard of reproduction. Inside, one will almost invariably find an ordinary mass-produced pickup and motor (or a mass-produced tape deck), quite ordinary loudspeakers in quite ordinary boxes and amplifiers to drive them of quite ordinary specifications. They may sound pleasant and, for many listeners, adequate, but this is simply a measure of what can now be achieved with "ordinary", non-expensive equipment.

A third class of reproducing equipment widely offered is the traditional domestic radiogram, which now quite commonly rejoices in such names as stereogram, TV-gram or tape-o-gram. Scorning portability or battery operation, these units are as much designed as items of furniture as they are for sound reproduction.

Most such units, and certainly all but the most expensive ones, again use ordinary mass-produced disc and tape playing decks, ordinary amplifiers and ordinary loudspeakers – in fact largely a repetition of what is found in the transportable equipment just mentioned. They may have some advantage in the overall balance of the sound, if only because the larger cabinet area provides better baffling of the loudspeaker, ensuring a somewhat better bass response.

A few—just a few—furniture-style radiograms incorporate amplifiers of more generous design and either better, bigger or multiple loudspeakers, to raise the standard of sound reproduction above the "ordinary". Such units are not necessarily distinguishable by any such label as "high fidelity" but rather by the fact that they are at the top of a particular manufacturer's line and carry a price tag and specifications that set them somewhat apart.

Price and specifications notwithstanding, however, virtually all furniture-style grams have the inherent limitation that acceptable cabinet styling dictates that the loudspeakers be too close together to permit development of a proper spatial effect. A significant part of the benefit of stereo is sacrificed thereby.

While requirements vary somewhat with circumstances, a good stereo effect depends on having the loudspeakers sufficiently far apart so that they subtend an angle of between, say, 45 and 60 degrees at the listener's head. To achieve this kind of situation with a single unit radiogram would require the listener to sit within about six feet of the cabinet. Even then, the loudspeakers would be pouring most of their high frequencies into the upholstery of the chair rather than into the listener's ears!

In this respect, transportable equipment may have an advantage, because the loudspeakers can often be separated by the required amount.

Out of all this, however, comes the rather sobering fact that, while the vast bulk of sound-reproducing equipment is either intriguingly small, handily portable or domestically imposing, it is still "ordinary" in its performance. This, despite liberal use of advertising superlatives.

Knowledge of this fact has sustained the breed of true high-fidelity enthusiasts who refuse to accept the "ordinary" and who pursue the extraordinary to the best of their ability and of their means.

And out of this comes many an argument with the wife who cannot see—or hear—anything wrong with "that lovely radiogram"; who simply fails to understand why it is necessary to put separate cabinets here and there ... and there! To which the appropriate answer seems to be:

"Are we buying it to listen to or to look at?"

The answer to such a question must be decided in each household situation, as also the amount of money, space and time that can be allocated to the enjoyment of sound reproduction. However, for those who seek to move beyond the bounds of the "ordinary" into the realms of true high fidelity, it is possible to set down certain guide lines:

*The system must be stereo.

*The loudspeakers must be separated by a suitable distance.

*The loudspeakers, with their enclosures, must be properly designed as complete units—not just any kind of loudspeaker in any kind of box. While large loudspeakers in large enclosures still offer an advantage, modern techniques have made it possible to obtain very gratifying results from well-designed, compact systems.

*Something more than the usual small amplifier, as designed into routine stereograms is desirable, particularly if it has to power small, compact loudspeakers, which are less sensitive than larger ones. Look for something about two or three times the power output and preferably with facilities for separately boosting or cutting the treble and the bass.

*It doesn't matter a great deal whether the amplifier uses valves, transistors or integrated circuits. All are capable of giving the required order of performance and, provided the basic fidelity requirements are met, the sound will be the same.

*The quality will be influenced markedly by the characteristics of the device producing the signal, normally a disc player or tape deck. Every effort should be made to secure the best one can afford.

*Discs and tapes should be selected carefully in the first place and cared for during their life. Watch the reviews for recommendations and keep your ears open when listening to other people's equipment. Careless handling and dust can ruin discs; careless handling and playing with magnetised heads can ruin tapes.

A few manufacturers in the "radiogram" field make available equipments with up-graded electrical specifications and with loudspeakers in separately mounted cabinets. For those who want to buy somewhat above the ordinary, but who are disinclined to get involved with custom-assembled systems, these are well worth considering. However, they are usually not cheap and enthusiasts commonly face the question as to whether they would not do better to assemble a system from selected off-the-shelf items, with less emphasis on handsome cabinet work.

The enthusiast who is so inclined is provided for by any number of high fidelity specialists, who can offer a wide range of radio tuners, disc and tape players, amplifiers, loudspeaker systems and, in many cases, the appropriate cabinet ware. From such items complete, higher quality amplifier systems can be built up, ranging in price and performance from something above the ordinary radiogram level to very elaborate, very expensive systems. Most enthusiasts settle for something in between.

Many firms specialising in higher quality reproducing systems will recomment, supply, install and service unitised equipments, so that the owner needs no more technical knowledge that is necessary to manipulate the rather more imposing array of knobs. On the other hand, through sheer interest in the subject, many audio enthusiasts do pick up a fair amount of technical background.

AMPLIFIER POWER RATINGS

In recent years, a great deal of confusion has built up around the subject of amplifier power output ratings. References are common to "RMS watts", "American watts", "music power" and so on. Some amplifiers seemed to have much higher ratings than others having a similar valve or transistor complement. Let us try to dispell some of this confusion.

There always has been a certain amount of confusion about amplifier power ratings but it is only recently that it has built up to almost bewildering proportions. The basic reasons for it will probably be understood best by going back to fundamentals.

The most matter-of-fact method of assessing the power output of an amplifier is to connect to its output terminals an accurate resistor equal in value to the specified load and capable of dissipating the anticipated order of power. Typically, this might be a resistor of from 2 to 15 ohms, substituting for a loudspeaker voice coil, and rated to dissipate from 5 to 20-odd watts.

The amplifier is then fed with a 1000-cycle tone from an audio generator and the level adjusted so that the output voltage across the load resistor, as viewed on an oscilloscope, is as large as possible, without the waveform being noticeably distorted. A more precise assessment of distortion level can be made but this involves a further order of complication. For most practical purpose, a sine wave which appears regular and symmetrical to the eye, can be regarded as "undistorted".
Having thus set the level just short of overload, the voltage across the load resistor is measured with an accurate RMS reading AC voltmeter and the effective power worked out on the basis that the power output in watts is equal to the RMS voltage squared, times the resistance in ohms. The answer represents the "steady tone" power which the amplifier can actually deliver to the nominated load.

FOR ACCURACY

Two points are worthy of special emphasis:

(1) The measurement must be taken across a resistor, not across a voice coil. The impedance of a typical voice coil at any given frequency is not sufficiently precise or predictable to form the basis of a power calculation.

(2) The AC voltmeter must be one of known high accuracy at the frequency of measurement, because squaring the volts, as required by the power formula, magnifies any error that might be present.

This method of measurement is fundamental and is the one on which our own laboratory designs are normally assessed. It is regarded as a fundamental characteristic in most audio laboratories, even though, for promotional reasons, sales personnel may seek a somewhat more euphemistic figure on which to publicise the equipment.

In point of fact, the urge to come up with a more impressive figure for power output has been responsible for a great deal of the confusion, past and present.

The urge is particularly strong in amplifiers using a cheap and inefficient output transformer. If the efficiency is down around the 50 p.c. mark, as it can well be, it is not at all flattering to have to admit to 2 watts of measured output from a 4-watt output stage, or to 5 watts of measured output, when the figure should be nearer to 10.

The shortcoming can be camouflaged by measuring and thereafter quoting the output power, as delivered into a resistive load across the PRIMARY of the output transformer. Since this does not include transformer losses, it can produce a quite satisfying figure, though a false one, because the power is not available from the amplifier's true output circuit.

(Where, in the past, we may have quoted primary watts, this has either been specified as such, or has involved high quality transformers with minimum losses.)

A still further liberty with the truth can be taken by not measuring power output at all but merely rating the amplifier on the figure which can be expected from the particular output valve(s) or transistor(s). For example, an amplifier might be rated as a 17-watt type, merely because data suggests that a couple of 6BQ5s can deliver that much output.

This kind of assumption is wide open to two major sources of error; First, it assumes that the output stage is working under the exact and optimum conditions on which the published ratings were based and that the power is not being limited by some unforeseen complication. Secondly, it ignores output transformer efficiency, as already mentioned.

Out of these initial considerations, therefore, one can legitimately question power output as to whether it is based on:

(1) A true steady power reading at the output terminals into the optimum load, or,
(2) A reading at the primary of the output transormer ignoring, perhaps conveniently, transformer losses, or,
(3) A mere assumption, based on valve or transistor data, and possibly quite inaccurate.

The steady-tone or steady-power kind of measurement, which has been described, has been criticised, with some justification, on the grounds that it represents some amplifiers as giving less power output than they are actually able to deliver, when playing normal program material.

The basis of this contention is that the potentials applied to many output stages, under class AB conditions, are about optimum for maximum power output, when the stages are not handling any substantial signal level; these optimum potentials can be maintained, for brief periods. by the storage effect of filter and bypass capacitors.

When such an amplifier is reproducing typical program material, with loud peaks interspersed with sound of a much lower level, the bypass and filter capacitors have sufficient storage effect to maintain optimum potentials. Examination of the output signal with a voltage calibrated oscilloscope will reveal undistorted power output, on brief bursts of loud signal, of a certain order.

However, a steady-tone test of the same amplifier will cause the output stage to draw more steady current, will change the potentials across the storage and filter capacitors away from optimum and, as a result, will produce a figure of undistorted power output lower than the previous one.

From this has arisen the practice, employed in some quarters, of rating amplifiers in terms of "music power". While the difference will vary with design, one might expect a "music power" rating to be about 20 per cent above a steady-tone rating.

In short, there would probably be little to choose between two amplifiers, one rated at 10 watts steady-tone, the other at, say, 12 watts "music power".

But the real upset in power ratings has followed the realisation by some amplifier manufacturers (particularly American) that, on peaks of an output sine wave, there is root-2 times the voltage and root-2 times the current, compared with the RMS value of each. Therefore, the peak output of the amplifier is root-2 squared or twice the RMS output.

The temptation offered by this luscious figure proved too much for the relevant publicity departments so now we have a whole flush of American amplifiers rated, often without saying so, in terms of peak output, being twice the normally accepted R.M.S. output.

If you want to follow suit, home constructors with single 6V6 amplifiers should cease forthwith to call them 4.5 watters! They can deliver 9 "American" watts.

By the same reasoning, that globe in your ceiling fixture is not really a simple 100-watt type as marked. You see it really consumes 200 watts on peaks!

In short, to the queries which have been listed already, must be added the strong possibility of having to divide by 2 the power ratings of any amplifier from an American or Japanese source.

Thus, there is every chance that an 8-watt American or Japanese amplifier will be no more powerful than your humble Australian or English 4-watter.

PEAK MUSIC POWER

In fact, having adopted the principle of rating an amplifier in terms of power at the peak of the audio cycle, there seems no special barrier to rating it for the peak power at the peak of a musical transient. So, going on the previously suggested figure, a humble steady-tone 4-watt amplifier, could graduate to something approaching 5 watts "music power", then to 10 watts "peak music power".

And what of stereo?

The most informative practice is to rate a stereo amplifier as, say, 2 x 4 watts, indicating that it contains two separate 4-watt amplifiers – typically a pair of single-ended 6AQ5 or 6BQ5 stages. If the stereo signal happens to be of the "ping-pong" variety, you can expect up to 4 watts from this channel, interspersed with 4 watts from that channel.

However, reaching once again for rosy-coloured spectacles, the 2 x 4 watt system can become 2 x 5 watts music power, or 2 x 10 watts peak music power. Before you know where you are, you have a 20-watt stereo system!

