G. SLOT - FROM MICROPHONE TO EAR

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From Microphone to Ear

MODERN SOUND-RECORDING AND REPRODUCTION TECHNIQUE

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FROM MICROPHONE
TO EAR

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Modern sound-recording and reproduction technique

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BY

G. SLOT

PHILIPS TECHNICAL LIBRARY

Translated by: E. Harker, Mitcham-Junction

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Foreword

The intellectual wealth of humanity appears to know no bounds. Many of these riches, however, have been acquired by the human mind only when human ingenuity provided the means for a large-scale exchange of knowledge. Science could not begin to soar until the art of printing was put to practical use and so enabled great intellectual truths to become common knowledge. At the same time, music and the spoken word remained tied to their original source for four centuries after the freeing of the written word.

Seventy-five years ago, these shackles were also broken.

At first the methods employed to record and reproduce sound were only very primitive and the results analogous to print in newspapers, hastily executed on poor quality paper, here today and gone tomorrow. These methods have improved almost out of recognition in the past decade, however, and are now fully equal to the task of recording for all time the most beautiful of musical creations, which once seemed destined to be heard only by a small circle of regular concert-goers. The result has been a marked and widely evinced growth of interest in the masterpieces of human intellect, as well as in its more playful expressions, showing that technology can enrich, as well as give pleasure to the public as a whole.

As this interest in music and speech recorded on tape and disc has increased, so also has interest in the technology which makes this form of communication possible by spanning time, as well as space, and so preserving words and music for posterity long after the performers themselves have departed this earth.

The writing of this book was prompted by queries on the subject of sound-recording and reproduction from all over the world, not only by sound-technicians, but also by music lovers without technical training whose aim is to reproduce music and speech as faithfully as they can.

This book is intended to answer as many as possible of the questions posed by gramophone owners, and to explain the more important technical problems in relation to one another. An understanding of these problems is essential as a means of recognising the limitations, and more especially the possibilities, of tape and disc. It is hoped that the descriptions of sound-recording and sound-reproducing techniques are such as to be understood by readers without technical training, since it is for them, as much as for trained technicians, that this book has been written.

Of course, the size of the book governs the amount of information contained in it, and for this, if for no other reason, it is impossible to cover every aspect of sound-reproduction.

Nevertheless, every effort has been made to describe satisfactorily what happens to sound between the microphone and the ear.

We wish to thank our colleagues for their support with the writing of this book and Mr E. Harker of Mitcham Junction for his translation.

G. SLOT

Contents

CHAPTER I

FROM TINFOIL TO MICROGROOVE

Section 1. The acoustic period

Although the barrel organ, the musical box and the pianola have all contributed to the evolution of recorded music, they can only produce sounds originally incorporated in their mechanism. Hence the "memory" element in such instruments falls far short of that in a gramophone, which, instead of pitch and rythm alone, records and reproduces an exact replica of any combination of sounds. The gramophone and the tape-recorder have no "voices" of their own, but are able to "play" any musical instrument; no wonder, then, that they have now completely superseded the earlier "music machines". After 75 years of evolution and revolution, they have now been perfected to the point where the listener can identify, say a particular singer, or the maker of a violin, merely from the sound reproduced.

However, the original recording instruments were far below this standard. The first practicable one was demonstrated by Edison. His original „Phonograph" was in essence a cylinder covered with tinfoil. The sound track was engraved in the foil, and the cylinder was rotated by hand. The reproduction, poor in itself, was further impaired by the speed variations arising from this method of rotation; in fact, the Phonograph of 1877 was not a great success. Though Edison was probably too engrossed in the development of his incandescent lamp to pay much attention to the Phonograph, Chichester Bell and Sumner Tainter carried it a stage further by substituting a wax cylinder for the tinfoil. This was a considerable step forward, enabling the "Phonograph" and the Bell and Tainter "Graphophone" to become the basis of the present-day trade in "canned music". Of course, the quality of the reproduction then fell far short of the present standard. The noise-level was high, the dynamics and the frequency range were severely limited, and the sound was harsh. Although it was at first possible to record two, and later four minutes of music on a single wax cylinder,

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Fig. 1
The original Berliner gramophone.

Fig. 2. Edison gramophone.

no method of multiplying the recordings was then known. Moreover, the sensitivity of the recording equipment was so inadequate that on the one hand a singer had almost to stick his head into the horn, and on the other the volume of sound reproduced was not really sufficient for an ordinary sitting room.

A notable drawback of these early days was that the soft material, i.e. wax, required for the recording was also used for the reproduction; hence the resistance to wear was very low. This particular problem, however, was solved in 1888, when Berliner demonstrated his "Gramophone"; he etched the sound track through a layer of wax into zinc, thus introducing a hard material as the playing medium. Berliner employed flat round discs, similar to the modern record, and made use of the sound track itself to move the transducer over the disc, instead of a special threaded spindle, as in the Phonograph. On the other hand, the needle-scratch in the first gramophones was very loud. Nevertheless, Berliner continued his work and, only three years later, evolved a method of multiplying recordings, which is essentially the same as that nowadays employed. One of his most difficult problems was to find a material suitable for pressing. „Durenoid", a shellac compound used to make buttons, proved most suitable for this purpose. The clothing industry made another contribution to the development of gramophones when Johnson, an engineer in a sewing machine factory, designed the first practical clockwork gramophone motor.

The year 1901 was in many ways a very important one for the gramophone industry, since an exchange of patents then resulted in a considerable improvement in the quality of the machines; moreover, is was then that Caruso began to make records. Although the gramophone of those days was not very different from the modern one, the "Phonograph" continued to dominate the market for many years, probably owing to the fact that the wax cylinders could be played with very much less needle scratch than the far from perfect gramophone records of those days.

Section 2. The electric period

Now, despite all the improvements so far described, the quality of reproduction remained low; since the best results were obtained from vocal recordings, they

Fig. 3. Recording in the acoustical period.

constituted the bulk of the record programme. The development of broadcasting by radio in the years immediately after 1925 was originally seen as a serious threat to the gramophone industry. As quite often happens, however, the new proved to be the friend, rather than the enemy of the old. Microphones and amplifier valves, together with magnetic cutters, enabled manufacturers to improve the quality of their records considerably, and with the advent of the magnetic pickup, the gramophone entered a new era. By virtue of these improvements, it was then no longer necessary to restrict recordings to the voices of operatic and music hall singers; then, as now, any piece of music could be recorded and reproduced satisfactorily. Record playing was made easier by the invention of the first automatic stop, the substitution of an electric motor for the original clockwork further improved matters, and in the nineteen thirties, when the first record-changers were introduced into the home, the playing of a complete opera really became a form of relaxation rather than a species of violent exercise.

It can be stated that the best, at least, of the gramophones produced in the years immediately before and after the second world war reproduced a reasonably wide tonal range, i.e. from 100 to 5000 c/s. The pickups then employed were relatively light, with needle pressures in the region of 1½ oz. (40 grams) as compared with 5 oz. (150 grams) in 1928: this enabled the amount of slate included in the record material, and therefore the needle scratch, to be reduced. Hence pianissimo passages could then be recorded more softly than before; moreover, improvements in the recording equipment gave scope to a relatively louder fortissimo, thus enhancing the dynamics of the music. Nevertheless, at 25 db this still remained considerably below the concert hall level, i.e. 60 db.

Between 1940 and 1945, important technical discoveries were made, including progress with plastics and with magnetic sound recording which proved very important

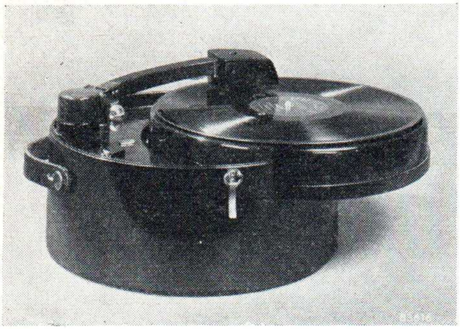


Fig. 4. "Porteldisc" electric gramophone, made in 1937.

to the gramophone. Further development of the crystal pickup enabled the needle pressure to be reduced to 1 ounce (25 grams), that is, a pressure low enough to make possible the use of records pressed from a certain type of vinyl compound. This material is grainless, therefore records made from it are virtually noiseless; moreover, it is unbreakable and does not wear unduly provided that it is used in conjunction with a light pickup. For general purposes however, the price of this material was prohibitive. The year 1945 saw the beginning of an all-out effort to improve the recording characteristics, which has since proved particularly successful in raising the standard of treble response.

In 1947, Philips produced the first feather-weight pickup (known in the laboratory as the "blacking tin"): this was the first light-weight pickup without balancing. Experiments with microgroove records were first carried out before the war, but the technique was not then far enough advanced for practical application; sufficient progress had been made by 1948, however, and by a fortunate coincidence the first wire and tape recorders for amateur use came on the market in the same year, their development having started as far back as 1890. Although at first widely regarded as a serious threat to the gramophone, like broadcasting, magnetic recorders eventually proved to be just the opposite. The tape recorder enabled long recordings to be made continuously, any imperfections being afterwards deleted from the tape, and it was found possible to copy such recordings on to gramophone records without impairing the quality of reproduction. In fact, this recorder emerged as an essential tool in the manufacture of long-playing records.

Now, with the advent of long-playing records in 1948 it was no longer necessary to reject the new noiseless record compound as being too expensive, since in view of the rather high price of records containing up to 45 minutes of music the fact that this compound makes them unbreakable then became very important.

The development of new recording heads and amplifiers, as also

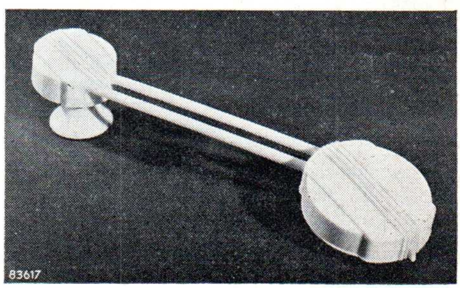


Fig. 5. The first featherweight pickup, type 2962.

the use of heated cutting styli, improved the quality of the records more and more, this improvement being reflected particularly in an undistorted reproduction of even the highest of notes. Another improvement arose from the introduction of the principle of variable groove spacing, enabling the playing time to be still further extended and the dynamics enhanced.

Sapphire needles, employed in most gramophones from 1947 onwards, and diamond needles, more and more common in recent years, together with the fact that the needle pressure is now extremely low, namely $^1/_3$ oz. (10 grams), have so reduced record wear as to justify the claim that, in general, the modern gramophone record offers virtually endless repetition of music reproduced with a fidelity hitherto unknown.

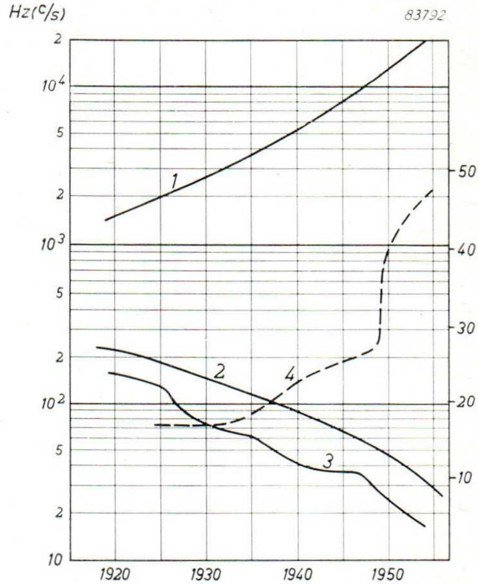


Fig. 6. Progress chart 1920-1955. 1) Treble response. 2) Bass response. 3) Distortion. 4) Dynamics

How this standard of reproduction is attained, and the conditions which the different components of the gramophone circuit must satisfy to ensure it, will be seen in the following chapters.

CHAPTER II
FROM HERTZ TO GRAMOPHONE

Section 1. On hertz and the decibel

Since the cardinal feature of any gramophone is the sound produced by it, this will be considered first. Sound embraces all the vibrations perceptible to the human ear, that is, mainly air vibrations; unless otherwise indicated, then, we shall refer to none other than air vibrations. Now, most musical instruments include a vibrating part, e.g. the skin of a drum or a string; it causes the air to vibrate, the vibration being propagated through the air. For example, the skin of a drum rises at a given moment, pushing air particles away from it; the motion thus imparted to the particles is then propagated. A moment later, the skin descends, sucking air particles towards it, this motion is likewise propagated through the air. Since the velocity of vibrations propagated in air is very specific, i.e. 1000 ft/sec (343 m/sec) the pulses of compression and rarefaction travel one after another in a continuous train; hence sound is propagated in the form of waves as will be seen in fig. 7. To avoid misinterpretation, it should be noted that the air particles themselves merely move to and fro; in other words, it is the pulses of the particles, not the particles themselves, that travel from sound source to listener.

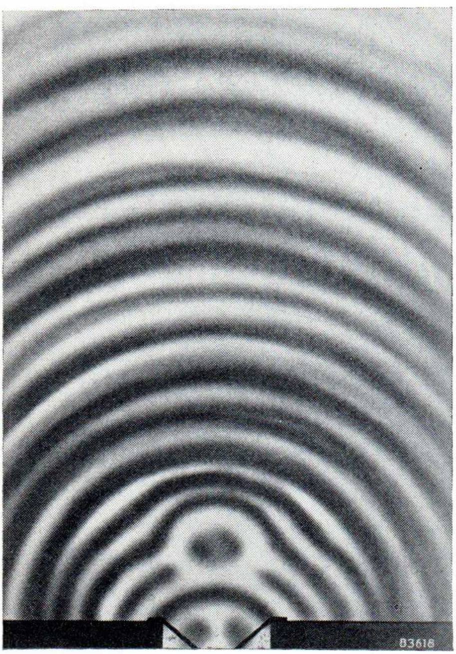


Fig. 7. Sound waves.

Given a drum or a string vibrating very rapidly, that is, producing a high note, the distance between the successive waves will be shorter than for a base note; this distance is known as the wavelength. Another measure of the pitch of a sound vibration is its frequency, or number of vibrations per second; the frequency of, say, the note A is 440 vibrations per second, or 440 c/s, and its wavelength 2½ feet (78 cm), wavelength being the velocity of propagation divided by the

frequency.
Let us consider one of these vibrations, namely what is known as a pure tone (without overtones); it varies in the manner indicated in fig. 8. The air particles at first accelerate in a given direction, and then gradually decelerate; at a certain moment they reverse direction, accelerate again, then decelerate, reverse direction again, and so on. Such a fluctuation is described as a cycle; hence it is usual to refer to the number of cycles/second rather than the number of vibrations per second.

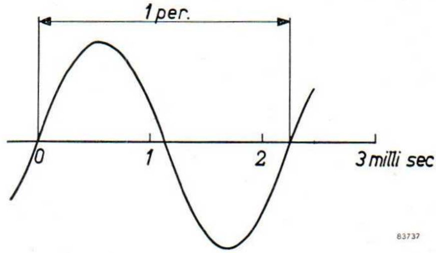
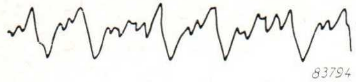


Fig. 8. Graphical representation of a pure vibration.

However, when considering the reproduction of music we are concerned not only with the tone, but also with the intensity of sound vibrations, or, more precisely, with the relative intensities of different notes, since the general sound level can usually be varied at will by means a volume control. In acoustics, the usual measure of relative sound intensity is the decibel (db). It is equivalent to 10 times the logarithm of the ratio of sound intensities, or 20 times the log of the ratio of sound pressures, at a given point. An increase in volume (or in the output power of an amplifier) by a factor of 2 is equivalent to a gain of 3db (10 log 2), and a 4 times increase in volume to a gain of 6 db (10 log 4). The corresponding increases in sound pressure (or in the output voltage of a pickup or amplifier) are factors of 1.4 and 2, respectively. One decibel is roughly equivalent to the smallest difference in sound intensity perceptible to the human ear. Now, the ear is a very sensitive organ, capable of perceiving a sound pressure as low as 0.0002 milligrams/sq. cm. The highest sound pressure to which it can be subjected without pain is 200 mg/sq. cm, that is, a pressure one million times, and a volume one billion times greater than the above-mentioned minimum. Accordingly the threshold of pain is about 120 db above the threshold of audibility. Orchestral music may well involve contrasts of roughly 60 db between pianissimo and fortissimo, the one then being, say, 20 db above the threshold of audibility (roughly the level of a whisper), and the other 40 db below the threshold of pain, that is, more or less the same volume of sound as will be heard by a passenger in a railway carriage travelling with the windows open.

The vibration shown in fig. 8 is a pure tone, virtually non-existant in music, although some of the lower flute notes are almost pure tones. Musical notes comprise a fundamental and several overtones. The overtones impart an inherent quality, or timbre, to the music and vary as between different musical instruments. Now, the frequency of the overtones is generally a multiple of the fundamental, that is, a factor of 2, 3 or 4 higher than the latter. The number, intensity and manner of appearance and decay of the overtones govern the timbre, and it is owing to them that, say, an oboe sounds different from a violin. Although music would be very



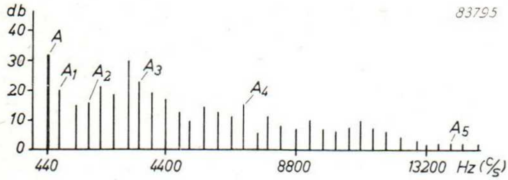


Fig. 9a. Oscillogram of a violin note.
Fig. 9b. Fundamental and overtones of a violin note.

much easier to reproduce if it had no overtones, it is unlikely that it would then interest anyone, since a symphony orchestra playing such music would sound exactly like a number of rythmically tuned morse transmitters.

It will be seen from fig. 9a, showing the vibrations of a violin note, that it is very much more complex than the vibration shown in fig. 8. Fig 9 b depicts the same note in a different manner; here, the vertical lines represent the fundamental (A) and the different overtones, the length of each line being proportional to the intensity of the particular tone. It will be evident that although it is the fundamental that governs the tone, the overtones are nevertheless very important in the reproduction. The violin, for instance, produces an unusually large number of strong overtones, whereas the sound of the oboe contains less than half as many.

The fundamentals of different musical instruments range from 16 to 4000 c/s (c_4 to C4), therefore a gramophone capable of reproducing this range does not lose a single **note;** to bring out every **tone** clearly, however, it must also reproduce all the overtones, and to do so must have a range extending to at least 16000 c/s.

Since the human ear cannot perceive tones higher than 16000 c/s, it is not strictly necessary to extend the range of reproduction beyond this point, although it is desirable to do for high fidelity reproduction.

Section 2. Principles of the gramophone

Given the above information, the gramophone itself will be very much more readily understood. The principles of it will now be explained.

Assume that we have an orchestra playing in a hall with only one small window, and that this window is sealed with a diaphragm, say, a thin sheet of mica. Sound vibrations striking this mica will cause it to vibrate, and so excite similar vibrations in the air outside the hall, enabling the music to be heard there, as well as within. Under ideal conditions, the movements of the diaphragm will precisely match the sound vibrations produced by the orchestra. Therefore, given the possibility of noting the movements of the diaphragm exactly, in the manner shown in fig. 9a, and later reproducing exactly the same movement of the diaphragm by referring to these notations, it would also be possible to produce music identical with that played by the orchestra.

This, in essence, is what happens in a gramophone.

In its most rudimentary form, this was simply a horn, sealed at the narrow end with a piece of mica and directed towards the sound source. A needle secured to the mica cut a track in a wax cylinder, or wax plate, this track corresponding exactly to the sound vibrations actuating the diaphragm and the needle. To play the "record", this process was reversed, the sound track then being moved past the needle, causing it to vibrate; the mica then vibrated with the needle, and a more or less true reproduction of the original tones came from the horn. Fig. 42 is a photomicrograph of such a sound track.

As stated in Chapter I, the actual recording is nowadays followed by a copying process; moreover, the recording itself is electrical. However, although this has made matters very much more complex, the original principle has of course been retained — also in tape recorders, however remote they may seem from Edison's invention.

CHAPTER III
FROM SOUND TO RECORD

Section 1. From microphone to ear

It always fascinates the writer to think that a sound, engraved in solid matter and after a series of most complex processes, can be reproduced at any moment and with such fidelity that the hearer almost forgets that he is not actually in the concert hall itself. In fact, the precision with which sound vibrations are "frozen" in a gramophone record is virtually without parallel in other branches of industry.

That the process is undoubtedly complex will be seen from fig. 10, which shows, in a simplified form, the long journey of recorded sound. Sound vibrations are intercepted by a microphone (1), which converts them into voltages. These voltages are strengthened in an amplifier (2), which then feeds them to a tape reorder (3). Here, the sound is preserved in the form of zones of different magnetic flux on a long tape coated with iron powder. The tape is played back on a play-back machine (3), the vibrations recorded on it being thus reconverted into actual electric vibrations to actuate a recording head (4), which engraves the sound vibrations on a disc known as the "Master". Impressions are taken from the master by a plating process (5), and these go to the pressing shop (6) to produce the records played by our pickup (7) at home. The waves on the record are converted by the pickup into electrical vibrations, and the amplifier (8) and the loudspeaker (9) do the rest — all that we have to do is listen.

Not very much need be said in this chapter about the microphone, since it operates on the same principles as the pickups, which will be described in Chapter IV. The functions of the two are very closely related; in fact, the Scandinavian name for a pickup, that is, "needle microphone", is really very appropriate. The tape recorder, and the amplifier, as also the principle of the recording head, are likewise described in other chapters; the actual cutting of the record, and the processes which follow it, will now be described.

Section 2. Cutting

Whereas a fairly hard material is preferred for the gramophone records, very soft materials are used for the masters. Originally, the master was cut in a thick slab of

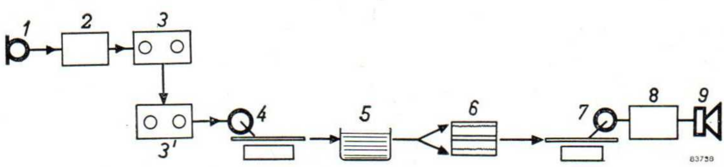


Fig. 10. From microphone to ear

wax, suitably smoothed and flattened by turning, but this has now been superseded almost entirely by what are known as acetates. They are flat metal plates, thinly and very evenly coated with acetate. Although liquid at relatively high temperatures, this sets at just above room temperature. Since the groove cut in such a lacquer disc is very fine indeed, the layer of acetate must conform to very stringent requirements... even the smallest of pores or irregularities in it would cause intolerable hissing or clicking noises when the record is played. To produce really high-quality records, moreover, the surface of the lacquer disc must be perfectly smooth. Although the surfaces of those lacquer discs from which no matrices are to be taken are subject to less stringent requirements and may therefore contain minute waves, those which are to be used for copying, that is, masters, would be no use whatever if they exhibited the slightest trace of "orange-skin effect", and are therefore mirror smooth.