Of course, this figure is very modest. Recently, someone drew my attention to an advertisement for an American stereo system putting "120 watts into your lounge-room" for so many dollars.

Man, what a system!

But I wonder.

Applying all the discounts, the 120 watts is almost sure to represent the sum of the two channels: 2 x 60 watts.

Again, this is almost certain to be a peak power rating so that, in terms of normal R.M.S. figures, it becomes 2 x 30 watts R.M.S.

MUCH LESS IMPRESSIVE

There is a further strong possibility that this would be the power on music peaks. On a laboratory type steady-tone test, one would expect 2 x 25 watts.

Now a couple of 25-watt amplifiers would be capable of no mean noise in a loungeroom but the figure looks much less impressive than an unqualified "120".

In fact, I imagine that there would be plenty of readers of these columns who are using amplifiers of this order, right now.

Until further notice, we propose to carry on exactly as we have been – rating amplifiers in terms of steady-tone R.M.S. power at the output terminals, with a reference to higher output on musical peaks where it is warranted.

If the resulting figures seem too modest by comparison with what you see in overseas journals and catalogues, mentally multiply them by at least 2, and you'll feel better!

LESS NOISE, MORE DYNAMIC RANGE

In terms of background noise, modern discs and tapes are a far cry from those of the immediate post-war era but engineers – and critical buyers – are not yet satisfied. The battle against background noise still goes on, as evidenced by recent developments.

Noise problems begin at the original recording site. Particularly with large orchestras and large organs, the recording equipment has to be taken to the performance, rather than vice versa. Microphones have to be placed so that they will "hear" the performance as a whole, involving something of the acoustic environment. All too rapidly does this mean that the microphones will pick up extraneous acoustic noise at sufficient level to be heard through the quieter musical passages. Largely, this is a problem of time, site and microphone placement and the problem is being eased by new auditoriums which have been designed with an eye to their likely use for recording and broadcasting.

However, this is only the beginning of the problem. In practical recording systems, the amplitude at which the strongest signals which can be recorded is limited by the onset of distortion in the medium – non-linearity in the magnetic or optical pattern, loss of tracing ability in a stylus system, etc. The weakest signals have to compete against the inherent noise in the system – grain structure in the film or tape, surface characteristics of discs, hiss and hum in the associated amplifiers etc. In practice, the ratio of the strongest to the weakest signals which can be successfully (and commercially) recorded, referred to as the "dynamic range" of the system, is smaller than is called for by many musical performances. As a result, either the musical performance itself has to be modified to the restrictions imposed by the recording system, or else the dynamic range of the signals actually recorded has to be compressed by manual or automatic manipulation of the recording amplifier gain.
In considering the problem, it is insufficient to think just in terms of the original process of committing the performance to a master recording. In practice, the signal on the master recording has to be transferred several times before it reaches the copy which a customer will actually buy. Each copying or "dubbing" process is likely to apply slight further restriction to the dynamic range, notably by adding a quota of noise to compromise signals which are at too low a level. The master recording, therefore, has to be engineered not as an end in itself but as the first link in a fairly long chain.

Many techniques have been employed in an effort to reconcile the naturally wide dynamic range of musical performances to the limitations of practical commercial recording systems. Typically:

(1) Setting the gain of the system to cope well with low level passages and manually lowering the gain in anticipation of loud passages.
(2) Setting the gain to cope well with loud passages but advancing it in anticipation of low level passages.
(3) Using automatic compression techniques, normally to restrict the amplitude of large waveform peaks.
(4) Use of multiple microphones and complex console techniques to pick up total or individual sounds from suitable distances.
(5) Use of highly sophisticated amplifiers and recording medium (e.g. magnetically coated 35mM film stock) to preserve the highest possible dynamic range during the early stages of recording.

Apart from the limitations of human operators, methods (1) and (2) in particular call attention to themselves in the final reproduction by their effect on the total noise and reverberation ambient of the particular performance. Anyone who has listened at all critically to reproduced sound will, many times, have noticed changes in background as some unseen and perhaps otherwise forgotten operator has turned a fader up or down.

Automatic compression or peak limiting (3) minimises this latter problem but all too easily introduces a distortion component in the waveforms upon which it operates. Nor is the process readily reversible; it is not easy to restore accurately the waveshape or to recover the original dynamic range, particularly for large orders of compression. This complicates any idea of using a compressed signal for the early recordings and transfers, and of restoring it, at least in part, on the final customer recording.

By using and especially combining techniques (4) and (5) records can be made having very low noise and a dynamic range with peaks of such amplitude that they are likely to exceed the tracking abilities of even high quality magnetic pickup cartridges. However, it is not always practical to use these techniques and the recording industry still has to face two rather harsh realities:

(1) Neglecting pops caused by surface effects, many discs are still quieter in terms of "white" noise than the master tapes from which they were taken, and
(2) While pre-recorded tapes on the market are free from clicks and pops and have better stereo separation than the equivalent discs, their signal/white noise ratio is markedly poor under similar playing conditions.

While commercial tapes and discs are both well able to meet the standards required by the present-day mass market, engineers and hi-fi conscious listeners alike are aware of these limitations and there is a constant challenge to overcome them. The aims may be expressed in three ways, all of which are really variations on the one basic theme:

(1) To achieve wider dynamic range from the ultimate consumer recording, minimising the problems of peak amplitude on the one hand and background noise on the other.
(2) To do so with less expensive recording and copying facilities.
(3) To increase the tolerance of the recording chain as a whole to signals of all amplitudes, so that the final result will depend less on critical setting up of gain and levels.

In the face of this situation, audio engineers and hi-fi enthusiasts alike are likely to show ready interest in any system claiming to increase the dynamic range and or reduce background noise, particularly if it is applicable to existing equipment. But, equally, they are likely to be dubious of any claim to have solved the problem in any easy way.

And they have certainly been dubious about the "Dolby Audio Noise Reduction System" which was announced some time ago. Until proved otherwise,"The Dolby" was fated to be treated as just another black box, surrounded by just another set of extravagant and unlikely claims. Gradually, however, the impression has grown that the Dolby unit may be worth a second look, even if only because it manages the job of compression and decompression far more effectively than earlier circuitry. Writing in "Hi-Fi Stereo Review" for July, '67, John Milder reflects some of this recent enthusiasm. What follows is a condensation of his article.

Over the past few months, recording engineers and executives have been talking enthusiastically about "the Dolby", a device whose purpose is to reduce the background noise of master tape recordings. According to reports from the recording industry, the new device has important implications not only for professional applications but for the ultimate quality of records to be played in the home. And from the evidence now supplied by the first two "Dolbyised" records produced in this country (one from Vanguard, one from Nonesuch), the reports seem to be justified.

I feel these two records represent one of the most clearly audible break-throughs in sound quality in many years. That is not the kind of statement I thought I would be making when given these recordings to evaluate, but the more I have listened to them, the more I have become convinced that the new Dolby system will become a sine qua non for recordings of serious musical material until some entirely new recording medium arrives.

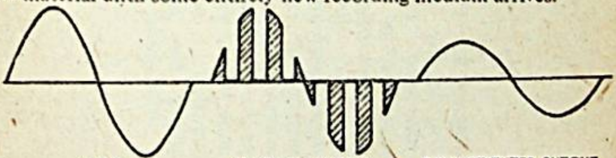


Figure 1: A system developed by Pye transforms the audio envelope temporarily into a pulse train. In this form the signal can be modified by varying pulse width.

The Dolby A-301 Audio Noise Reduction System is a simple-looking "black box" designed for connection to the inputs of a tape recorder during recording and the outputs during playback. The invention of Ray M. Dolby, an American audio engineer now living in England, it is designed to combat not only the high-frequency tape hiss added to any original signal during tape recording, but also many other kinds of background disturbances (including print-through echo, crosstalk, and scrape noise) that inevitably appear during the tape-recording process.

The new Dolby system does nothing at all to loud signals. Instead, working in four separate segments of the frequency range, it begins by boosting the level of all signals below a certain strength just before they are recorded. Then, during playback, it cuts these boosted signal areas back to their original level, and, in the process, reduces to the same degree any noise that has been added to the signal during the recording process.

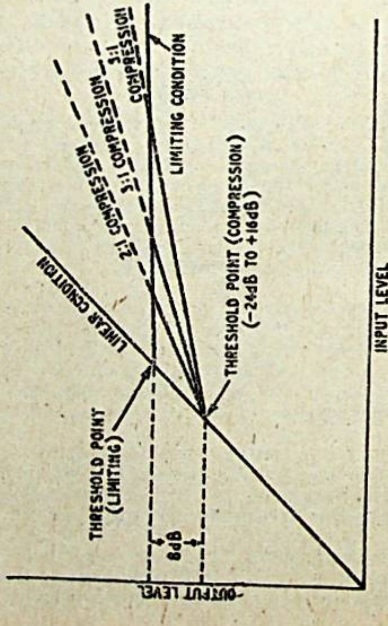
This "backward" process has been attempted before, but the Dolby system succeeds where previous gadgets have not, and for two reasons. Firstly, by handling four separate frequency bands and secondly because its operation is perfectly symmetrical in its boosting and cutting of signal strength.

It means that a tape can be stored for months or years between the two halves (input and output) of the Dolby's operational cycle, and that all tapes and recorders using the Dolby system are completely interchangeable. In the process, it assures that the print-through noise that often accumulates during the storage of tape will be greatly diminished when the tape is put through the second (or playback) half of the cycle.

As far as the recording industry is concerned, the big advantages of the system are, first, the tremendous basic gain in signal-to-noise ratio (10 to 15dB, depending on the frequency range) and, second, the ability to re-record ("dub") tapes for processing with virtually no discernible increase in noise from copy to copy.

Can you hear a 10 or 15dB reduction in noise? Yes, you can, and the results are far more dramatic than you would guess. The outstanding characteristic both of Vanguard's and Nonesuch's first Dolbyised records is clarity – to an almost incredible degree. It is not simply the absence of tape hiss or other noises during a quiet passage heard in a quiet room, but the absence of

Figure 2: Illustrating the performance characteristics of the Pye unit operating in linear mode, with three orders of compression or as a peak limiter with a 20:1 compression ratio above the limit threshold.



all sorts of effects, unidentifiable in themselves, that add a slight haze to the reproduction of musical instruments. The effects of print-through, crosstalk, and other kinds of middle- or low-frequency noise are unquestionably subtle individually or in combination, but there is nothing subtle about their absence.

Aside from clarity there is also a definite decrease in the amount of distortion perceptible in loud passages. This is a function of the engineer's ability, with the Dolby, to set peak recording levels a bit lower without having to worry about noise in quiet passages. And it is highly noticeable in the undistorted burr of the trombone in Vanguard's L'Histoire du Soldat and the unfuzzed fortissimo of the piano on the Nonesuch recording.

As for tape hiss as such, for all practical purposes it simply isn't there. Late at night, in a quiet room, at a listening level louder than my neighbours will freely tolerate, I couldn't hear the slightest sound of hiss until I came within a foot or two of a loudspeaker. In a direct comparison with the best record I know of, derived from a 30ips master tape recording on half-inch tape, instead of the usual 15ips recording on quarter-inch, the difference in favour of the new records was very audible. And after several hours of listening and crosschecking, I became convinced that virtually all of the "material noise" that I–and you–have been hearing from recent discs is actually the result of tape hiss in the master recording. (This is not to say that there can't be a bad pressing made from a Dolbyised master tape.)

The absence of tape hiss is not nearly as important for most listeners as the question of overall clarity. But it does become important to those who own very-wide-range loudspeakers.

Finally, it is worth emphasising that the Dolby system appears to add no distortion of its own worthy of the name. No spurious effects of any kind appear to be added. Aside from Mr Dolby's thoroughgoing engineering, this seems to be a function of the system's doctoring of only the lowest signal levels.

The two recordings referred to by John Milder are:
RACHMANINOFF Sonata in C Minor for Cello and Piano, Op. 19.
KODALY: Sonata for Cello and Piano, Op. 4. Harvey Shapiro (cello); Earl Wild (piano). NONESUCH H 71155.
STRAVINSKY: L'Histoire du Soldat, Madeleine Milhaud. Narrator; Jean-Pierre Aumont, the Soldier; Martial Singher, the Devil, Gerald Tarack (violin); Charles Russo (clarinet); Theodore Weis (trumpet); Julius Levine (double bass); Lorin Glickman (bassoon); John Swallow (trombone); Raymond Desroches (percussion); Leopold Stokowski cond.
Vanguard VSD 71165/66 two discs.

Whether John Milder's enthusiasm for the Dolby system is justified, only time will tell. It could be that some of the advantage in the records reviewed is stemming from improvements in other directions, or from special attention which detail often receives when something new is afoot. Again, Dolby may not have a mortgage on the ideas behind his unit and other engineers and companies may well have their own developing answers to the whole problem.

As a matter of interest, in this connection, the May 1967 issue of "Industrial Electronics" refers to a system which the Pye Company in England has developed to compress audio signals fed to sound transmitters, to obviate the effects of inadvertant overmodulation on peaks.

The article refers to the known limitations of the type of compressor in current use, using non-linear elements to reduce gain on unduly large peaks of signal:

(1) Distortion tends to be excessive if the system operates at too high a signal level, especially with large orders of compression.
(2) If signal levels are restricted to minimise distortion, signal/noise ratio becomes a problem.
(3) It is difficult to define precisely the characteristics of non-linear circuit elements and to provide their converse where required.
(4) Distortion components introduced by normal compression methods can themselves become modulation components, so that an attack on one problem introduces another.

In the new Pye system, the audio envelope is interrupted at a frequency much greater than the highest audio frequency being handled. This interrupted waveform, as illustrated in figure 1, is subsequently passed through a low-pass filter, which removes the pulse component and reconstitutes the original envelope, with minimal distortion. If no other circumstances intervene, the system is quite linear in its operation and output very closely approximates input.

However, at the point in the chain where the signal is in the form of discrete pulses, it is possible to modify their energy content in terms of pulse width and according to a predetermined mode governed by signal amplitude or pulse height. Any such modification of pulse width appears as a modification of output waveform relative to input waveform.

Figure 2 indicates the claimed performance parameters. Unlike the Dolby unit which leaves the peaks unaltered and elevates the low level signals above the noise, the Pye unit achieves a similar end result by elevating the entire recording level and operating to compress the louder signals. Alternatively, it can operate in a "peak limiting" mode, similar to existing peak limiters used by broadcast stations.

In compression mode, the unit operates above a defined pivot or threshold point with selectable conditions ranging from linear (no compression) to ratios of 2:1, 3:1, 5:1.

As a peak limiter, the threshold is raised by 8dB but the compression ratio is increased to 20:1.

While the whole of the discussion is in the context of compression and/or peak limiting for broadcast transmitters, its relevance to the recording situation is evident. Again, only time will tell whether the new generation of records and tapes on the market will be "Dolbied" or "Pyed".

WHY THE ELLIPTICAL STYLUS?

Hi-fi enthusiasts have doubtless noted references, of late, in advertisements and articles, to the "elliptical stylus". This brief article explains what is meant by the term and the possible advantages that this type of stylus may offer.

The problems which the new elliptical stylus are intended to alleviate start right back at the cutting stylus used to inscribe the original groove in the master.

Like the cutting tools used in an ordinary lathe, the recording stylus must have a certain critical configuration, if it is to cut a smooth, noise-free groove in the lacquer medium. The task is complicated by the fact that the stylus position is normally "modulated" vertically and horizontally with respect to the groove track and the facets must be so arranged that the stylus can still cut a clean groove, even when moving obliquely in respect to its nominal cutting path.

While a great deal of know-how therefore lies behind the production and use of cutting styli, it is sufficient to observe, for the present purpose, that cutting styli have to be of a fairly narrow, wedge-shaped configuration, as illustrated at the left of figure 1. This is a planview sketch, drawn as if looking directly down on the surface of the record.

If the stylus was required to cut only an unmodulated groove, the operation would be about as simple as it would be pointless. The groove could be "V" shaped and include any desired angle; the bottom could be sharp or radiused; the groove could be as deep and therefore as wide as the pressure on the stylus might determine and, more important, the groove width and groove angle could be maintained uniform from start to finish.

MODULATION

In fact, practical grooves are not unmodulated ones but those carrying the speech and music signals which we normally want to reproduce. And herein lies the difficulty.

When modulation causes the stylus to move obliquely in relation to the normal path, it cuts a groove of different shape. Looking along the groove, it would appear to have become narrower at the top, with walls at a steeper angle. The greater the "oblique" velocity, relative to the forward tracing speed, the more marked does the effect become.

Relating this statement to factors which immediately concern us, the constriction of groove width and angle tends to increase with signal frequency and amplitude and with reduced radius and lineal speed near the centre of disc. The problem therefore tends to concentrate around loud passages, containing high frequency energy, recorded on the inner tracks.

Since recording levels and useable disc area are dictated by other and important considerations, we have to live with the problem of varying groove shape, more or less as a heritage of the system.

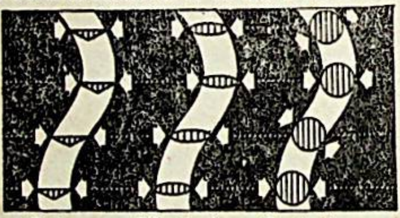
When it comes to playing back a disc, "non-cutting" qualities must be sought in the playback stylus just as diligently as the reverse qualities are sought in the cutting stylus. Broadly, this means a polished, rounded shape and, for fairly obvious practical reasons, the one which found universal acceptance is a stylus with a tapered, conical end, with its tip ground and polished to a hemispherical shape.

So again, as a heritage of the system, we have been compelled to trace, with a hemispherical tip, a groove which was originally cut with a wedge-shaped stylus.
This is illustrated in the right-hand sketch of figure 1. At the peaks of the waves, where there is momentarily no oblique travel, the groove is at its maximum "unmodulated" width. Elsewhere, and particularly crossing the zero axis, where the "oblique" velocity is greatest, the groove looks narrower and steeper to the hemispherical playback stylus being dragged along it.
Inevitably, the stylus must rise in the groove as it traverses the narrower portions so that it receives a vertical modulation, which is not part of the original modulation at all, but a by-product involving the difference between the shapes of the recording and playback styli.

Because it has such a fundamental basis, this so-called "pinch effect" is as old as lateral recording itself. However, the order of distortion attributable to it has largely remained well below the total distortion from other sources.

STEREO PICKUPS

More recently, a general reduction in distortion has caused pinch effect to assume relatively greater significance. Particularly is this so in the case of stereo pickups, because they have the fundamental ability to convert vertical modulation into electrical signal, the spurious component being substantially a second harmonic of the lateral movement.



Cutter Elliptical Conical

Figure 1: Reproduced from Shure literature, this diagram illustrates a track cut by a wedge-shaped stylus (left) and replayed either by an elliptical stylus (centre) or a conical-spherical stylus (right).

It has long been understood that pinch effect distortion is reduced by using a playback stylus with a smaller tip radius, the reduction being a fairly linear function over a range of values where the stylus can still be considered as a reference sphere riding the walls of a strictly V-shaped groove.

Unfortunately, a number of practical considerations make it rather unrewarding, at present, to use a stylus any smaller than with a 0.5mil radius.

The first is that, with a very small tip radius, the thrust per unit area is likely to exceed the rupture limit of the disc surface. This means that, by and large, it is only practical to use a tip radius of less than 0.5mil with rather exotic pickups tracking at about one gram.

Even more important is the fact that not all discs have a sharply defined bottom to the V-groove. There is therefore a very real possibility that a fine stylus will find itself riding in grooves having a rounded bottom of radius greater than its own tip radius. With no wedging action, "groove skating" will be evident, with its associated unpleasant distortion.

A still further problem is that dust particles are likely to settle, if anywhere, in the bottom of the groove so that, under other than ideal conditions, the bottom of the groove can be the noisiest zone.

The idea behind the elliptical stylus is to produce something that will have the tracing qualities of a very small-radius tip, while avoiding the aforementioned difficulties associated with the bottom of the groove. It is illustrated in the centre sketch of figure 1, and in figure 2, both reproduced from "Shure" literature.

Figures published by Shure indicate that their present elliptical stylus, viewed from the front or rear, has a radius of 0.9mil so that, if anything, it tends to ride higher in the groove than does the standard 0.7mil mono/stereo spherical tipped stylus.

However, the elliptical stylus is much finer, viewed from the side and Shure quote an effective radius of 0.2mil for the shoulders which actually contact the groove walls. They thus claim the advantage of a small radius relative to groove tracing, with a simultaneous freedom from the difficulties which concentrate at the groove bottom.

Their claims for improved performance are supported by measurements showing reduced tracing distortion.

CRITICAL CONTOUR

On the other hand, they admit to the critical nature of the contour, if such benefits are to be realised, and the problems of maintaining the requisite tolerances in manufacture and testing. They also point out that the surfaces which actually support the stylus are of very small radius, so that tracking weight must be kept down – preferably in the 1 gram region.

While the proposition is a very interesting one, it certainly cannot be accepted on face value.

For example, it can be shown fairly simply, by graphical means, that pinch effect is much less evident with a small sphere riding in a given V-groove than with a large sphere. It is by no means self-evident that an elliptical stylus will combine the best characteristics of both by simply merging their two radii into a bi-radial or elliptical shape.

It may be necessary, rather, to examine the action on the basis that the stylus rests on support points which are the greater distance apart at instants when the groove is in line with the mean path; that, when the groove deviated obliquely and constricts, the support points move around the shoulders and become separated by a smaller distance; that zero pinch effect modulation will be evident when the distance between the support points matches the constriction of the groove as it deviates with modulation.

STEREO TAPE TRACK STANDARDS

A letter from a reader raises a question to which I personally have not had occasion to give much attention, namely the recognised disposition of left and right information on the tracks of a stereo tape. The matter is probably less urgent here than in countries where pre-recorded stereo tapes are widely used but it may be a good idea, nevertheless, to initiate discussion.

The letter in question reads as follows:

Dear Sir,

Before the tumult and the shouting die over your recent stereo tape recorder, it might be in order for you to make an authoritative statement concerning the convention used in recording which channel (left or right) on which track (upper or lower).

It may prevent the annoyance of later having to swap connections every time a "foreign" tape is played on equipment having no "Stereo Reverse" control.

To my knowledge, this matter has not been raised recently in any magazine although, from the instruction book of one commercially available machine, with stereo playback facilities (Philips EL3541), the left-hand channel is taken from tracks 1-4 (upper), the right-hand channel from tracks 2-3 (inner or lower).

Is this the agreed convention?

Yours faithfully, M.J.M.

As I said above earlier, I have not had occasion to investigate the matter personally, any stereo recording I have done to date being of a purely personal, transitory nature, as when testing equipment, etc. Since there was no intention to store the recorded material, it mattered little which way the microphones were disposed, provided the speakers were arranged to suit.

However, where one is building a library of stereo material for subsequent use, it is worthwhile to make it compatible with commercial practice so that the need is avoided, as our correspondent points out, for swapping leads to keep right and left information where it should be.

But what is "commercial practice"? To find the answer to a question like this normally involves searching through all the books one can find and talking to everyone in the field you can think of, to discover customs, practices, discrepancies, "official" agreements and so on. As surely as one makes an "authoritative" statement without doing so, one is certain to be reprimanded for ignoring some substantial body of practice.

It so happens, however, that a recent article by Herman Burnstein in the magazine "Audio" surveys current tape practice and, among other things, displays diagrammatically the prevailing convention (as he sees it), the disposition of information with the half-track and quarter-track systems. Happily, this corresponds wit the practise quoted for the Philips recorder so that some correspondence is established between European and American practice.