When a gramophone record is played, the pickup is guided over it by the spiral groove. During recording, however, there is no groove to follow; hence the recording head must be so controlled as to describe the proper spiral track without it. This is done by attaching the recording head to a threaded spindle rotating in such a way as to move it gradually inwards from the periphery of the revolving disc. Now, in some micro-groove records the distance between the grooves is a mere 1/300 part of an inch (0.08 millimetres) and since the grooves themselves, however fine, nevertheless occupy a certain amount of space, and may not, in any circumstances, cut, or even touch one another, the guiding mechanism of the cutter must of course be very accurate. Any play in this mechanism or variation in its speed, however small, will inevitably produce faults serious enough to necessitate the rejection of the particular record. Again, any vibration of the mechanism may spoil the groove. A droning noise during the playing of a record is also evidence of such vibration during the cutting. These requirements, already difficult enough to fulfil, have been augmented in recent years through the introduction of variable groove spacing. Even in the earlier 78 r.p.m. shellac records, the distance between grooves was not invariably the same, as will now be explained. About 4½ minutes of music can be recorded on one side of a 12″ record. However, if the playing time of the piece of music to be recorded happens to be, say, only 4 minutes, the manufacturer will nevertheless cut the groove so as to cover the entire playing surface, since a record having a large plain zone in the centre will not sell readily. To do so, he slightly increases the space between the grooves, accomplishing this by varying the gears between motor and lead screw. Now, the opposite of this problem, that is, when the particular piece to be recorded is slightly too long for the record, is more difficult.

Here, the grooves must be cut closer together, involving a risk that they will merge during the louder passages of the music. This problem was solved at one time by attenuating the louder parts of the music slightly, but this impairs the dynamics of the composition, an effect which must be avoided as far as possible. The obvious course would be to employ a wider groove space during the louder passages, and a narrower space during softer ones, but this cannot be done since it is not possible

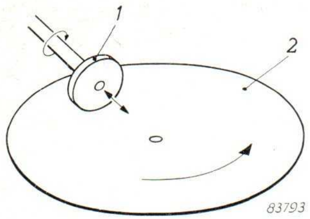


Fig. 11. Spindle drive for variable margin control

to change the gears during the cutting of a master. However, a number of methods of varying the speed of the cutter continuously have been evolved in recent years. One of them is to replace the driving gears of the lead screw with a rubber-tyred wheel (1 in fig. 11) resting on a disc which rotates at a constant speed (2). Wheel 1 can be shifted along the spindle, its speed of rotation then depending upon the point at which it touches the driving disc: the closer the wheel to the centre of the disc the lower its speed. Accordingly, the speed of the recording head, and therefore the groove spacing, can be varied very smoothly by shifting the wheel.

The only other difficulty is that it is necessary to know well beforehand when a loud passage will occur in the music, since the groove spacing must be widened, of course, to accommodate the fortissimo parts, and this must be done very gradually to avoid noise. Although it would be possible to effect the necessary variation at the correct moments by referring to the score, it would place too heavy a burden of responsibility on the technician cutting the record, since he is already fully occupied with other problems. However, since masters are now invariably copied from tape recordings, the movement of the recording head can be controlled automatically in the following manner. The recording amplifier is connected to a play-back head. Another play-back head is placed a certain distance from the one connected to the recording amplifier, and the voltage from it is applied to a special amplifier, which converts this voltage and passes it to a unit automatically regulating the position of the wheel on the guiding spindle. A moment before any strong signal reaches the recording head, wheel 1 shifts towards the periphery of the driving disc (2), returning automatically towards the centre of the disc after the particular fortissimo passage. This is only one of many possible methods. Another is to drive the guiding spindle by means of a synchronous motor supplied by a generator whose frequency is proportional to the voltage on the auxiliary play-back head. Of course, the method of varying the groove spacing does not really interest the ordinary record lover, who is concerned primarily with the fact that it gives him records with maximum playing time and full dynamics.

Records with variable groove-spacing, or margin control as it is also called, are easily recognised owing to their characteristic light and dark surface rings; although these rings do not give the record a very attractive appearance, they certainly enhance its most important feature, the tone, very considerably. Of course, this system is employed only where it is most effective; not, say, to record a simple violin solo.

During the cutting of a record, the groove is continually scrutinized through a microscope, (fig. 12), a very necessary precaution, since the width and depth of the groove must be maintained perfectly uniform. Moreover the thread of material cut from the disc must be prevented from becoming entangled with the cutter, since in so doing it would damage the groove; also, the edges of the groove must be perfectly even, that is, free from the burr occasionally formed owing to the pressure of the cutter. Again, the recording technician must take care to avoid overcutting the grooves.

In fact, the above-mentioned thread (or swarf) precents quite a problem, as all amateur record makers know to their cost. In amateur record cutting, a camel hair brush is used to guide the swarf away from the cutter, and in professional recording studios it is sucked away by means of vacuum pumps, the hoses of which will be seen in fig. 12. Despite these precautions, however the swarf remains a nuisance owing to the fact that is collects an electrostatic charge and so tends to cling to everything it touches; hence the studio professional also prefers to have a brush handy.

Although the cutter itself is only a tiny piece of sapphire or diamond, many highly technical articles have been written about it. The shape and accuracy of finish of this minute tool are very important in that they govern the smoothness

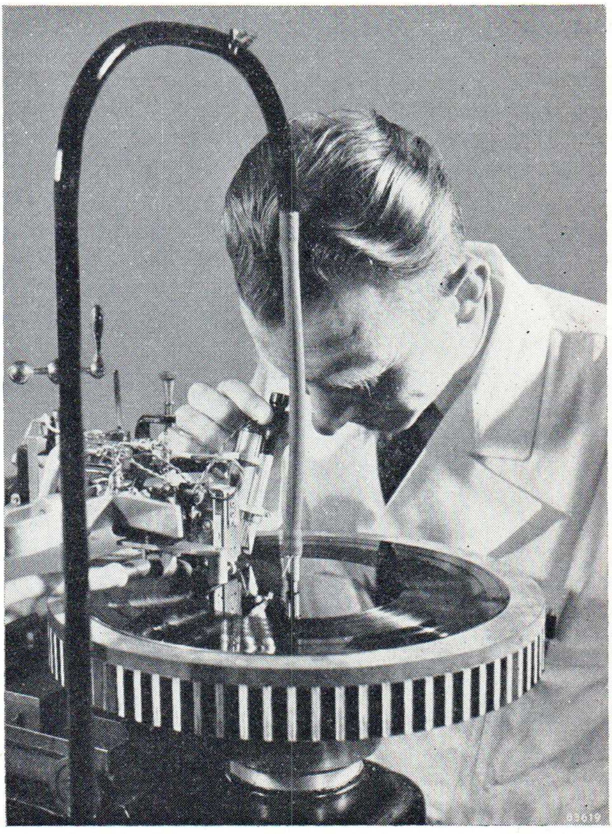


Fig. 12
Cutting the lacquer plate

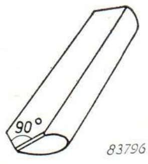


Fig. 13. The cutting stylus

of the finished groove. As will be seen from fig. 13, the tip of the cutter is flat: the effect of this upon the reproduction, in which — to avoid undue record wear — the record is played with a round needle, will be explained later.

Recent improvements in record quality are largely attributable to the use of improved cutters, as also to what is known as the „Hot Stylus". It goes without saying that a heated stylus cuts through the lacquer more easily and cleanly than a cold one, and therefore gives very much better results, especially as regards the higher frequencies (short wavelengths); also, it cuts a smoother groove and so minimises record noise.

Now, it is not practicable to heat the whole surface of the record, since that would only melt the laquer; the art is to heat the cutter just enough to soften the lacquer at the point of contact, but enable it to harden again as soon as the cutter moves on. The stylus may be heated, say, by means of an electric spiral. However, since it must not heat either the adjacent recording head or the record as a whole, and the distances between the three are only a matter of millimetres, the "Hot Stylus" is more easily described than constructed. However, it amply repays the trouble of making it, since a record cut with an ordinary cold stylus is invariably lamentably short of treble overtones, and roughly 15 db noisier than one cut with a hot stylus.

The cutting of the starting and stopping grooves is relatively simple, since it is done with the aid of special equipment; it is also worth mentioning that the lacquer plates are invariably taken a size larger than the actual record, in order to leave an outer rim on which to adjust the cutting head before the rcording takes place, as also to enable the record to be gripped during subsequent operations. Lacquer plates 16" in diameter are usually employed.

Before the advent of the tape recorder, record cutting was a very wearisome business, since the slightest error in direct recording meant that the artists had to repeat their performance from the beginning.

The same applies, of course, to the tape recorder, but it is a patient instrument, always ready to give a repeat performance without complaint if one of the lacquer plates goes wrong; hence it relieves the recording technician of a great deal of nervous strain and enables him to concentrate more fully on his task. Moreover, it drastically reduces the expense of re-cutting a record, so that recording studios are more inclined to do this now, even for minor errors, than when it involved keeping a hundred or so musicians wasting valuable time in the studio.

Section 3. From Master to matrix

The next step in the manufacture of gramophone records is to make the matrix. This is done by the plastic electroplating method, that is, placing the Master, or one of the subsequent phases, in a chemical solution and passing an electric current through the solution to deposit metal on the recording. Firstly, then, it is necessary to make the Master conductive. At one time, this was done by applying a very thin layer of graphite (i.e. so thin as to leave the shape of the groove almost unchanged) to the wax recording; metal was then deposited on the graphite to form a layer of the required thickness, which was then removed. A matrix, or negative impression of the master (groove instead of ridge and ridge instead of groove), was thus obtained. This method has now been improved, but at the same time has become very much more complex.

The structure of the top layer produced by coating with graphite was too coarse and therefore added considerably to the record noise. Nowadays, the laquer plates are dipped in tin chloride solution for a moment or two, and rinsed immediately afterwards so that only a very thin coating of tin chloride is formed on them.

Next, the Master is sprayed with silver nitrate solution; a chemical exchange then takes place as between the tin on the Master and the silver in the solution, and in next to no time a clean, shining coat of silver, only a few molecules thick, forms on the surface of the plate.

Another method is to place the Master between an anode and a gold cathode in a vacuum chamber and apply roughly 4000 Volts D.C. between the two electrodes. The discharge thus initiated deposits molecules of gold on the Master. This process of coating with gold was very popular a few years ago, but owed its reputation chiefly to an association of ideas, namely "gold is expensive" and "expensive is good". Experience has now shown that the silver nitrate process is no less effective and very much more convenient in practice; hence it is now employed by almost all record manufacturers. The silvered plate is transferred straight from the solution to the rinsing spray to remove the surplus nitrate; it is then ready for the next process, which is the application of a reinforcing layer of metal to the silver film by electroplating. Either nickel or copper may be employed for this purpose. The plating process must be carried out very gradually at first to give the deposited metal a very fine structure, but may be accelerated when once the layer reaches a certain thickness. As

Fig. 14
Making a "father" in a plating bath

soon as the metal backing is strong enough, the disc is withdrawn from the bath and rinsed again; a hammer and a special chisel are then used to separate the metal copy from the lacquer disc, which usually means breaking the latter. The metal copy is usually called the Father.

The next step is to remove the rapidly oxidising layer of silver from the Father; being very thin, it can be removed without impairing the quality of the recording. The Father is then passivated, that is, so treated as to enable metal to be deposited on it without adhering to it. Although is would be possible to employ the Father itself as a matrix, this would necessitate cutting a new Master whenever the Father is damaged or shows signs of wear, which would be most uneconomical.