At this stage I can do no better than reproduce with acknowledgement the relevant portion of Herman Burnstein's article. If readers happen to have arrived at different views, after appropriate study of the subject, it provides the perfect opportunity for debate.

In the approximate decade and a half that the tape recorder has been with us, a device of this complexity cannot help but have undergone changes, some under the impact of stereo and others for different reasons. On the whole, these changes have been of an evolutionary sort.

In the past several years, the attention of the high-fidelity tape fan has probably been most strongly focused on the question of format, that is, of track arrangement. This has been quite an unsettling question, involving a series of changes that tended to render tape machines obsolete unless subjected to fairly expensive modifications.

Fortunately, it now appears that the issue of format has been satisfactorily decided for a substantial time to come.

For a goodly number of years, while all was mono, format was simple and quite standard. Most home machines operated on a half-track basis, as in (A) of Fig. 1.

After a tape was recorded or played in one direction, using nominally half the tape width (with a safety island to prevent cross-talk between tracks), the reels were reversed by the operator so that the other half of the tape width could be used.

If one desired to operate on a full-track basis, it was usually necessary to get full-track heads on special order.

The confusion began with the advent of stereo, which employed tape as a popular medium before it made widespread use of disc and radio.

Initially, stereo also operated on a half-track basis, as shown in Fig. 2, with the upper track being used for one channel and the lower track for the other.

TWO PHASES

Half-track stereo went through two phases. First, a staggered head arrangement, as in (B) of Fig. 1, was used; that is, two heads were employed, spaced about 1.25in apart. The gap of one head spanned the upper track, while the gap of the other head spanned the lower track.

But this was a cumbersome arrangement and after a while the in-line head (C), of Fig. 1, made its appearance, consisting actually of two heads within a single housing, so that the gaps were one above the other in a vertical line. The in-line head called for a decided advance in the art of head manufacture, an important part of the problem being to prevent crosstalk between what were, in effect, two heads in very close proximity.

Hence quarter-track stereo (or four-track stereo as it is often erroneously called) was a natural development. Fig. 2 shows the quarter-track format. The tracks numbered 1 and 3 are used when the tape is operated in one direction, and tracks 4 and 2 are used in the other direction.

While quarter-track stereo solved the problems of convenience and of tape cost per minute of recording, it raised other problems. The narrower track means less signal is presented to the tape playback amplifier, making it more difficult to achieve a good signal-to-noise ratio.

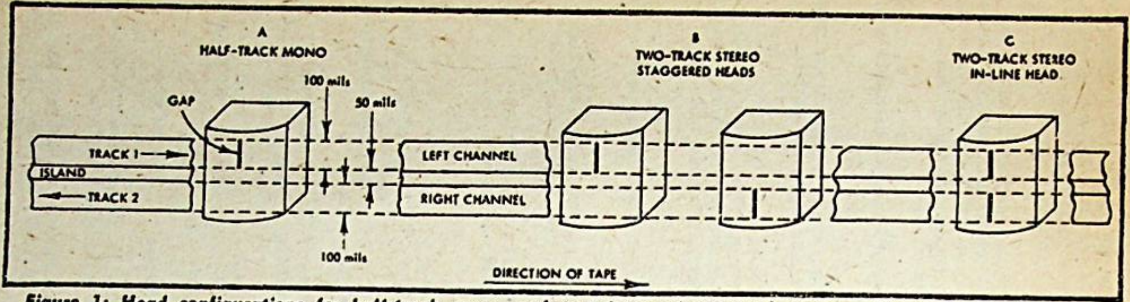


Figure 1: Head configurations for half-track mono, and two forms of two-track stereo. The in-line head arrangement, as at "C," is now the recognised standard and corresponds with the in-line heads used on four-track machines.

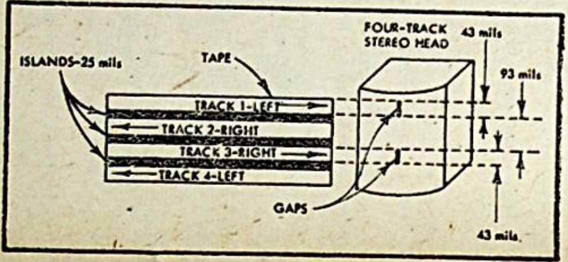


Fig. 2. The normal disposition of information on four-track stereo tape.

Also, the narrower track means there is less chance for tape irregularities to average out, resulting in greater likelihood of such disturbances as drop-outs (sudden, brief drops in volume). However, improvements in heads and in tape have gone a long way toward coping with these problems.

The introduction of quarter-track heads has made quarter-track mono operation possible, which means putting four instead of two mono tracks on tape and thereby doubling the amount of recording time for a given amount of tape.

To take advantage of the possibility, some manufacturers have incorporated the necessary switching facilities in their tape machines to that the user can record one channel without erasing the second channel.

Even for new models of tape machines, it has not been easy to keep pace with changes in format. Thus, today there are still some transitional problems. For example, some machines provide for stereo recording on a half-track basis, while permitting either half-track or quarter-track stereo play-back.

To alternate between half-track and quarter-track playback, either of two methods is used: (1) A quarter-track head is employed for both modes of operation; (2) Two playback heads are employed, one quarter-track and the other half-track.

Where only a quarter-track head is employed, some machines permit this head to be shifted up or down, depending on the mode of operation, so that it will span as much of the recorded track as possible in each mode; other machines keep the head stationary to avoid the possibility of azimuth misalignment as the head is moved up or down.

The speeds principally in home use are hardly different today than they were almost at the outset of the tape era. The major difference is in the performance, particularly with respect to high-frequency response, attainable at a given speed.

Stated, conversely, for a given level of performance the speed requirement has been lowered by 50 per cent.

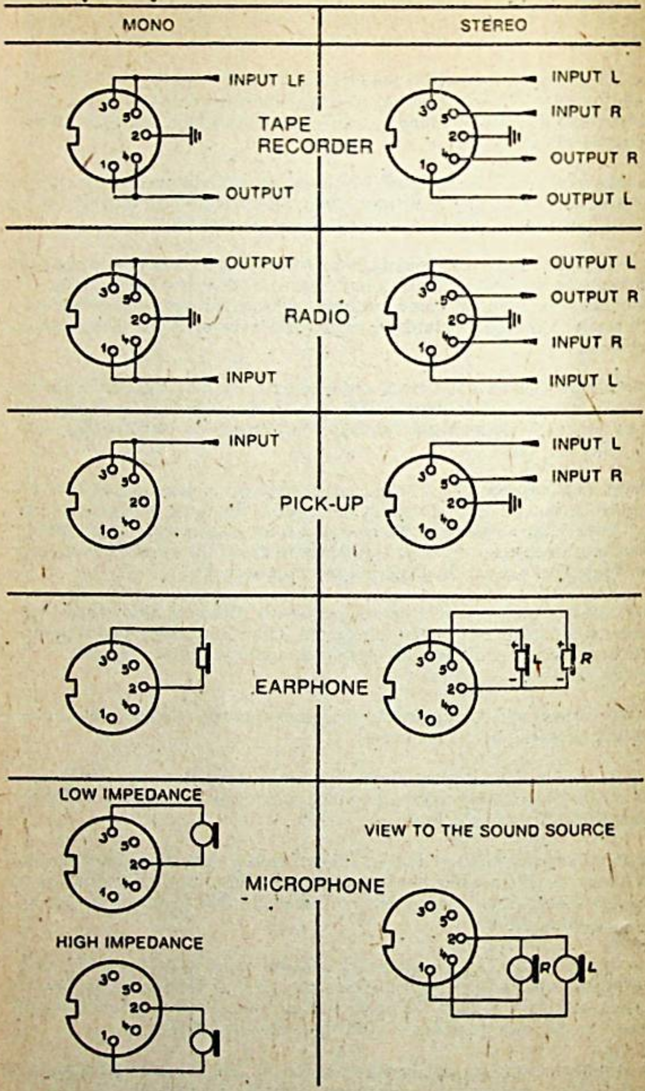
Top quality home machines almost without exception used to offer a speed of 15 ips. Today some do and a number don't, with 7.5 ips being the highest speed of the latter.

Taking home machines as a class, 7.5 ips has been the most popular speed for many years. However, the upper response limit used to be considered about 8000 cps at this speed, whereas today 15,000 cps or more can be reproduced at 7.5 ips.

A coupling of the 7.5 ips speed with 3.75 ips has long been popular and continues to be so. Although 3.75 ips used to be derided as a medium for anything resembling high fidelity, today it is taken fairly serious because it permits response to about 10,000 cps, which is quite good.

Inasmuch as today we can do about as well at half the speed of yesteryear, the 1.875 ips speed is coming into increasing vogue. Response to about 5000 cps can now be attained at 1.875 ips, and there are substantial hopes for a still better future.

WIRING DIAGRAM OF DIN CONNECTOR SOCKETS



In some, 7.5 ips is the accepted speed for high fidelity home tape recording. Virtually all prerecorded tape is made at this speed. Although there was an attempt several years ago to build a market for 3.75 ips pre-recorded tape, this effort seems to have fallen by the wayside.

The improvement in frequency response at a given speed is due to several factors, including playback heads with narrower gaps, improved tape oxide formulations, and better mastery by tape machine manufacturers of a rather sophisticated art.

The principal factor has been the development of heads with narrower gaps for playback (a narrow gap is unimportant in recording; in fact, a relatively wide gap tends to be superior).

Originally, tape playback heads (or heads for both recording and playback) had a gap about .0005in wide, which limited response to not much more than 7500 cps at 7.5 ips. The next development brought heads boasting gaps of about .00025in, which made playback response to 15,000 cps feasible at 7.5 ips.

In the last two years or so, gaps have been reduced to the order of about .0001 in which theoretically permits playback response to 30,000 cps at 7.5 ips and to 15,000 cps at 3.75 ips.

However, as depicted in Figure 3, certain magnetic phenomena that occur in recording (demagnetisation and bias erase) make it impractical to try for 30,000 cps and 15,000 cps response, respectively, at 7.5 and 3.75 ips.

This doesn't mean it is impossible to record 15,000 cps at 3.75 ips. But it does mean that undue sacrifices in terms of distortion and scignal-to-noise ratio, which are interrelated with frequency response, would be required. On the other hand, it is quite possible that future developments will make it possible to attain at 3.75 ips that which we now enjoy at 7.5 ips, in the same way that 7.5 ips has replaced 15 ips.

In fact, on a laboratory basis, response to 15,000 cps along with good performance in terms of distortion and signal-to-noise ratio has been demonstrated at a speed as low at 1.875 ips—but to repeat, on a laboratory basis.

One area in which home tape machines have shown notably slow progress is signal-to-noise ratio. In part, this is due to the change from a half-track to a quarter-track format, resulting in a signal loss exceeding 6 db inasmuch as a quarter-track is somewhat less than half as wide as a half-track (because of the need for three safety islands instead of one).

A tape of good quality inherently has a signal-to-noise ratio of something like 65 to 70 db at 7.5 ips. This means that a 400-cps signal recorded at a level producing 3 per cent harmonic distortion on the tape will be about 65-70 db above the level of noise due to the tape itself. However, what we ordinarily have most to contend with is not the noise of the tape but of the tape amplifier, particularly the playback amplifier.

BEST MACHINES

The very best home tape machines are able to achieve signal-to-noise ratios of about 55 db on a quarter-track basis at 7.5 ips. True, they could also achieve 55 db years ago, but then it was on a half-track basis. Hence, in relative terms, there has been improvement. Such improvement is due to the development of superior play-back heads having high inductance and therefore capable of delivering relatively large amounts of signal (in absolute terms, still only a very few millivolts). But these heads are found in machines having separate heads for record and playback—that is, in the better tape recorders.

In contrast to their generally poor showing in the matter of signal-to-noise, many home tape machines have made amazing strides with respect to wow and flutter.

Whereas wow and flutter amounting to 0.5 per cent or more was not uncommon in early units, today a fair number of tape recorders, including some that are quite moderate in cost, fall well within the professional limit of 0.2 per cent.

In fact, a number of home machines have been credited with wow and flutter less than 0.1 per cent.

On the other hand, exceedingly few home machines have managed to come up to the professional specification that a machine shall operate within 0.2 per cent of exact speed. Most home tape recorders depart from exact speed by 0.5 per cent, some over 1 per cent.

NOT NOTICED

Fortunately, most of us cannot detect, and therefore are not bothered by, deviations from correct pitch until they are over 1 per cent, often well over this figure. But those few who have a keen sense of pitch must search carefully for a home tape machine that meets their needs, at least if they plan to play prerecorded tapes. If they plan to record and playback on the same machine, the speed error, if any, will be minimised.

It is interesting to note that even home machines with hysteresis motors, which operate at quite constant speed, may have speed errors ranging from about 0.5 to 1 per cent. What these machines lack is precise dimensional accuracy of the capstan, and possibly other parts, so necessary for very accurate speed.

On the other hand, a hysteresis motor helps ensure constant speed from the beginning of the reel to the end of the reel, between recording and playback, and between one day's use and the next day's use of the tape recorder.

In constrast, other types of motors are subject to some speed change with varying voltage conditions and with the changing load on the motor as the take-up reel accumulates tape.

COMPENSATING SIDEWAYS DRAG

In all modern pickups, the arm is curved or the head is offset to minimise tracking error – a thoroughly commendable objective. A small but unfortunate complication of the practice is a resultant force which tends to push the stylus inwards towards the label. Many hi-fi enthusiasts consider it worthwhile to neutralise this force.
Consider first the simplest case of a straight, non-offset pickup, pivoted at point A, with the stylus at point B and with a line joining the two making a tangent to the particular groove at B.

With the record rotating in a clock-wise direction, as indicated, the frictional drag of the groove passing under the stylus pulls directly along the line AB. The only effect of the drag is to place a slight loading on bearing A.

Correction for tracking error normally involves using a bent arm or an offset head and either lengthening the arm, or pivoting it closer to the spindle, so that the stylus advances around the groove by an appreciable distance, in the direction of groove travel.

The result, with dimensions considerably exaggerated, is illustrated by the shape ADE, with A the base pivot and E the stylus. Line DE, representing portion of the arm, or a line drawn along the offset head, forms a tangent to the groove at point E.

It should be obvious from the sketch that frictional drag of the groove under the stylus point lies in the direction of the groove travel at the point of contact, or along the line ED. But the pickup is still pivoted at point A and a drag in the direction indicated must cause the arm to swing to the left, as viewed.

Unless countered in some way, this force must tend to increase the pressure of the stylus against the inside groove wall and to decrease the pressure against the outside wall. This could, conceivably, cause some imbalance in the pickup's response to undulations in the walls, being relatively more significant with the stereo system.
Also, combined with other disturbing factors, it may aggravate the tendency for the stylus to jump grooves.

The effect has been considered as minor enough to be ignored by the designers of most pickups, though measures to combat it have been suggested for users of some of the more specialised arms. These measures include:

(1) Dressing the outgoing leads in such a way as to apply slight opposing torsion.
(2) Tilting the turntable as a whole so that the head tends to swing towards the outside of the disc.
(3) Using a light thread and weight to pull the arm outwards by a suitable amount.

The first method is likely to be rather finicky and indeterminate; the second will not work with a fully dynamically balanced arm; all three suffer the objection that they apply an intermediate order of correction, whereas the thrust they are to combat actually varies with groove speed. However, it might justifiably be claimed that an "average" correction is better than no correction at all.

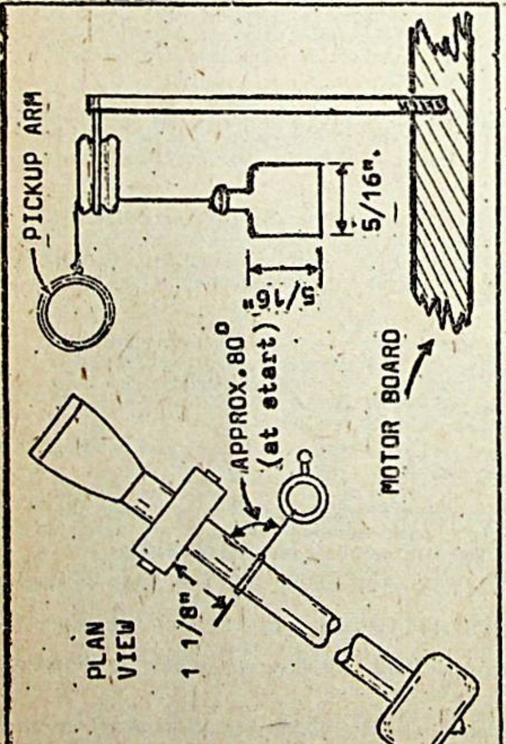
DO-IT-YOURSELF

It is against this background that we pass on to publish a short do-it-yourself article on the subject, submitted by Mr A.J. Barnes, of 11 Wilkinson Lane, Telopea, N.S.W. Our contributor says:

"It is now commonly agreed that the inwards dragging force which the rotating disc exerts upon the pickup can, to a distinctly audible degree, unbalance stylus pressure on the walls of the stereo groove.

"This inwards drag can easily be observed by resting the pickup on a blank recording disc. If you try this, be careful, because the pickup may swoop from the edge of the disc and, in one revolution, plough grooves in the centre label – not too good for the stylus!

"Manufacturers of some high-grade pickup arms now incorporate a simple device for neutralising inwards drag and there have been developed overseas many such devices for attachment to arms not initially so equipped. The device described and used by the writer has been influenced by these overseas devices.



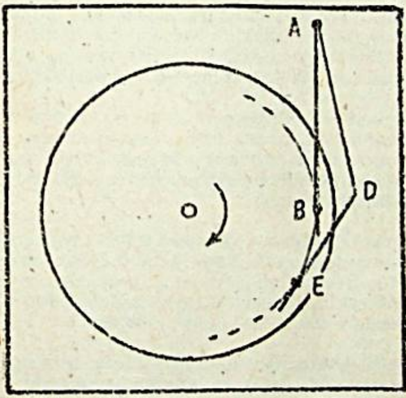
These diagrams illustrate how our contributor arranged the thread and counterweight. Where it is more convenient, the thread may be arranged to pull in the opposite direction on the counterweight or, as in the SME arm, on an extension from the bearing cage.

"The principle is simple, merely a weight and thread to exert an outwards pull on the pickup arm but care is required in its application. The critical factors are amount of weight, attachment position of thread to arm, the angle formed by arm and thread at start and finish of groove, and the friction of the thread through its supporting bracket.

"The device described here has been installed and tested for the popular "Decca" stereo pickup and can be installed as shown with confidence. The same idea can be applied to other arms but a blank disc will be required for checking the effect. With a blank disc revolving on a level turntable one should be able to rest the pickup on any part of the disc without the arm swinging one way or the other.

"The writer's weight is exactly three grammes and was formed by holding a piece of five-sixteenths inch brass rod in a drilling machine and forming the small eye at the top with a file held against the rotating brass rod. Those people handy with a lathe can scarcely do a better job!

"Two tiny flats are filed to provide a centring point for a one-sixteenth inch hole which forms the eye. Drill the eye before you cut off the surplus rod which acts as a stock to hold, whilst you drill the hole. Now the weight may be cut off to a length of five-sixteenth inch, exclusive of eye.



Sideways thrust is not produced as is sometimes supposed, by tangential forces but by the simple drag of the record on the stylus in a direction not in line with the pivot.

SUPPORT BRACKET

"The bracket for the supporting weight and thread was made from a small agate ring, as used for the tips of fishing rods and a small piece of one-eighth inch brass rod. The long ends of the metal clip were cut off short and were soldered to the brass rod, whose bottom end is threaded to screw into wood or metal of the pickup supporting baseboard.

"The choice of thread is important. Fine silk thread will do, but the writer uses nylon thread as supplied by chain-stores for connecting necklace beads in series! Single-strand nylon fishing line is definitely no good; it is not nearly limp or soft enough to bend properly and slide over the agate ring.
"Positioning of thread and bracket is important. For the "Decca" pickup everything will be OK if the following points are noted:–

(1) Weight, three grammes:
(2) Thread attached to arm at a point 1-1/8 inches from swivel bracket;
(3) Angle formed by thread and arm approx. 80 degrees at start of disc;
(4) Thread horizontal from centre of arm;
(5) Turntable level.

"Finally, swing the pickup over the turntable and adjust the length of thread. A tiny touch of cement will prevent the thread from moving away from its proper position on the arm.
"This is no wishful thinking job – when properly done one can really hear the improvement!"

EDITOR'S NOTE: Some may be prepared to debate our correspondent's claim that you "can really hear the improvement". A great deal would depend on the individual's acuity of judgment, the quality of the equipment in other directions, the record(s) involved and the sensitivity of the pickup to the order of side thrust envisaged. Sufficient to say that many critical high fidelity fans consider such compensation worthwhile.

There has also been a good deal of argument about the validity of the test suggested by our correspondent, using a plain, unmodulated recording disc. The argument centres around the degree of drag such a disc imposes on the stylus, relative to what it experiences when pinched in a triangular, modulated groove of different material.

Perhaps the most eloquent comment I can make is that I personally use a corrective weight on my pickup on the basis that it is a step towards the ideal, whether or not its effect is frequently or immediately obvious. Secondly, I set up the weight, originally, using a vinyl recording blank, because no better procedure was immediately available.

Of one thing I am very glad. Attention to dynamic balance as a prime consideration (September 1962), together with the use of compensation for sideways drag (along the foregoing lines) has rendered the pickup much less sensitive than heretofore to external shock.

Whereas, once, a slight bump on the cabinet or a too-heavy footfall would cause the pickup arm to jump grooves, now, if it jumps at all, it almost invariably drops back into the same groove, with hardly more than a dull boomp to indicate that adhesion has momentarily been lost.

EQUIPMENT PERFORMANCE – SOME TYPICAL FIGURES

Newcomers to the field of high fidelity sound reproduction frequently complain that the performance figures quoted in catalogues and advertisements mean little to them. This seeks to explain, in quantitative terms, such things as frequency response, distortion, power output, etc.

As we have seen, it is possible to discuss the performance of sound reproducing equipment in purely general terms but the fact remains that conversations, specifications and advertisements will be far more meaningful to the enthusiast who can understand – and perhaps quote – facts and figures. Within the scope of one short article, it is not possible to examine all the data which can be recorded about audio equipment, but the enthusiast should be able to acquire a working knowledge of the more important quantities and qualities without too much mental effort.

Let's begin by thinking in more specific terms about frequency response.

A fairly logical assumption is that a sound reproducing system should be capable of reproducing the whole range of frequencies audible to the human ear, without significantly accentuating or attenuating any frequency (or group of frequencies) relative to other frequencies in the spectrum. To use common technical phraseology, its response should be "flat" over the entire audible spectrum; the term "flat" envisages a graphical plot of performance against frequency, the result being a straight, horizontal line. We shall have more to say about this later on.

There are those who claim that this concept is too limited and who maintain that we may well have some obscure kind of a response to sonic energy outside the range of frequencies, which we hear in the normal way, and which we traditionally associate with speech and music. The claim is little more than speculation, however, and quite unproven. Until otherwise demonstrated, we can afford to ignore it.

It is a fact that certain items of audio equipment do have a response extending beyond the audible spectrum but this does not support the theory just mentioned, and it does not necessarily confer any advantage in terms of ultimate reproduction. In designing equipment to cover the full audible spectrum, it is fairly common to find that the response extends to a greater or lesser extent beyond the target figures.