Accordingly, another metal copy, very appropriately described as the Mother, is obtained from the Father in the same way that the latter is taken from the Master; in this case, however, metal is deposited on a negative copy (the Father) instead of on the positive original (the Master). When strong enough, the Mother (at least in the record factory) is readily separated from the Father to give us an exact, playable copy of the Master; it is, in fact, played back by means of a pickup, and also examined through a microscope, to enable the quality of the recording to be assessed. The Mother is then returned to the bath and another layer of metal is applied to it in the manner already described. The copy thus obtained is employed later as the actual matrix, but in its original form it is referred to as the Son.

Now the Son, like the Father, is negative; hence it cannot be played back, but can be used to produce positive impressions in a plastic compound. It is reinforced by applying a thin film of chromium (a hard metal) to it so that it will wear longer and be less readily damaged.

The Father goes into storage when once the Mother is made, but the latter has work to do, being required to produce a new Son whenever the existing one becomes worn. The Father is required only if the Mother is accidentally damaged or is itself subjected to excessive wear, that is, in the case of an unusually popular recording.

What is not implicit in the above, rather matter-of-fact description is that this is really one of the most spectacular phases in the manufacture of gramophone records. The chemical solutions in the plating baths assume colours of surprising beauty and intensity, which, combined with the continual agitation of the liquids to ensure an even deposit of metal, produce an effect quite beyond the scope of mere photography and really worthy of the most talented artist.

In the factory, however there is no time to admire this effect, since the Son must now be made into a matrix. The first step is to smooth the back of it so that it will fit unerringly in the press. The special lathe employed for this purpose must accommodate Sons up to 12″ (30 cm) in diameter, and therefore seems very much larger than is consistent with the desired accuracy. Accordingly, the turner must be not only skilful, but also careful to look after his machine.

Next, the centre-hole invariably so distorted during the preceding operations as to be unusable, must be re-bored. This involves centring the Son, by now for all practical purposes a matrix, on a special machine; in effect, then the centre of an

eccentric spiral, that is the groove, must be determined accurately to within 1/250 part of an inch (1/10 millimetre) to avoid wow or variable noise during subsequent reproduction. It will be evident that this is not easy, and in fact it is even more difficult than the above description suggests.
The method employed will now be described. A needle, connected by a lever to a pointer, is drawn along the groove of the matrix, causing the pointer to oscillate. The control knobs of the centring machine are then manipulated until the pointer becomes fairly steady, usually after several attempts. As soon as it is steady enough, a dimple is punched at the centre of the matrix. Next, a hole, centring upon the dimple, is made in the matrix and a block carrying a pin of the correct diameter to punch the centre hole in the record is inserted in the hole. This method enables the pins, which wear more rapidly than the matrix itself, to be readily replaced. Without this facility, there would be a risk that some of the records produced would have too small a centre hole and therefore would not fit on the spindle of the turntable.

Section 4. Pressing the record

In principle, the process as so far described is the same for microgroove, as for standard groove records, albeit that the former must be manufactured even more carefully, at all stages, than the „coarse" standard records. However, 78 r.p.m. records are made of what is known as shellac, and microgroove records of a certain type of vinyl. Shellac records are pressed from a compound really containing one or two other constituents, such as Copal (a hard, natural resin also widely used as imitation amber), stearine and carbon black (very finely divided soot) from which the record derives its colour, as well as the actual shellac, that is, the secretion of an insect common in many parts of India.
The greatest care is exercised in the grinding, sifting and mixing of the materials; these processes are amongst the noisiest in any industry. In fact, the noise of the cyclones is so deafening that visitors to the grinding shop should really be given ear-plugs before entering, and aspirin after leaving, it.
Unwanted gramophone records, usually rejects and scrap from the

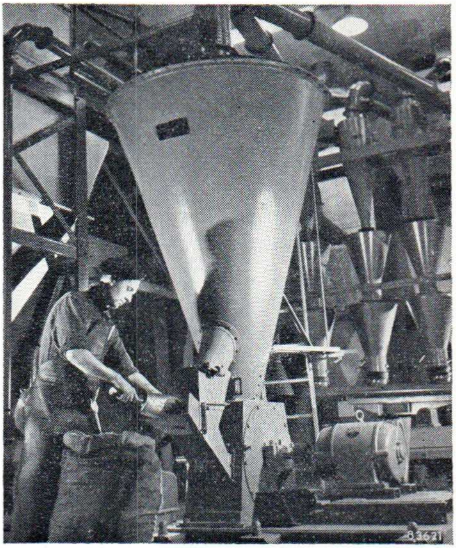


Fig. 15. Grinding and sifting the record materials

pressing shop, are ground with each fresh batch of raw materials. When mixed thoroughly enough, the raw materials are heated until reduced to the consistency of dough, then moulded thoroughly and taken in layers from a huge mechanical "rolling pin". These layers are transferred to a long table and divided into slabs, each large enough for one record. Immediately before being pressed, each slab is reheated and moulded into an oblong lump, known as the "loaf", albeit a black loaf which only the record press can "digest".

The principal material in microgroove records is "Vinylite", supplied to the factory as a white powder; the carbon black already referred to is also employed. These materials are treated in very much the same way as those of shellac records, but experience has shown that greater uniformity is ensured by pressing vinylite records from grains, or "pellets", instead of loaves; special machines are employed to make large quantities of these pellets.

Because they are made of almost pure vinyl, microgroove records are flexible and, in fact, practically unbreakable; moreover, they have a very low noise level. However, this material is too expensive to be used for ordinary 78 r.p.m. records. To complete the picture, it is also worth mentioning that another Vinyl compound is employed instead of shellac for 78 r.p.m. records, where the latter material is not readily obtainable. Owing to the composition of this compound, however, the properties of records made of it are very much the same as those of ordinary shellac records, that is, they are relatively brittle and have a rather high noise level, the ultimate quality of the recording depending very largely on the percentage of admixtures, whereas in good microgroove records this percentage is maintained as small as possible.

Shellac records and microgroove records are made on presses of the same type. Two matrices are employed, the one fixed to the bottom table of the press and the other to the, movable, top table. Both matrices are steam-heated and water-cooled. Now, the operator places a label in the bottom matrix, a pre-heated loaf on the label and another label in the top matrix. He then applies steam pressure to close the press, which softens the loaf, or the pellets, so that the compound spreads over the whole surface of the matrix and penetrates into all its grooves. After a certain time, he cools the matrices and opens the press, revealing the record almost ready for use. All that is then necessary is to remove the flash and give the rim a smooth finish. It is impossible to recognise the type of record being pressed merely by looking at the press; however, the pressure, pressing time, cooling time and temperatures involved in the manufacture of microgroove records are different from those employed for ordinary shellac records. The accuracy of the different cycles of the process affects the quality of the records very noticeably; hence they are all controlled automatically.

The presses are fed and opened by hand, however, since a machine to perform these tasks would be so complex that the associated costs would far outweigh those of employing an operator. Nevertheless, the task of adjusting the press to bring the two matrices into exact register is no sinecure; it requires much skill and takes so much time that the pressing of small series of records is impracticable from the point of view of economy, much to the regret of collectors, who deplore the fact that the less popular records are invariably withdrawn from circulation.

Although occasional mistakes by the operators are inevitable, it does not often happen that records are sold with the wrong labels, or with the labels reversed. This is because all the records are inspected carefully before being dispatched, and any copies with visible faults are rejected and sent to the grinding shop to be pulverised and mixed with another batch of raw materials. Not all of the records are actually played before being dispatched, however; the pressing time is very much shorter than the playing time, especially with long-playing records, whose playing time may be forty times as long as the time required to make them. To employ 40 inspectors for each press would increase the price per record beyond all reason; instead, then, it is usual to take random samples large enough to enable any defects discovered in them, and likely to occur also amongst previous copies not inspected, to be readily eliminated from the latter.

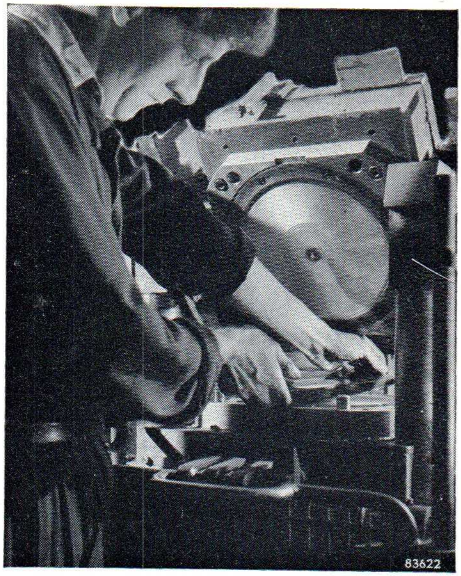


Fig. 16. The record press

However, given suitable precautions during the actual manufacture of the records, the number of serious defects will be very small; at any rate, most of them are detected by the highly experienced mechanical inspectors before reaching the music test.

Now, there is often an interval of several months between the making of the tape-recording and the pressing of the first record; nevertheless, from the very first moment the sound is preserved so perfectly that the tape can be stored almost indefinitely without losing anything of the original character of the recording. Also, despite the series of complex processes involved, the overall loss of quality as between the original tape recording and the record itself is so small as to be only barely detectable even with the best of equipment, and quite undetectable with any reproducing system not in the very first grade. In effect this means that the sound track on the record differs less than 1/10 micron, or less than 1/250.000 part of an inch from that on the Master; hence it will be seen that the reference to extreme accuracy at the beginning of this chapter is well justified.

CHAPTER IV-A

PICKUPS — OPERATING PRINCIPLES

Section 1. General requirements

The function of the pickup is to convert the sound vibrations impressed in the record into electrical vibrations. These sound vibrations are "frozen" in the record in the form of a continuous groove, whose variations cause the gramophone needle to vibrate as the record revolves past it. In essence, then, the task of the pickup is to convert these needle vibrations into electrical ones; hence it is essential that the movements the needle correspond as closely as possible to the original cutter movements. Now, apart from certain inherent limitations, to be discussed later, owing to which the two vibrations cannot be made to agree exactly, the translation of the groove modulation into needle motion may be affected also by factors within the pickup itself. Hence the manner in which the needle is coupled to the pickup is highly important, so much so, in fact, that the two should invariably be considered together, not as individual components.

Primarily, then, a pickup should convert mechanical vibrations into electrical ones with absolute fidelity, that is, without adding anything to the pattern of vibration, since this would produce distortion or background noise, without overemphasizing any part of it, and so giving certain notes undue prominence, and without suppressing part of the pattern, thus altering the timbre of certain notes or even eliminating them altogether.

Also, pickups should be reasonably efficient in effecting the conversion (i.e. the output voltage should not require undue amplification) and should effect it without producing undue record wear, since a record should always outlast its appeal. This means that classical records, in particular, should be almost immune from wear under normal conditions. Other requirements imposed on pickups are that they should be so robust that they need no special protection and only a minimum amount of maintenance, that is, the needle should be long-lasting and, even more important, the pickup itself should not require overhauling within a short time of being put into service; although such overhauls might be necessary for some of the pickups employed in broadcasting studio's, they would be most inconvenient to the ordinary user.

To end this list of requirements, it is worth mentioning that the price must be as low as possible; this, of course, is inconsistent with all the above-mentioned requirements. Accordingly, it is seen that the development and manufacture of pickups involve many problems; they have been solved in many different ways; in fact, almost every known method of converting mechanical vibrations into electrical pulses has been applied to pickups at one time or another.