The lower frequency limit of hearing is about 15 cycles per second. To use a term which is gaining favour, we can alternatively define the limit as 15 Hertz, abbreviated to 15Hz. If ears are energised at a lower frequency than this, our senses tend to interpret the successive pulsations as separate events, rather than as components of a unit sound.

The upper limit of audibility for humans is about 18,000cps or 18,000Hz, which can alternatively be written as 18 Kilohertz, or 18KHz. Such a figure is common enough for people up to about their middle twenties and gifted with a high degree of aural acuity in terms of frequency. The limit reduces markedly with various physical impairments and advancing years and is more likely to lie in the region 12-16KHz for the average adult in the 30-50 age bracket. However, for many adults, and particularly those in the over-50 age group, the upper limit falls below 10KHz.

At the other extreme, frequency components approaching 18KHz have a debatable significance in most program material and are imperceptible, anyway, to a large proportion of the potential listening audience.

More practical – and less demanding – frequency limits to aim for in sound reproducing equipment are 30Hz and 15KHz, and equipment which is substantially flat between these limits will leave little or nothing to be desired in terms of frequency response.

Any retreat from these amended limits must be regarded as a retreat from true high fidelity standards, as we currently accept them. This does not mean that sound reproduction to lesser specifications need by unacceptable. A response which is flat between say, 50KHz and 10KHz will sound only marginally "restricted" to critical listeners and actually impressive to those who are accustomed to less ambitious sound. This last group would actually represent the majority of the present population.

A medium-priced furniture-style stereogram could be expected to have a useful, though not necessarily flat, response between about 70Hz and 7KHz, using its own disc or tape source and with the tone control at full treble. On "AM" radio, the useful response would more likely lie between 70Hz and 4.5KHz.

Much the same figures would apply for the large portable (or transportable) 'grams and players, with perhaps, some further restriction on the bass due to a limited cabinet size.

For still smaller equipment, limits can be expected to narrow progressively with decreasing size and cost to, say, 170Hz as the lower limit and 4KHz as the upper limit, and not very flat in between!

So much for figures of frequency response. But what do we mean by such terms as "not very flat", "substantially flat", etc.? Some quantitative assessment is obviously called for.

To assess the frequency response of a piece of audio equipment, engineers energise it with input signals of known (and usually standardised) amplitude and at a number of frequencies distributed across the audible spectrum. They measure the output from the device at each frequency and, by ordinary graphical methods, plot a curve of output against frequency.

In the case of a pickup or a tape head, the "input" is normally a series of test tones, ranging from very low to very high in frequency, which have been pre-recorded for just such a purpose on a frequency test disc or a frequency test tape. The output from the pickup cartridge or tape head is usually measured in terms of the resulting signal voltage, either taken directly across the unit or from an interposed network or amplifying stage, where this is appropriate to the unit concerned.

In the case of an amplifier the input is normally derived from an audio signal generator, producing signals of the required amplitude and frequency. The amplifier output is measured in terms of the audio voltage which the amplifier produces at the connecting terminals for the loudspeaker.

These figures all refer to the limit at which the aural response "cuts off", meaning that the ears fail to resolve sound, as such, almost irrespective of its intensity. The range of frequencies over which ears might be considered to be reasonably sensitive would be more limited again.

Based on all that has been said, sound reproducing equipment can be considered as satisfying all possible requirements of frequency response if its response is substantially flat between the limits of 15Hz and 18KHz.

In practice there are good reasons why these limits might be regarded as unnecessarily wide.

Very low frequency sounds approaching the 15Hz limit occur very rarely in speech and music and, in any case, are extraordinarily difficult to sustain in the limited environment of a listening room.

Loudspeakers are tested by applying to their terminals input signals of appropriate voltage and frequency and measuring their acoustic output in a suitable environment with a microphone/amplifier system of known characteristics.

In tabulating or plotting the results of such tests, engineers commonly make use of the "decibel" abbreviated in both the written and spoken language to "dB". In matters affecting the level of reproduced sound the decibel provides a more meaningful basis for evaluation than results expressed only in terms of voltage, current or power.

This follows from tests which, long ago, established that our evaluation of loudness is not related in a linear fashion to the power involved. If we start with one unit of power and then double it, the average listener will notice a certain increase in loudness. To obtain further similar increases in apparent loudness, the power has to be raised to four units, then to eight units, then to sixteen, and so on. This non-linear relationship between actual sound pressure and apparent loudness is the reason, in fact, why the same ears which can register the faintest rustle, can also cope with the sound of a multi-jet aircraft.

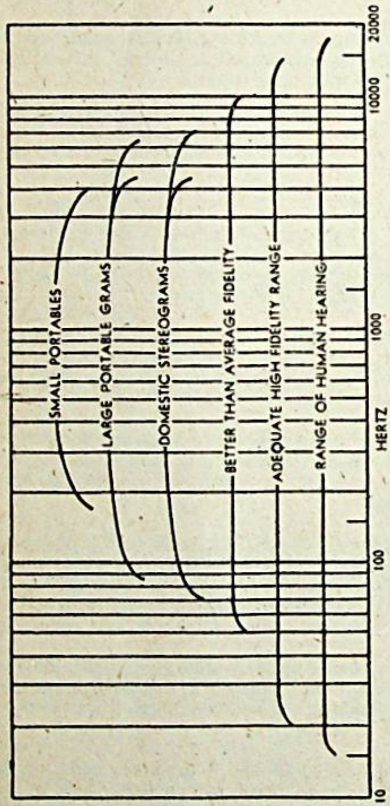
It transpired that the relationship closely follows a logarithmic ratio and the decibel, so beloved of audio engineers, is a figure obtained simply by taking the logarithm of any given power ratio and multiplying it by 10. A power ratio of 2 (log 0.3) is expressed as 3dB; a power ratio of 3 (log 0.48) would be 4.8dB; a power ratio of 4 times is 6dB, and so on.

In evaluating the performance of a piece of audio equipment, the tests aim to express in decibels what will be the effect, on the ultimate level of reproduced sound, of any departure of that equipment from the theoretical ideal.

By way of example, a pickup may be found to produce a certain output voltage over most of the range or at an arbitrary reference level such as 1KHz. But let's say that its output voltage is found to increase by 1.5 times at 7KHz, possibly due to a resonance in the stylus mechanism. If the level of signal voltage fed to the amplifier at 7KHz is 1.5 times normal, the actual power delivered by the amplifier would be 1.5 squared or 2.25 times normal. The log of 2.25 is .35 and, multiplying this by 10, yields the result that the pickup is "up" by 3.5dB at 7KHz.

In rather similar manner, variations in the overall amplification (or gain) of an amplifier, and variations in the acoustic efficiency of a loudspeaker can be expressed in decibels, insofar as they affect the level of the reproduced sound.

Visualising curves of output in decibels plotted against frequency, it is common terminology to suggest that a certain piece of equipment is "up" or "down" by so many dB at such and such a frequency. Again, the response curve might be described as exhibiting "peaks" or "troughs" or "prominences" – all words which are fairly obvious in their meaning.



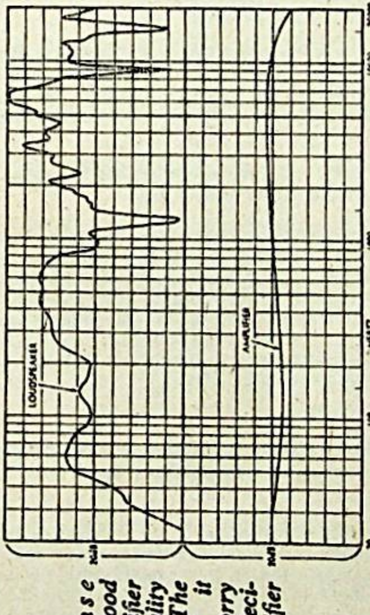
In a general way, this diagram depicts the frequency range for young and unimpaired human ears, and the frequency range of typical reproducing equipment. Note the progression from small portables to true wide-range high-fidelity equipment.

The value of the decibel in this context is evident in that it establishes a relationship between such otherwise divorced quantities as the voltage output of a pickup or tape head, the amplification or gain of an amplifier and the acoustic efficiency of a loudspeaker system. What is more, a peak of 10dB in a particular frequency region amounts to the same thing, as far as the final sound is concerned, whether the peak arises from pickup, amplifier or loudspeaker alone, or from individual contributions (e.g., 5dB, 3dB, 2dB) which happen to add up to 10dB.

Subjective tests have established that the smallest change in power level which listeners can detect under laboratory conditions is a ratio of about 1.6 times or 2dB.

A change in power level of 2:1 (or 3dB) is more easily noticeable but still not obvious to any but an attentive listener.

Based on these findings, it can reasonably be maintained that a sound reproducing system is indistinguishable from perfect, in terms of frequency response, if the sound output level as heard does not vary by more than 2dB across the frequency spectrum. With a minimum of lenience, this figure can be extended to 3dB.



The response curves for a good quality amplifier and a good quality loudspeaker. The latter makes it pointless to worry about the last decibel of amplifier response.

But such is the tolerance of our ears that variations of as much as 10dB, between different groups of frequencies, are interpreted as sound of slightly different balance but, none the less, of high fidelity. The realm of "ordinary" or "poor" sound would seem therefore to envisage variations over the spectrum of much greater than 10dB.

And here a word about terminology. All the references to this juncture, 2dB, 3dB, 10dB, etc., are on the basis that they represent the ratio between the highest and lowest power points on the curve. If a unit is credited with a frequency response which falls within plus and minus 2dB, the scope for variation is a total of 4dB. Similarly for any other figures.

In practice, it is not difficult to make an actual amplifier, these days, which is within 2dB from below 30Hz to above 15KHz.

The very best pickups and tape heads are hard put to it to stay within the 3dB limit over the same range, and the less expensive units under practical conditions come nowhere near doing so.

Loudspeakers in practical enclosures and in practical listening rooms never make the 2 or 3dB limits and are lucky to stay inside the 10dB limit! This is why even the best high fidelity loudspeakers sound "different" from one type to the next, even though all might be regarded individually as acceptable.

The accompanying curves will serve to validate the foregoing observations and also show what frequency curves look like.

Immediately obvious will be the very erratic nature of the response of even a high quality loudspeaker system in a typical listening environment. Because an amplifier has to be heard through a loudspeaker, and because it has to be fed from a signal source, with its own somewhat undulating response, it is quite pointless to make a fetish of the last dB in an amplifier curve. By all means seek an amplifier which has a flat and wide response, but there is nothing to be gained by insisting on a curve wider than the limits of audibility nor flatter than the inconsequential 2dB.

Let's turn, now, to distortion.

Used in its most general sense, the word distortion could logically be made to embrace any shortcoming in the reproduced signal occasioned by a technical imperfection.

In normal usage, however, the word has to do with lack of amplitude linearity in the equipment concerned; this simply means that the output of the device does not exhibit a strictly linear relationship (or is not strictly proportional) to the input. All practical equipment exhibits some degree of nonlinearity over its working range, the effect becoming much more evident as any inherent limitation on the signal or power handling capacity of the equipment is approached.

The effects of non-linearity are commonly referred to by the collective term non-linear distortion. They can be heard, observed, defined and evaluated in a number of different ways.

Most obviously, perhaps, non-linearity modifies the waveshape of a test signal, as viewed on an oscilloscope. For example, a gramo. pickup tracing a sine wave recording (i.e. a pure single tone) might, at some frequencies, deliver an output waveform which departs noticeably in appearance from the smooth sinoidal form. Again, an amplifier fed with a sine wave might, under certain conditions, deliver an output in which the peaks appear slightly compressed or even obviously flattened; or there may be kinks in the sloping sides of the waveform.

It transpires that waveforms which are thus misshapen contain spurious frequency components which have been superimposed on the original signal as a result of the non-linear characteristic. Because these spurious frequencies are in harmonic relationship to (or are multiples of) the original signal frequency (or frequencies) they are described by the term harmonic distortion.

The presence of spurious frequencies can be detected and measured by the use of equipment such as a Distortion Factor Meter, which gives a net figure for all distortion present; or a somewhat more specialised instrument called a Wave Analyser which allows the harmonics which have been added to a test signal to be evaluated individually.

The amount of harmonic distortion present in the output of audio equipment has long been accepted as a measure of its quality, in terms of linearity; so much so that, unless otherwise indicated by qualification or context, the word distortion occurring alone can be taken to mean harmonic distortion.

Non-linearity has another effect in that it causes original signals which are being handled simultaneously to intermodulate, producing additional spurious frequencies equal to the sum and difference of all frequencies so involved. Thus, signals at 400 and 1000Hz might intermodulate to produce resultants at 1400 and 600Hz. This, in addition to natural harmonics, as already outlined.

Test procedures have been evolved to measure intermodulation distortion and it is frequently quoted, at least for amplifiers, in addition to the more commonly quoted (Harmonic) distortion. For a given amplifier, the figure yielded by an intermodulation test is usually from three to five times that of the harmonic distortion test and this should be allowed for in looking at specifications.

While the effects of non-linearity can thus be observed and measured by special instruments, the question must follow as to how these observations and measurements relate to the audible result.

In fact, listener reaction to non-linearity and the resulting distortion varies enormously with circumstances.

Teenagers quite commonly operate portable receivers under such conditions of overload that the distortion can only be described as gross, and unacceptable by any other standards.

At the other extreme, high fidelity enthusiasts can become sharply aware of very small amounts of distortion, and may be critical of sound quality which the average person will accept as perfectly normal.

It is relevant also to mention that listener reaction to distortion varies widely with the order and the magnitude of the spurious harmonics and the signal frequencies from which they originate.

Loudspeaker systems, for example, commonly suffer from serious non-linearity when fed with signals of high amplitude and low frequency – say 60Hz or lower. They tend to "double" or "triple", producing an inordinate proportion of spurious signal at double the original frequency (second harmonic) or triple the original frequency (third harmonic). By and large, this very considerable order of distortion at the low bass end passes unnoticed by most or raises no more comment than something to the effect that the bass is "high pitched" rather than "throbbing".

Again, gramo. pickups fairly commonly encounter difficulties in accurately tracing modulation patterns at the higher frequencies, say 7KHz or above. The resulting output waveforms are noticeably misshapen on an oscilloscope and would yield quite high distortion readings on a distortion factor meter. Fortunately, the audible effects of such distortion are minimal because the spurious frequencies so generated are near to or beyond the upper limit of human hearing, whereas the same order of distortion related to signals lower in the range might be quite intolerable.

In the face of these and other such considerations, it is not possible to make any simple statement about the amount of distortion which might be tolerable in a high fidelity situation. Too many qualifications would have to be added about the order of the harmonic involved (2nd, 3rd, 4th, etc.) and the original signal frequency. But, even if this were set out, it would be useless because the information is rarely available for key items of equipment.

Distortion in the output of an audio device is normally expressed as a per centage of the total output.

For an amplifier, the figure can be derived fairly easily, using a high quality audio signal generator and a distortion factor meter. Distortion figures are commonly quoted for amplifiers and are an important element in competitive marketing.

Distortion is much more difficult to measure accurately for a gramo. pick-up or a tape player because of the difficulty of arranging a disc or tape signal source which has the very precise characteristics necessary for measurement purposes over a wide selection of frequencies.

With a loudspeaker, it is even more difficult, because few laboratories in the world can provide the sound-proof echo-less (anechoic) chamber and precision microphone equipment which is necessary.

Because of these difficulties and the unflattering figures which might emerge, anyway, accurate distortion figures are seldom taken on transducers (pickups, tape heads, but particularly loudspeakers) and are published even less frequently. Resort is had, instead, to fine descriptive phrases!

During the thirties, a figure of 5 per cent was fairly commonly quoted as an arbitrary limit for the permissible distortion level in an audio amplifier.

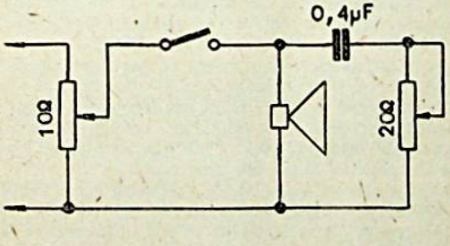
After the war, with the emergence of better output transformers and a greater utilisation of negative feedback techniques, the figure was progressively reduced until the majority of well-designed quality amplifiers were able to exhibit a distortion content of 1 per cent or less. In fact, a figure of 0.1 per cent gained acceptance as some kind of a design ultimate.

The current generation of solid-state amplifiers, with no output transformers and with a very high order of negative feedback, meets and betters this specification easily enough over most of the frequency range – though, curiously, some produce rather more distortion at low power output levels.

While it makes good sense to select an amplifier having the lowest available distortion content, there is no point in making a fetish of the exact figure, below a certain minimum. Purely as a guide, if an amplifier can be shown to exhibit less than 1 per cent distortion, it is extremely doubtful whether any improvement could be discerned in the final sound if it were replaced with another having a lower figure.

As has already been mentioned, loudspeakers commonly introduce distortion on to test tones which is plainly audible to the ear and plainly visible on an oscilloscope fed from a monitoring microphone. Such orders of distortion are almost certain to be so far in excess of the fractional percentage attributed to the amplifier that the latter is of little consequence. It is further swamped by distortions which occur before the program signal ever reaches the amplifier input.

In short, buy the best amplifier you can but don't make a fetish of harmonic distortion figures below 1 per cent. Concentrate rather on providing a loudspeaker and gramo. pickup which, in your own opinion, or that of accepted reviewers, sound particularly "clean", "transparent", etc. –



How to Control Volume on a Second or Supplementary Speaker

RECORDING 78 R. P. M. DISCS WITH MINIMAL BACKGROUND NOISE

After consideration we felt that we were a little nearer any satisfactory solution than when we started. Having taken some lengths to explain the existence of an enormous variety of recording characteristics, the only practical solutions offered are (a) the inclusion of a loudspeaker voice-coil treble attenuator, or (b) a filter.

The voice coil attenuators shown have very little flexibility and by this surely admit their own inadquacy for use with records from the wide variety of manufacturers which any collector is likely to possess.

Also, the majority of the discussion dealt with the replay of electrically recorded records, whereas perhaps the majority of collectors are most enthusiastic about their more primitive acoustic recordings, which undeniably can be made to sound aurally tolerable to the musically interested today".

In the instance quoted i.e. that he has some Caruso records that he wishes to play, I agree entirely that rather than go to a great deal of trouble, expense and probable frustration, the best reproduction would be secured by duplicating his records with expertly made microgroove re-issues. Caruso's entire output is available on microgroove and the EMI group re-issues of this and other artists are extremely well dubbed.

This course is not open to recordings of the majority of other artists, however, and here I will stick my neck out and say that the majority of re-issues on the market are appallingly badly done, and reproduction of even a worn original on a mechanical gramophone is likely to be far superior sounding (subjectively) to the under or over-filtered efforts, frequently dubbed at the wrong speed, which emanate ad nauseam under a variety of house flags and at a variety of prices.

What seems to be called for without going to the length of having a series of variable band-pass filters covering the pertinent spectrum, is a pre-amplifier unit along the lines of the Quad II with its highly efficient and flexible top-cut filter; but also a steep cutting filter at the bass-end of the scale.

The apt remarks about the inability of the average treble control to remove heterodyne whistles are equally applicable at the bass end of the scale, where the average bass control does not sufficiently remove the high rumble-levels encountered on certain recordings.

For instance, a system we once used produced very considerable distortion from certain acoustic recordings which were in good condition, and this trouble was traced to a high level low frequency signal, almost inaudible which was overloading the amplifier, causing severe distortion in the mid-range which we could hear! This was eliminated by the inclusion of a simple CR network at the pre-amp input.

However, the major cause of distortion is replaying early records, as pointed out but did not, we think, sufficiently stress, is poor fitting of the stylus into the groove. It is desirable to use an oversized rather than an undersized stylus -- the so-called "standard" 78 is made to the standard, as you are aware, of the R.I.A.A. which was not generally adopted until 1954. As by this time all records of any worth from anything except perhaps the smallest companies had their microgroove equivalents, the majority of records which should be played with this stylus size are hardly worth keeping at all.

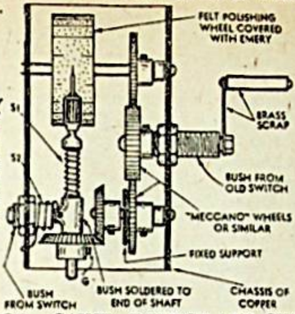
It is true that British Columbia used a narrow groove from the beginning, which will replay effectively with an R.I.A.A. 78 stylus, and this narrow groove, combined with particularly good surfaces on most Columbia records shows up others in a poor light, e.g. H.M.V. which

were cut with a very wide groove.
H.M.V. records made up to about 1932 will play properly only with a stylus of .004 to .005 tip radius, and unfortunately for the collector the majority of first-rate material is on this label. Oversize styluses of the dimensions I have just mentioned are virtually unobtainable and even when found, must usually be fitted to the cartridge shank by the user, as most makers are naturally unwilling to fit other people's styluses to their cartridges.
To the casual user, or the person without a sufficiently large collection to justify expensive pre-amps and custom ground jewel styluses, there is still a good, and comparatively cheap way out, which you did not mention. This is to use a thorn or fibre needle. These are still available, and cause almost no damage to a shellac groove. "Expert" pickups produce a superior miniature thorn moving coil pickup which has stood the test of time and can be fitted to any reasonable shell or pickup arm. With its associated matching transformer it is capable of fairly high output.
For a long time we have satisfactorily used E.M.I. type 12 cartridges, originally designed for steel needles with miniature thorns. Naturally, this being a 'needle armature' type, the output falls considerably when used with a non-ferrous needle and due to the alteration of the moving mass, the damping often requires adjustment but with correct adjustment it is possible to secure a reasonably level response, and the wear is low as can be gauged by the fact that one can play 20-30 sides without appreciable increase in distortion -- this with acoustic discs and their correspondingly low modulation.
A person investing in a miniature thorn pickup can, in many instances, dispense with any electrical filter and merely rely on the natural flexibility of the thorn to provide treble attenuation, but if this is found inadequate, a blob of petroleum jelly of similar viscous material works wonders in lopping off the HF response when smeared over the armature and in the magnetic gap.
We put stylus fit first in our list of problems, and while we mentioned steel and thorn needles, we did not pursue their use. Hence our assumption that listeners with only a casual interest in old records would want to use present day cartridges with, as stated, their problem of optimum tip radius.
But, having defined the problems we must admit to the conviction that many potential listeners would not persist with records with grooves which they could not trace with readily available cartridges and styluses.
A thorn needle can provide a quite practical answer to the problem of groove fit, though it will involve the user in finding or resurrecting a worthwhile pickup capable of accepting one.
If the pickup is one requiring a fairly substantial playing weight, the thorn will be ground into a suitable shape fairly quickly but it will only give a limited number of playings before shoulders appear near the top of the groove.
If a very light playing weight is used, it will take longer for the thorn to adapt itself but, of course, it will not wear out so quickly.
It raises the question, in fact, as to whether the best procedure is not to plan for a single optimum playing of each old disc, for the purpose of dubbing it on to tape. Thereafter, the tape can be played while the disc is passed around for inspection!
The merits and demerits of thorn and fibre needles were for years the subject of rather "tweedy" debate in journals and ther are doubtless