The principal types of pickups — in order of their popularity — are:

the crystal (piezo-electric) pickup;

the magnetic pickup;
the dynamic pickup, and
the condenser pickup.

Photo-electric, variable-resistance, and electronic pickups are also made, but at present so rarely that it is not worth while discussing them.

The first of these, that is, the crystal pickup, is generally the least expensive, and therefore the most popular of all. The popularity of the others is governed mainly by the price. However, leaving popular opinion out of account, since it is usually based on experience with older models whose quality is by no means the best attainable, the writer considers that very good results can be obtained with any of the pickups here discussed and that, in principle, there is little to choose between the different methods of converting mechanical into electrical vibrations. Although one or two or the crystal pickups now on the market give better reproduction than other, notably more expensive models operating on different principles, this does not justify any generalization — there are good and bad pickups of every type.

Section 2. The crystal pickup

In 1880 Pierre Curie and his brother Jacques discovered an effect which they described as piezo-electricity (or "pressure" electricity); it is exhibited by a small group of materials all of which are crystalline. Such materials acquire an electric charge when subjected to mechanical pressure and, conversely, undergo mechanical deformation when a voltage is applied to them. Although we are concerned primarily with the fact that any force applied to a piece of piezo-electric material is converted into an electric charge, let us first consider what happens in the opposite case, that is, whan a voltage is applied to the crystal. Fig. 17a shows that when a positive voltage (+) is applied to the top face, and a negative one (—) to the bottom face of a crystal plate conductive at both sides, this plate expands longitudinally. If the voltages are reversed the plate contracts (fig. 17b). Again, a voltage applied to two such plates one on top of the other in the manner indicated in fig. 17c makes the top one expand and the bottom one contract, so that if the two are cemented together they must bend. Of course, the expansion or contraction of the plates does not happen at random, but is governed entirely by the manner in which they are cut from the original crystal.

Conversely, such a combination of two crystal plates produces voltages as indicated in fig. 17c if bent one way, and voltages

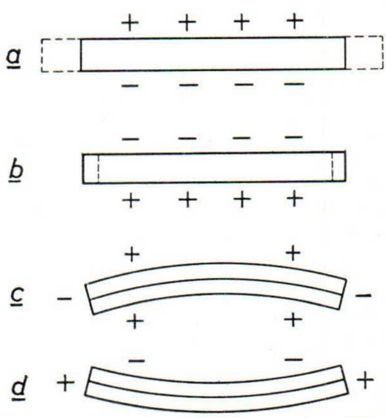


Fig. 17a, b, c, d
Principle of the crystal pickup

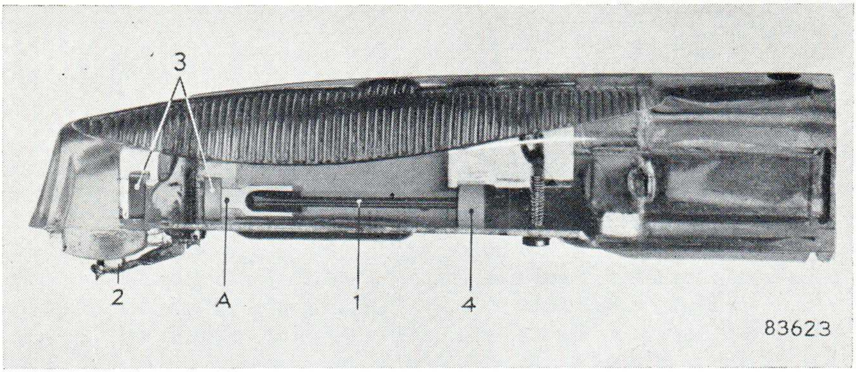


Fig. 18. Construction of crystal pickup type AG 3010

of the opposite sign if bent the other way (fig. 17d). Here, then, is the principle on which crystal pickups operate.

A combination of two crystal plates both sides of which are conductive is provided with a needle at one end and is clamped at the other end. As the needle tracks the groove of a record, it is deflected to and fro and therefore bends the crystal plates from side to side, thus charging them first in one direction and then in the other; the charge is extracted from the conductive layers on the plates as an A.C. voltage. So much for the principle. Practical pickups are slightly more complex, as will now be explained. In practice the crystal plates may be twisted instead of bent without affecting the principle. However, it is not practicable to secure the needle direct to the combination of plates, or bimorph as it is called, since this would seriously impair the quality of reproduction.

Fig. 18 shows the construction of a practical pickup. The bimorph (1) is clamped at the right-hand end (4) and the needle (2) is inserted in one end of what is known as the armature (A), seen on the left in the picture, operating in two ring bearings (3) which enable it to follow the movements of the needle; the other end of the armature is a fork gripping the left-hand end of the bimorph and transmitting the movements of the armature to it, thus twisting the bimorph and producing a voltage between its conductive layers.

This, also, is a somewhat simplified description. All the individual parts of such a pickup have certain resonant frequencies and thus produce natural notes which come out very strongly in the sound reproduced. Since such resonances cannot be tolerated in good quality reproduction they are damped, e.g. by inserting a thin disc of rubber between the bimorph and the armature fork. For the same reason, the bimorph holder (4) and the bearing rings (3) are made of carefully selected grades of rubber and the bimorph itself is embedded in a special jelly to damp the unwanted resonances or suppress them altogether. This jelly also serves another purpose, Not all crystalline substances have piezo-electric properties — the most widely used among piezo-electric crystals is Rochelle salt. Plates for crystal pickups are cut from giant crystals of this salt, cultivated by placing a seed crystal, not very much larger than the head of a pin, in a tank containing a saturated solution of Rochelle salt at a temperature of roughly 50° C. and reducing this very gradually, that is, in 4—6 weeks, to room temperature; during this time the crystal grows until it finally weighs several pounds. The fully grown crystals are cut into slabs, the slabs into strips and the strips into plates as used in pickups. These plates are then coated with metal or graphite to make them conductive, cemented together in pairs, and provided with leads. Now, in addition to many good properties, Rochelle salt has one or two bad ones. For example, it dissolves readily in water or a very humid atmosphere, and is soon reduced to powder by desiccation in dry atmosphere. Also, at temperatures above 50—55° this salt exudes enough water of crystallization to dissolve it. Accordingly, crystals of Rochelle salt are most at home in a climate of moderate temperature and humidity; under such conditions they last a very long time. On the other hand, the heat generated in a gramophone, or any undue humidity owing to a sudden drop in temperature in tropical, or even in sub-tropical regions may affect a Rochelle crystal seriously. To minimise the effect of moisture. bimorphs are coated with wax or lacquer, but, although this protects them to some extent, it is by no means fully effective, since water vapour will ultimately penetrate the thin protective layer.

Owing to the presence of the needle it is not practicable to enclose the bimorph completely in a soldered tin cartridge; hence the only solution is to embed it in grease or jelly, thus sealing off all openings without impeding the movements of the needle.

Also, the jelly enhances the quality of reproduction, as we have already seen, and protects the bimorph from damage. Now, the earlier crystal pickups often broke down at temperatures well below 50° C; moreover, they were rather rudimentary in design, so that the reproduction was by no means perfect. The principal advantage of the crystal pickup is that its output voltage is relatively high, i.e. of the order of 1 volt, and can therefore be used even with the simplest of radio receivers without pre-amplification. Also, such pickups are light (the one shown in fig. 18 weighs only 6 grams, or less than ¼ oz.), which is of course a great advantage from the point of view of record wear. In fact its lightness combined with the relatively low price explains the current popularity, despite initial drawbacks, of the crystal pickup. We shall see from the description of the electrical properties that this popularity is also merited from the point of view of the quality of reproduction.

Now, apart from Rochelle salt, two other materials are used to make plates for piezo-electric pickups; one of them, ammonium dihydrogen phosphate (A.D.P.), better known under the trade name P.N., can be used at temperatures up to 100° C. It does not dry out but is equivalent to Rochelle salt as regards sensitivity to moisture. On the other hand, the low internal capacitance of such crystals (some hundreds of pico-farads as against 2000 pF for Rochelle) necessitates a very high resistance (R) at the input of the amplifier to prevent undue

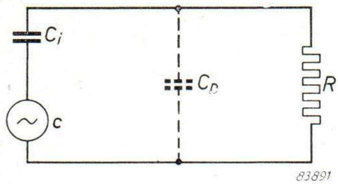


Fig. 19. Equivalent circuit of crystal pickup

attenuation of the bass notes (see fig. 19). Owing to this resistance the pickup flex is unduly receptive to hum: moreover, line-capacitance in a long flex cuts down treble response.

In view of these disadvantages A.D.P. pickups are employed only under conditions consistent with the use of a very short amplifier lead. Also, this material has a high input resistance and a relatively low output voltage; hence it is not used as widely as Rochelle salt.

The other material, barium titanate in various forms, differs from those already described in that it is multicrystalline and, even more important, ceramic; hence its life is not noticeably affected by variations in temperature or humidity. In fact this material is theoretically everlasting.

On the other hand, barium titanate has one or two drawbacks. Like all ceramic materials it is very brittle, which necessitates relatively thick plates which are accordingly rather rigid; this affects both the treble and the bass response of the pickup, and so limits the output voltage that it is usually too low (of the order of 0.1 volt) to load an ordinary radio receiver. Moreover, such plates crack readily despite their thickness.

Another peculiarity of barium titanate plates is that they require polarisation to make them piezo-electric; also, overheating deprives them of this effect, although it can be restored by re-polarisation. All that is necessary to produce polarisation is to apply a high D.C. voltage between the bimorph connections.

Barium titanate plates are even more difficult to make than good crystals of Rochelle salt; in fact, the production of such plates has been called more of an art than a science. It is this, together with the other disadvantages referred to, which still restricts the use of ceramic pickups.

Section 3. The magnetic pickup

Until 1940 this was the most widely used type of pickup, but its popularity declined owing to two inconvenient features, namely that it is heavy and has a relatively low output voltage. Recent improvements in design have reduced the weight and enhanced the quality of reproduction of the magnetic pickup considerably, however, thus restoring its popularity.

Magnetic pickups operate on a principle altogether different from that considered so far. This principle will now be explained. Given a coil in a magnetic field, any variation in the intensity of the field will generate a voltage in the coil, the direction of this voltage depending upon the manner in which the field varies. The simplest example of a pickup operating on this principle is a coil into which a bar magnet is inserted; the magnet produces a positive voltage at one end of the coil on being inserted, and a negative voltage at the same point on being withdrawn. As long as the magnet is stationary, it produces no voltage in the coil.

The voltages generated by the moving magnet are proportional to the speed and magnitude of the flux variation in the coil. The direction of the flux also affects matters; it should follow the axis of the coil, since flux at right angles to the coil does not generate a voltage in it.

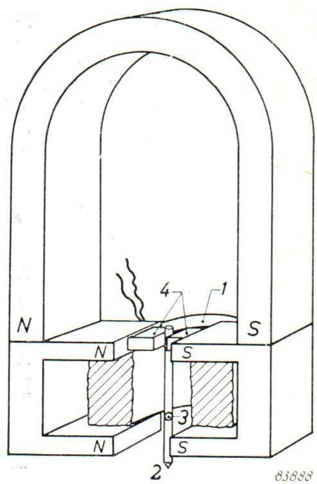


Fig. 20
Magnetic pickup

Fig. 20 is a diagram of a simple magnetic pickup. Note the coil (1; cross-sectional view) between two pole pieces attached to the horseshoe magnet; the left-hand pole shoes are magnetic North poles and the right-hand ones magnetic South poles. The armature, a rod of soft iron with a gramophone needle (2) at the lower end, is seen at the centre of the coil; it pivots in a bearing (3). As long as the armature is exactly vertical, that is, equidistant from the North and South poles, none of the magnetic flux passes through it. When the needle moves to the right, however, the bottom of the armature approaches the lower South pole and the top of it approaches the upper North pole, thus enabling some of the flux to pass through the armature, that is, along the axis of the coil. As soon as the armature moves from the central position, then, the magnetic flux through the coil changes, thus generating a voltage in the coil; the faster the movement of the armature, the higher the voltage generated. On moving to the left, the needle produces a similar effect, but the magnetic flux then passes through the coil in the opposite direction. As the needle is deflected to and fro, then, an alternating magnetic field is produced in the coil, which in turn generates an A.C. voltage in it. It will be evident that the different forces acting upon the needle would tilt the armature over to one side; they are prevented from doing so by special dampers (4). These dampers, usually made of rubber, hold the armature in a central position without preventing it from following the variations in the record groove. Although the quality of reproduction of magnetic pickups can be very good, they are rather heavy owing to the weight of iron and steel in them. The rubber damping pads, which are much more a structural part of the pickup than in the crystal type, tend to harden or perish in time, thus impairing the quality of reproduction and possibly enabling the armature to become polar, that is, to some extent permanently magnetized; armatures thus affected tend to be dragged sideways, which further impairs the quality of reproduction.

The output voltages of magnetic pickups are usually very much lower than those of the crystal variety, being of the order of 0.1 volt in standard, and still lower in really good quality, pickups. This arises from the fact that although relatively high pickup sensitivity can be ensured by making the air gap between the armature and the pole shoes small, it also affects the variations in the field through the armature in such a way that they are no longer proportional to the movements of the armature; this results in distortion. On the other hand, a large air gap, as employed for the above reason to ensure good quality reproduction, affects the size of the output voltage.

Modern pickups operating on the magnetic principle are known as variable reluctance pickups. Although there is no essential difference between a variable reluctance pickup, as shown in fig. 21, and earlier magnetic pickups, the former has one or two advantages. Two coils are employed instead of only one, and the armature is centred between two pole shoes secured to the coils (1); note also the magnet (4). As the armature moves to and fro, the flux through the two coils alternately increases and decreases. This arrangement enables the armature to be so pivoted that it does not require damping pads to keep it in a central position; also, it reduces the effect of any permanent magnetization of the armature.

On the other hand, the output voltage of this modern magnetic pickup is still too low (roughly 8 millivolts) to enable extra amplification to be dispensed with; it weighs only some 30 grams, however, which is a notable improvement on the earlier models.

Section 4. Dynamic pickups and microphones

The principle of the dynamic pickup is very much the same as that of the magnetic type; in essence, both are based on the same physical effects. In magnetic pickups a voltage is generated by varying the magnetic flux in a coil; in the dynamic model the direction of a coil relative to a magnetic field, or that of the field relative to the coil, is varied. In both cases the movement generates a voltage in the coil, the size of the voltage depending upon the speed of the movement.

Fig. 22 is a diagram of a simple electrodynamic pickup. Here we have a very small, light-weight coil (1) between the pole shoes of a horseshoe magnet. This coil carries the needle (2) and pivots in bearings (3); when pivoting it moves in relation to the magnetic field between the two pole shoes, thus generating a voltage, As the needle is deflected to and fro by the groove in the gramophone record it imparts a similar movement to the coil; it therefore generates an A.C. voltage, which is applied to the amplifier.

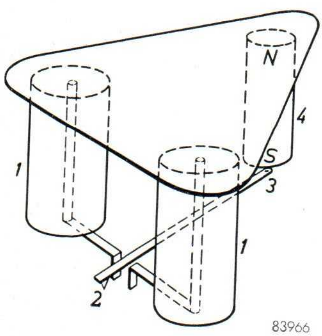


Fig. 21
Variable reluctance pickup

Like the magnetic pickup, this construction must include some form of support for the coil to prevent it from tipping over; however, the main problem is the weight of the coil.

To be light enough is must contain only a few turns of very thin copper wire; hence the output voltage of the pickup is very low,that is, a matter of one or two millivolts. On the other hand, the internal resistance of the pickup is also low, i.e. only one or two ohms as against some thousands of ohms in magnetic pickups;

this enables a matching transformer to be employed to step up the output voltage and so limit the amount of pre-amplification required. At the same time, this extra transformer adds to the cost, the more so since it necessitates effective screening to prevent hum. Electrodynamic pickups are not very widely used.

Now, by substituting a diaphragm for the needle we obtain a microphone; if properly designed, such microphones are of exceptional quality and are therefore employed in many broadcasting studios. Since the air-vibrations driving the diaphragm are of course even weaker than the vibrations imparted to the pickup needle by the record groove, light-weight moving parts are even more essential in the microphone than in the pickup; hence the moving coil is replaced, in some cases, by a single aluminium ribbon, which also serves as a diaphragm. Some of these ribbon microphones are very efficient. Of course, ribbons can also be employed instead of coils in electrodynamic pickups, but this severely limits the output voltage.

Let us now consider the magnetodynamic pickup. This became a practical possibility with the advent of a magnetic material known as Ferroxdure, thin rods of which can be magnetized laterally as well as longitudinally.

With magnet steels, such as Ticonal and Reco, only longitudinal magnetization is possible.

A laterally magnetized rod of Ferroxdure (a ceramic material) may be imagined as having its North and South poles indicated by coloured stripes down each side; similarly, magnet steel rods may be imagined as having their poles indicated, not by longitudinal stripes, but only by coloured end faces.

Fig. 23 shows the working principle of the magnetodynamic pickup. A magnetized rod of Ferroxdure (only 0.8 mm thick) is placed between the ends of a U-shaped bracket carrying two coils (1). The North pole of the rod faces outwards and the South pole inwards, both being exactly midway between the „legs" of the U. When the needle (2) is deflected by the record groove, the North pole, say as seen from the needle, moves to the right and the South pole to the left, producing a magnetic flux through the bracket. Accordingly, an alternating flux passes through the coils on the bracket, producing an A.C. voltage in them. Since the coils do not move, they can be wound very much larger than those of electrodynamic pickups; hence magnetodynamic pickups can be made very much more sensitive than electrodynamic models of the same overall weight, despite the fact that the small Ferroxdure magnet is not as powerful as the magnet in an electrodynamic pickup.

On an average, the output voltage of magnetodynamic pickups

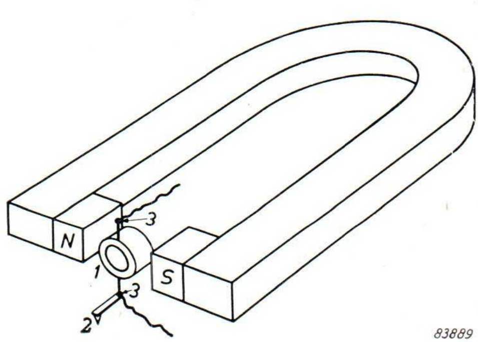


Fig. 22. Electrodynamic pickup

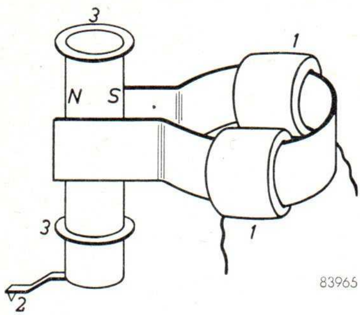


Fig. 23
Magnetodynamic pickup

is 20 millivolts; the relatively large number of turns in the coils enables the matching transformer to be dispensed with. Moreover, the Ferroxdure rod is lighter than the lightest of moving coils, which is of considerable importance from the point of view of treble response.

Again, owing to the cylindrical shape of the moving part of the magnetodynamic pickup, its moment of inertia is very low. On the other hand, the output voltage still requires pre-amplification to enable it to load a radio receiver.

Section 5.
Condenser pickups and microphones.

Pickups and microphones belonging to this group differ from those described so far in that they require an auxiliary voltage. However, the principle is quite simple.

When two metal plates, one rigid (1') and the other free to move (1), are placed close together, they form a capacitor whose capacitance is governed by the size of the plates and the space between them. When such a capacitor is operated as a pickup with the needle attached to the moving plate, the space between the plates varies with the deflections caused by the record groove, producing capacitance variations which are more or less proportional to these deflections. The next step is to convert the capacitance variations into alternating voltages; there are two methods of doing this. The first is based on the fact that the charge (or quantity of electricity) of a capacitor is the product of its capacitance and the applied voltage, or formulated, $Q = C \times V$. Given a constant charge, any variation in the capacitance must produce an inversely proportional variation in the voltage, which is suitable for applying to the amplifier. To satisfy the above conditions, the pickup or microphone is connected across a very high resistance to a D.C. supply of, say, 200 Volts. Owing to the high resistance, e.g. 100 megohms, the charge on the capacitor cannot change quickly enough to follow the vibrations of the movable electrode; hence the voltage on the capacitor fluctuates. The advantages of the condenser microphone outweigh the disadvantages that it requires an auxiliary voltage, and necessitates incorporating the pre-amplifier valve in the microphone head to avoid loss of sensitivity owing to line capacitance.

In fact, this microphone is eminently suitable for use in acoustic measurements, but as far as is known at the moment there are no pickups of this type on the general market. The other method of converting capacitance variations into alternating voltages is to employ the microphone or pickup element as the tuning capacitor of a miniature F.M. transmitter. The vibrations of the movable capacitor plate modulate the frequency of the transmitter to produce a signal which is applied to a simple F.M. receiver; this receiver converts the signal into A.C. voltages at audio

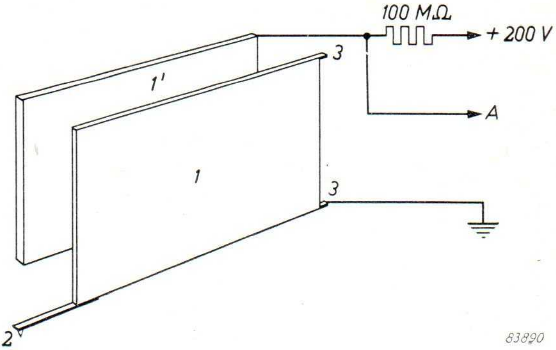


Fig. 24
Condenser pickup

frequency, which are then amplified and passed to the loudspeaker. The advantage of this principle as compared with the other is that the A.C. voltages employed are very much higher and therefore very much less likely to be subject to hum. Hence it is used for pickups as well as for microphones. Although the F.M. transmitter must be mounted close to the microphone it can be connected to the F.M. receiver and other equipment by a suitable cable, if necessary several yards long. To ensure adequate sensitivity, however, some microphones and pickups of this type are so dimensioned as to produce a certain amount of distortion. A similar effect has been referred to in connection with the magnetic pickup (see page 25) and must also be taken into consideration in the designing of dynamic pickups; however, this effect appears to be even more noticeable in condenser pickups. To enhance the sensitivity of the pickup, the space between the fixed and movable plates is so reduced that the capacitance variations are no longer proportional to the movement of the diaphragm, so that increases in the amplitude of vibration produce more than proportional increases in the capacitance variation. However, it is possible to impress a similar distortion but of opposite sign on the F.M. transmitter, so that the two distortions cancel each other out.

Condenser pickups are not very widely used, however, because they necessitate a good deal of extra equipment and circuitry, which is rather complicated at any rate for amateur use; at the same time condenser microphones are used extensively in recording studios.