plenty of old-time readers who could resurrect the arguments if they were so inclined.
We prowled through gardens (private and botanical) looking for likely thorns, deciding finally that the most appropriate were those which "decorate" those tall cactuses appropriate to Mexican desert scenes. By a certain amount of skullduggery, we even arranged to grow our own supply -- the only trouble being that, by the time supply was assured, we no longer needed them!
One of the problems of the time was that the conventional thorn sharpeners could produce only a fairly steeply angled point. Attempts to produce a long taper were complicated by a tendency for the grinding wheel to "shed" the surface of the thorn, or snap the tip off just at the wrong moment. It also produced a high order of flexibility (and loss) in the taper.
We came up with our own kind of thorn sharpener which ground along the line of the thorn and produced a concave taper. The result was a long, fine tip which delayed the formation of shoulders, and without the flexibility that would have been apparent had the same apparent tip angle been continued as a normal taper.
At the time, we claimed a better surface finish, viewed under a microscope, than for thorns taken from a commercial packet.
Tested in a crystal pickup of conventional design, the reproduction was shown to be quite clean ....
However, there was a startling increase in the number of sides which could played with one sharpening.
With a typical crystal pickup and average orchestral recordings, it was found that a new needle fresh from the packet would play about 3 sides of 12 inch discs. Much the same was obtained from needles resharpened to the conventional conical point.
Against this, a point ground in concave fashion on one of our new sharpeners, played twelve sides or so before there was any suggestion of distortion.
Indeed, this figure has been shown to be conservative. Lengthy tests have been run on two entirely separate outfits, the pickup in each case being a crystal, with a needle pressure of slightly less than 2 oz.
Yes, 2 onces! 50 grams or more! How our ideas and concepts have changed.
So if you want to experiment with thorn styluses, first find a pickup or cartridge that will accept one. Then either buy a packet of thorn needles (if you can locate them) or find them au naturel and prepare them yourself.
As far as frequency compensation is concerned, we reproduced the set of curves for electrical recordings with further reference to "The hotchpotch of practice which was current during the days of mechanical recording". We said this to counter the original correspondent's apparent belief that he was mainly up against a problem of accurate compensation.
Our attitude was - and still is - that the question of compensation is complicated by so many hazards, known and unknown, that it is better to arrive at optimum sound by more or less arbitrary variation of bass and treble response.
The voice coil top cut filters were suggested as a measure to lop off that portion of the spectrum which contained only - or mainly - distortion components. It was not intended to double as a means of compensation.

The idea of using a control unit with adjustable and steep top-cut filter is an acceptable alternative, while the bass-cut filter would also be valuable in certain cases.

sharpener for thorn needles. The "chuck" was an old-time terminal device originally intended to grip wires. Spring S1 and grubscrew G permit axial positioning of the stylus tip, while spring S2 holds the tip against the grinding wheel.



PICKUP TRACKING ERROR - CAUSE, EFFECT AND CURE

Over the years, our editor has received a consistent influx of letters from readers, asking for advice on pickup tracking problems. This article discusses briefly the basic principles which has to be considered when setting up a tone arm for optimum tracking and gives some practical hints for mounting up and adjusting the tone arm and cartridge on the deck.

The basis of the tracking problem is probably known well enough to most audio enthusiasts, but here it is again in brief.

The grooves are produced in the first instance, by a cutting head which is transported across the face of the disc on a suitably driven lead screw. The geometry of the system is such that the cutting stylus moves inward along a perfectly straight line (which is actually a radius of the disc) as it impresses the modulation on the disc's surface.

Looking at it another way, it may be said that the axis of the cutting stylus's suspension is aligned with the tangent to the groove being traced.

If this requirement is not satisfied, the "axis of reproduction" is such that the stylus tip is not able to move in the same manner as the original cutting stylus - that is, at exact right angles to the groove tangent. Instead it moves at a slightly oblique angle and is, therefore not able to recreate the exact voltages which were used in the first instance to actuate the cutting stylus. The result is distortion in the reproduced waveforms.

Distortion can, of course, originate in any part of the reproducing chain and it is virtually impossible to distinguish between distortion from one source or from another. However, it is safe to say that distortion from tracking error will contribute to the overall distortion and the greater the tracking error, the more distortion will originate from this cause.

Unfortunately, however, desirable it is to have the "axis of reproduction" exactly tangential to the groove, it is not practical to satisy this requirement in record reproducing equipment as we know it today. In the first place, for a number of very practical reasons the pickup head cannot be mounted on a lead screw, as is done with the cutting head in the recording process. It has to be supported by a radial arm.

Use of a very long pickup arm is a possibility which approaches the ideal because the stylus would then be tracing an arc which would be a very small part of a very large circle, and therefore reasonably close to a straight line. If the tip of the stylus were then accurately aligned with

the centre of the turntable, we should be fairly close to tracing a radius across the disc.
But there are practical difficulties which limit the length of arm we can use. One is the mass and inertia of a very long arm. The other is the difficulty of accommodating such an arm in a cabinet of practical dimensions.
Some years ago an attempt to overcome the problem was made by designing an arm with an elaborate system of pivots attached to the pickup head, which allowed the head to adjust to correct tangential relationship to the groove as it tracked across the disc. Whatever the merits of the system, it apparently did not prove attractive to purchasers as it apparently disappeared from the market in a short time. And while it may have met tracking requirements, the extra mechanical complication may well have made it more difficult to meet the equally important requirements of a well designed arm - freedom of vertical and horizontal movement, dynamic balance, controlled internal resonance, and so on.
Another more recent attempt to provide a system with zero tracking error is the Rabco servo-controlled straight-line tracking arm which relies on information supplied by the head itself to maintain the correct tangential relationship between stylus and groove. However, this is expensive and requires a somewhat clumsy and bulky mounting arrangement. Whether this will succeed, or suffer the fate of earlier attempts to overcome the problem, remains to be seen.
It is undoubtedly significant that virtually all pickups currently available including a variety of specialist types use relatively short, radial arms, with varying degrees of car and attention paid to other characteristics such as those mentioned above.
Fortunately, a relatively simple measure is available to pickup designers to reduce the tracking error with radial arms, though it cannot provide a complete solution. This involves placing the axis of the stylus so that it does not lie parallel with a line drawn from the stylus tip through the main pivot centre of the arm; rather does it make with the line a deliberate angle. This practice is commonly referred to as "offsetting the head".
In practice, the required offset is secured in a variety of ways. Some pickup arms describe a gentle curve practically all the way from the pivot to the stylus end. Some arms are straight, with only the head shell offset.
Some arms are straight head shell and all, with only the pickup cartridge inclined within the shell.
As far as tracking is concerned, it is quite immaterial which of these methods is used. They might have varying degrees of aesthetic or "professional" appeal, but the only dimensions that really matter, for tracking, are the distance from stylus point to pivot, the inclination of the stylus support to the line joining the two, and ultimately the position of the arm relative to the turntable.
As we have already said, many other important factors do intrude in the overall design of pickup arms. These include the mass of the arm, its stiffness and the tendency to exhibit internal resonance effects, excited by the vibration from the stylus. Then there is the whole question of dynamic balance, or how sensitive the pickup assembly is to tilt or movement of the motor board as a whole. Mechanical convenience also has to be considered - the degree to which various shapes impede or facilitate particular pivot systems or lend themselves to the receipt of different cartridges.

However, to get back to tracking, as distinct from these other considerations, the angle of offset is not just an arbitrary bend in the arm - or it shouldn't be!
It has been the subject of a good deal of mathematical and graphical analysis, as will be evident from a number of the more advanced audio textbooks.
By assuming that a pickup has a certain dimension between the stylus and pivot points, and that it is to track across a 12 inch disc with a minimum tracking error, it is possible to derive the two remaining vital figures:
(1) The optimum offset angle and
(2) The distance from the centre of the turntable spindle at which the pickup pivot must be located. The distance can alternatively be nominated by the position of the stylus, when swung towards the turntable spindle - whether it falls short of the spindle or rests its centre or overhangs.
In fact, optimum tracking normally involves some overhang of the stylus beyond the centre line of the turntable spindle, as well as a specific amount of offset.
In his book "High Quality Sound Reproduction", James Moir quotes the following mathematically derived figures:

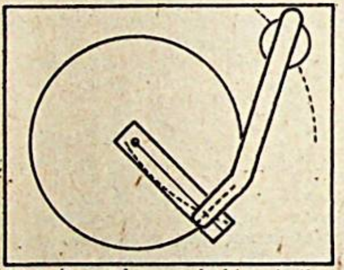
Dimensions, pivot to stylus tip	Optimum Offset Angle	Optimum Overhang inches
7 in	26^{o}	0.6
7.5 in	24^{o}	0.56
8 in	22^{o}	0.52
9 in	20^{o}	0.47
10 in	17^{o}	0.42

One point emerges from this which is not always appreciated: There is not just one point on the motor board which is optimum for the particular pickup. For the deduced minimum tracking error, the pickup pivot can be centred anywhere on a complete circle around the turntable spindle, at any point which will give the required amount of overhang.
In practice, the choice is generally limited to the small arc of the circle which will put the arm into a convenient position for handling, for placing on its rest, and to clear the sides of the enclosing cabinet.
At this point a word of warning might be appropriate. A common trap in mounting tone arms is not to allow sufficient clearance for the arc covered by the rear end. In many modern tone arms, the adjustment of the counterweight involves a rearward movement of the whole tail section, so it is necessary to ensure that the counterweight is in the fully extended position when checking for clearance.
If one could be sure that all pickup arms were designed according to these figures, the story would end right there. It would simply be a matter of measuring the point-to-point distance, assuming the angle to be as quoted and mounting to give the right overhang.
Another complication can arise from the substitution, in an arm, or a pickup cartridge different from the one originally intended. Though most pickup cartridges these days, mount with two screws on half-inch centres, there is no standardisation as to the distance between the stylus tip and these mounting screws.
Taking extremes, it is conceivable that changing a cartridge in an arm could shift the stylus tip relative to the sprindle by as much as half an

inch. Therefore, what was intended as a 7. 5 in arm with 26 degrees offset, could effectively be reduced to 7 in or increased to 8 in, without the facility of modifying the offset angle by the required 2 degrees. Serious tracking error would be introduced if, in addition to this departure from optimum offset, alteration of the stylus position meant that the stylus overhang was reduced to almost nothing or increased to over an inch.

Where the mounting of the arm can be varied, the problem is easier, because the pickup can be shifted bodily to read just the amount of overhang and thus compensate for variations in the stylus position.

Many tone arms purchased separately nowadays come with a card template intended to assit the purchaser in installing the arm correctly, with the minimum of trouble. This template has a hole at one end which is slipped over the turntable spindle, and the centre of the mounting hole which will accommodate the shaft of the pivot is indicated by a small hole, or perhaps a cross at the other end. The proper position of the stylus in the head shell is usually indicated also, and should be observed.



A simple method for locating a tone arm for optimum tracking angle. It requires only a piece of card about 6in x 1in with a hole at one end to take the turntable spindle, and a line down the centre. The method is explained in the text.

Where circumstances leave a doubt as to the proper mounting position for an arm, it is possible for anyone with a critical eye to determine a suitable position visually using only a carefully scribed piece of cardboard, as illustrated.

Take a strip of good quality card, about an inch wide and eight inches long. Draw a clean, bold line straight down the centre. Near one end make a hole, centred on the line, and forming a neat fit for the turntable spindle. Slip the card over the turntable spindle and lightly mark on the line the distance from centre of the inner and out grooves of a full 12 in LP disc.

Now examine the pickup to see whether the axis of the stylus support - and probably the cartridge as a whole - can be related visually with the shape of the head shell

With the pickup supported in a likely tentative position on the motor board, rest the stylus carefully on the line of your piece of cardboard about midway between the marks representing the inner and outer grooves. Looking vertically downward, the line along the pickup head shell and the line on your cardboard should form something very close to a right angle.

Without altering the position of the pickup pivot on the motor board, rest the stylus on points along the line to the outside groove, then to the inside groove. Note how the tracking angle varies from the optimum right angle toward the extremes of groove radius.

By now moving the pickup pivot slightly closer to the turntable spindle or further away - thus varying the amount of overhang - the influence on tracking error can be observed.

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