CHAPTER IV-B

PICKUPS — CHARACTERISTICS

Section 6. Frequency characteristics

Before discussing the characteristics of pickups in full, we shall now consider the basic shape of the recording characteristic. As recording characteristics in general will be described in detail in chapter V, the present discussion will be confined to the pre-war 78 r.p.m. recording characteristic, which really dates back to the period of acoustic reproduction. Now, given equal sound intensity, the displacement of the cone in a loudspeaker or of the mica diaphragm in the sound-box of an oldfashioned acoustic gramophone is greatest for the low notes; it decreases according as the frequency increases. The same applies amongst other things to the various stringed instruments, whether played with a bow or plucked, and to percussion instruments. Given the means of measuring the displacement of the vibrating part in a sound source, it will be found that, the sound intensity being equal, this displacement is inversely proportional to the frequency. We refer to sound intensity in the physical sense, not to loudness, which may differ from the physical intensity of sound owing to the peculiarities of the human ear (see page 91). Given two notes equal in sound intensity, e.g. A : 440 c/s and A1 : 880 c/s, the amplitude (displacement) of the vibrations of the highest, is half that of the vibrations of the lowest, note. The higher A completes a full vibration (cycle) in half the time required by the other A. It follows from these two statements that the velocity of vibration of the air particles is the same for both notes.

The sound intensity being equal, then, the amplitude of vibration is inversely proportional to the frequency, and the velocity of vibration of the air or diaphragm is independent of the pitch. This is demonstrated in fig. 25. Here, the solid line represents the velocity, and the chain one the amplitude, of vibrations at different frequencies. In an ideal recording, the sound vibrations would be impressed in the record without being distorted. However, since no gramophone is entirely free from background noise and the records themselves, especially pre-war ones, produce noise, even the smallest amplitude should be large enough to ensure that the associated notes are not drowned by noise or other interference.

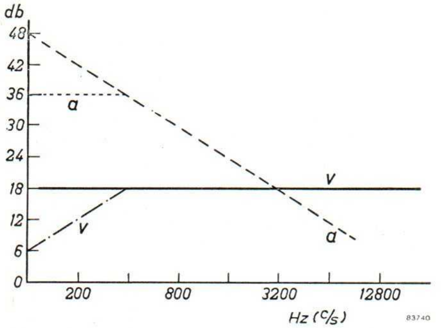


Fig. 25. The fundamental recording characteristic

Now, it is seen from the diagram

that the smallest amplitude corresponds to the highest note reproduced. If the highest note is, say, 12,800 c/s and the associed amplitude 2 microns, the amplitude at 100 c/s will be very much larger, i.e. 256 microns. Since the space between the grooves of a 78 r.p.m. record is only 100 microns, the lowest notes must be attenuated considerably to avoid overcutting the grooves. In practice, however, the theoretical limit of 50 microns may be raised to 64, so that amplitude attenuation by a factor of 4, that is, 12 db, is required at 100 c/s. The attenuation required to limit the amplitude to 64 microns at 200 c/s is 6 db, and at 400 c/s attenuation may be dispensed with; accordingly, the dotted line in the diagram represents the permissible variation of the amplitude characteristic below 400 c/s. Of course, this variation also affects the velocity characteristic, the low-frequency end of which is here indicated by a chain-dotted line.

With acoustic recordings the necessary correction was obtained quite fortuitously owing to the fact that the horns employed were not very efficient at low frequencies; with electrical recordings it is obtained by means of correcting filters.

As we have seen in sections 3 and 4 of this chapter, the output voltages of magnetic and dynamic pickups are proportional to the velocity of the moving part, that is, in essence, the needle velocity; hence line v in fig. 25 is also the response characteristic of magnetic and dynamic pickups.

The output voltage of piezo-electric and condenser pickups is proportional to the amplitude of the needle movement; hence line a in the diagram is the fundamental response characteristic of these pickups. In the one case, then, we have a deficiency of bass notes, whereas in the other all notes above 400 c/s are attenuated. Both can be corrected. In principle, the crystal pickup is capable of producing a relatively high voltage, say, 15 volts, which is ample; hence the response characteristic can be flattened to some extent by making the coupling between the needle and the crystal plates looser at low than at high frequencies. That very good results are obtained by this method will be seen from fig. 26, showing the response characteristic of pickup AG 3012. This characteristic, plotted with the aid of H.M.V. record DB 4037 (up to 8000 c/s), shows that the difference in the sensitivity of the pickup as between 400 and 8000 c/s is only 9 db, instead of 27 db.

Apart from the fact that the slightly exaggerated bass response of this pickup is appreciated by many listeners, it ensures an almost perfectly straight response characteristic in conjunction with a simple electric filter (chain line in fig. 26). On the other hand, the output voltage of magnetic and dynamic pickups is too low to enable effective use of such correction to be made.

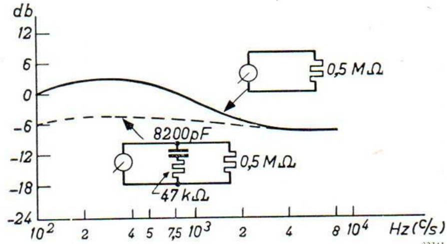


Fig. 26. Response characteristic of pickup type AG 3012

Although the bass attenuation can be partly compensated by making the

armature resonant at, say, 150 c/s (in effect, tuning it to this frequency), this method, which seriously impairs the quality of reproduction, is now for all practical purposes obsolete; the modern method is to bring the bass notes to the required level by means of a correcting filter in the amplifier (see page 90).
The response characteristic of a pickup is governed not only by the properties of the pickup itself, but also by the characteristics of the particular record used to determine it; hence response characteristics are useless unless they specify the particular record or at any rate the type of record. Although the plotting of response characteristics is one of the simplest of laboratory measurements, there is far too much evidence that published characteristics are not always accurate and that they are far too readily drawn freehand. Some magnetic and dynamic pickups are provided with response characteristics for a hypothetical record having constant needle velocity over the entire range; there is no objection to this, provided that a suitable test record is employed and corrections are introduced only in respect of its deficiencies at the lowest frequencies.
Another source of error is that microgroove test records and standard-groove shellac records cut with the same recording characteristic may have different response characteristics. This is particulary important from the point of view of pickups with two needles, one for microgroove and the other for standard-groove records, which in many cases are tested only once. Owing to the greater elasticity of the vinyl of which microgroove records are made, the highest notes will be weaker when a microgroove record is played than in the case of shellac records; also, pickup resonance will be slightly lower with the microgroove record.
Pickups with straight characteristics are supersensitive at a certain frequency, and fall far short of normal sensitivity at frequencies beyond it; in most cases this is due to armature resonance. This small peak in the response characteristic, and the possible sharp drop in response beyond it, enhance the adjacent frequencies very much more than the diagram itself suggests; in fact, the resonance intensifies the tone appreciably. Hence we invariably try to take the peak as far as possible above the highest audible frequency. It is clear, then, that the response of a pickup to microgroove records cannot be deduced accurately from tests on shellac records. Fig. 27 shows to what extent the results obtained by playing a shellac record (1) and a vinyl record (2) with the same pickup may differ. Now, pickup response characteristics fall into three categories. Firstly, characteristics which, with or without electrical correction, are virtually straight until they reach the highest audible frequency, or even beyond it. Provided that the other pickup properties are equally good, we then have what is known as High Fidelity (1 in fig. 28).

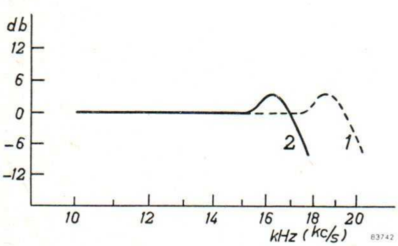


Fig. 27. Effect of record material on response characteristic

Secondly, characteristics which are

straight up to a certain frequency, but fall off gradually beyond it. Although electrical correction of such characteristics is possible, it is generally impracticable owing to the loss of amplification involved. Such pickups are rich in tone and reproduce even the highest frequencies in some degree, but lose much of the characteristic timbre of different musical instruments.

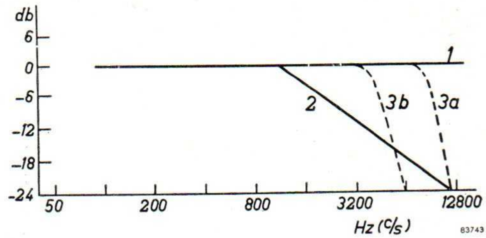


Fig. 28. Different response characteristics

Lastly, there are characteristics which fall off steeply beyond a certain frequency (3a and 3b in fig. 28). The effect is that the tone becomes disconcertingly shrill, giving the impression of more treble than is actually present.

Although this effect cannot be fully explained, it is analogous to adding a touch of blue to white to make it even whiter, as is done by painters, and by laundries in the washing of sheets and so on. Similarly, an audio frequency spectrum limited in a certain manner gives the impression of being more complete than it really is; being „whiter than white", however, the sound has a tendency toward shrillness.

Let us consider case 3a in fig. 28. Here, the cut-off frequency is on the high side and the tone rather strident; nevertheless, to a listener concentrating on treble response this may well seem better than the effect produced by a pickup with characteristic 1. In fact, such reproduction is fatiguing to the ear and the difference, say, between a violin note as originally played and as reproduced by a rather shrill pickup is noticeable enough when the two are compared.

The purpose of characteristic 3b is altogethed different. It cuts off just above what is normally the highest fundamental tone in music. Accordingly, a transducer with this characteristic brings out every **note,** although is does not by any means reproduce every **tone** to full advantage. Hence it suppresses much of the distortion, confined mainly to the higher frequencies in old 78 r.p.m., or less succesful microgroove records.

Such pickups also suppress much of the noise inherent in shellac records; moreover they are rich in tone and even reproduce inferior records adequately. A suitable choice of the descending portion of the characteristic enables the tone to be brightened slightly; carried to far, however such brightening makes the reproduction unduly shrill.

Now, the shape of the low-frequency end of the response characteristic is also important in quality reproduction. Given a straight portion up to 25 c/s, with or without electrical correction, even the lowest notes recorded will be reproduced quite faithtfully. However, some pickups fall off steeply below 100 or 150 c/s; hence they partly suppress the fundamental tones of many musical instruments. Although this loss is masked to some extent by the overtones reproduced, the particular tone is not reproduced with all its original quality.

At the same time, certain pickups are designed to be deficient in this respect in order to suppress possible motor-rumble.

Section 7. On forces and masses

Pickups should exert enough pressure on the needle to ensure that it does not leave the groove during loud passages, or even show a tendency to do so, since even this is enough to distort the reproduction and wear the record unduly. On the other hand, the needle pressure should not be too heavy.

For every pickup there is an optimum needle pressure, usually betewen 10 and 12 grams at the present stage of technical development. The required needle pressure depends upon several factors, the more important of which are the compliance and the moving mass. Both refer to the motility of the needle, the compliance at low, and the moving mass at high frequencies. When the needle moves away from the central position, a force acts upon it to overcome the stiffness of the needle suspension. Now, this stiffness can be measured by applying a known lateral thrust to the needle and measuring the resulting displacement by means of a measuring microscope; stiffness thus determined is expressed in grams per centimetre, say, 500 g/cm = 50 milligrams per micron. However, the practical value of such tests is limited owing to the fact that the real needle movement is a rapid oscillation involving friction and resonance-effects not reproduced by the above method.

A more accurate, but at the same time very much more difficult method is to measure the displacements and forces whilst moving the needle rapidly to and fro, say, 50 times per second. In this way we measure the compliance, that is, the needle displacement, in centrimetres, produced by a force of 1 dyne acting upon the tip of the needle (1 dyne = 1 milligram, incorrect from a scientific point of view but permissible in the present discussion). Compliance can be measured in various ways; but the results do not always agree. Since this can cause a great deal of confusion, firms employing tests tending to produce results on the low side do not publish them, whereas others whose methods produce quite impressive results do so very readily. To be on the safe side, then, the compliance associated with a needle pressure of 0.4 oz. (10 grams) should be roughly 10^{-6} cm/dyne (1 millionth cm/dyne), that is, roughly 0.01 micron per milligram. Given a maximum amplitude of 64 microns, the force acting upon the needle is 6.4 grams. Since the angle of the groove wall is 45°, it follows that the needle pressure should be at least 6.4 grams. Because several factors are involved, however, a certain safety margin is necessary; hence it is usual to employ needle pressures in the region of 10 grams. Greater compliance would enable the needle pressure to be reduced, but matters are complicated by what happens at the higher frequencies.

Frequencies above 2000 c/s are not affected by the stiffness of the suspension, but only by the moving mass. It is very much smaller than the total mass of the pickup and comprises the greater part of the needle with part of the armature. This mass oscillates very rapidly, i.e. up to 10.000 times per second or more; hence it is periodically accelerated and decelerated. Mathematically this may be explained as follows:

Assume that the maximum velocity of the needle point is 25 cm/sec; the instantaneous velocity of this point will then be $v_m = 25 \sin\omega t$. Accordingly, the acceleration is $a = \frac{dv}{dt} = 25\ \omega \cos\omega t$; peak acceleration is therefore $2 \times 25\ \pi f$, where f is the frequency. At 10.000 c/s, then, the acceleration is roughly 1,600.000 cm/sec^2, or 16 km/sec^2 (10 $mile/sec^2$). With 3 minutes of such acceleration, the needle would travel the distance from the earth to the moon and it is only because the period of acceleration is in fact very much shorter that the needle is not literally shaken to pieces. Even so, the instantaneous forces of acceleration are very powerful indeed and, as will be seen from the well-known formula $f = m \times a$ (force = mass times acceleration), the force to impart the above-mentioned acceleration to a mass of 1 milligram is 1600 dynes, or 1.6 gram. With a moving mass of 4 milligrams — by no means excessive — the forces are of the same magnitude as those associated with a compliance of 10^{-6} cm/dyne at low frequencies. Hence the small size of the needles in modern pickups, since every milligram of extra weight makes it more difficult for the needle to follow the groove properly on fortissimo treble notes, and too large a moving mass, although it does not necessarily cause the needle to leave the groove, nevertheless results in distortion and undue record wear.

Now, the moving mass is roughly 6 milligrams in the pickup shown in fig. 18, 8 milligrams in that shown in fig. 21, and only 3 milligrams as measured in the magnetodynamic pickup shown in fig. 23, which is about the lowest value encountered at present. If anything, this comparison shows that reducing the needle pressure of certain pickups would almost inevitably lead to difficulties with treble response, especially when high-fidelity microgroove records are played; also that this pressure should not be based upon the compliance only.

The method employed to ensure a given needle pressure is also important. Provided that the pickup itself and the pickup arm are light enough, a needle pressure of 10 grams can be obtained without difficulty, but if the two are heavy it is necessary to balance them. That it would be a mistake to underestimate the importance of the means of procuring a given needle pressure will now be shown by means of a simple example. Suppose that a heavy iron rod 6 feet long and a thin wooden lath of the same length are both suspended exactly at the centre without friction and are therefore perfectly balanced. If a needle be attached to the end of each rod and a 10 gram weight placed on top of it, the needle pressure will be exactly 10 grams in both cases (fig. 29). If we try to lift first the one and then the other by pushing the needle up with one finger, however, we find that whereas the light wooden lath lifts quite easily, the heavy rod offers so much resistance that the needle pricks the finger. Conversely we find that when once lifted, say, 6 inches, the iron rod takes very much longer to descend to its original level than the wooden lath. Although this is taking matters to extremes, the results nevertheless hold good also for balanced and non-balanced pickup arms. If the record lifts, say, owing to shock or vibration, the needle must rise with it. Heavy, balanced arms resist this movement and therefore damage the record, whereas light, non-balanced pickup arms, although offering a certain amount of resistance, impose a very much smaller load on the record. If the vibration changes direction, light pickup arms then follow the record down at once, but with heavy ones there is a certain amount of delay; hence the needle hovers over the record for a moment before descending, quite possibly at the wrong point. For the same reason, heavy pickups tend to dance about on any records which happen to be slightly warped.

It is also possible to demonstrate the above in the form of a simple calculation assuming that the mass of the pickup arm is concentrated at the end of it.
Supposing that some force or other causes the record to rise with a certain velocity v, it will impart the same velocity to the needle, and therefore to the mass (m) of the pickup arm. The associated kinetic energy, $\frac{1}{2}$ mv^2, is derived from the record; hence a large mass giving rise to a large amount of kinetic energy, may damage the record. This may be explained as follows. Let h be the vertical distance between the normal level of the record and the peak of its shock-induced upward movement, and D the needle pressure. The potential energy of the pickup arm at the moment when the record drops backs to its normal level is therefore D $\times$ h. Now, to ensure that the needle does not lag behind the record, this product must exceed $\frac{1}{2}$ mv^2;

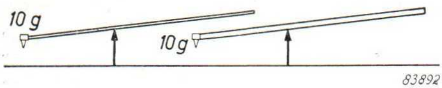


Fig. 29. Light and heavy pickup arms

$$D.h > {}^1/_2\ mv^2 \quad \text{or} \quad \frac{D}{m} > {}^1/_2\ \frac{v^2}{h}.$$

Accordingly, stability is conditional upon the needle pressure (D) being as far as possible heavy in relation to the mass (m); in other words, to avoid groove-skipping and the damaging of records, the needle pressure should be as far as possible heavy in relation to the weight of the pickup arm. Given equal needle pressure, then, a non-balanced pickup arm is better than a very heavy, balanced model. Nevertheless, since balancing cannot always be dispensed with, it is worth mentioning that springs should invariably be employed for this purpose, not counterweights, which would increase the mass. It will be evident that the above applies not only to vertical, but also to horizontal vibrations. All the effects described in this section can be demonstrated quite easily by means of a slightly warped record or by placing a 45 r.p.m. record eccentrically upon the turnable.

Section 8. Distortion - Two varieties

Any difference in the intensity ratios of tones at different frequencies as between the original playing and the reproduction is called linear distortion, and sounds other than hum, rumble and noise, not present in the original sound but arising from faults in the technical equipment are referred to as non-linear distortion. The latter, ordinarily described as simply distortion, is due to a disparity between an increase in voltage somewhere in the circuit and the associated increase in current, the one increasing, say, by a factor of 2 and the other by another factor slightly greater or smaller. Owing to such distortion purely sinusoidal vibrations (pure tones, see fig. 8) assume a different shape, say, as shown in fig. 30a.

On closer examination such distorted tones are found to comprise several individually pure tones, namely the original vibration and a number of weaker vibrations at twice or three times the frequency, or even at relatively very much higher frequencies in tones which are really seriously distorted. (See fig. 30b).
Such distortion changes the tone, usually for the worse, and if the sum of the amplitudes of the higher harmonics (really the total r.m.s.) reaches a certain intensity relative to the fundamental tone, the distortion is quite audible; this effect is produced by distortion exceeding 5—10%.
At the same time, the reproduction of music invariably involves not just one pure tone, but several vibrations occurring simultaneously. Under such conditions another effect emerges, different from, albeit related to, the one considered so far. In intermodulation distortion, as it is called, the different tones affect one another and non-linear distortion introduces fresh vibrations all the more distracting because they are not harmonically related to the original tones. The frequencies of the new vibrations are the sums and differences of the original vibrations.
Given two different tones, say 100 c/s and 4000 c/s reproduced simultaneously, any intermodulation distortion occurring during the reproduction will introduce extra vibrations at 3900 and 4100 c/s. Since almost all musical notes comprise a fundamental and several overtones and orchestral music, at any rate, is invariably produced by many instruments played simultaneously, intermodulation distortion going beyond a certain limit at once affects the quality of reproduction very seriously.
Such distortion may be caused in many different ways and may originate not only in the pickup, but also in the particular record; hence it is not easy to measure.
Special intermodulation test records carrying two notes, usually 100 and 4000 c/s, which are recorded simultaneously with an intensity ratio of 4 : 1, are now on the market. These notes are not uniform in intensity over the whole of the record; peak intensity is at the periphery; the needle-point velocity is, say, 20 cm/sec and from the periphery inwards the intensity of the test notes gradually falls off to zero. This test will be described more fully in chapter VIII; the intermodulation curve of a magnetodynamic pickup is seen in fig. 31. It is worth mentioning that in this case the intermodulation of record and pickup was measured.

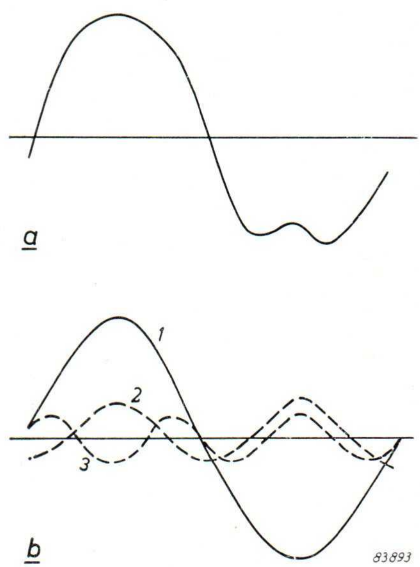


Fig. 30 a) Distorted vibration
b) The fundamental with harmonics

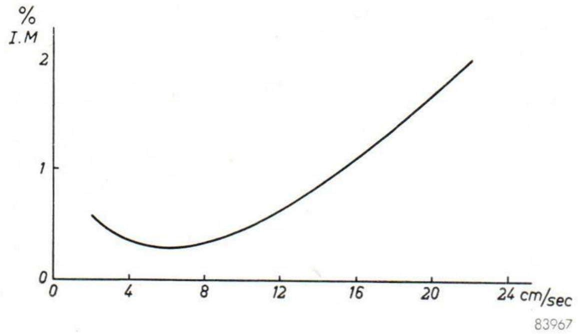


Fig. 31. Intermodulation characteristic of magneto-dynamic pickup AG 3021

The variation is not always perfectly gradual; intermodulation distortion is sometimes quite strong with small needle-amplitudes, but decreases when the strength of the recording increases. This suggests that the armature movement is affected by friction, which is most in evidence when the movements are very small.

Apart from intermodulation distortion, we have what is known as beat-note distortion; here the difference between two continuously variable test-frequencies is precisely 1000 c/s.

Good pickups are not seriously affected by such distortion.

Section 9. Needle talk

Electric pickups also convert part of the needle motion direct into sound. This is not distracting in itself provided that the gramophone is enclosed in a cabinet, but may be a drawback in the case of open models especially if the loudspeaker is rather far from the pickup.

Such needle talk, as it is called, may be caused in several ways, e.g. by sympathetic vibration of the pickup shield or arm; under certain conditions such vibrations may react upon the needle in such a way as to accelerate record wear. Again, if the compliance of the needle is too low or its effective moving mass too great at certain frequenties, this may likewise cause needle talk and record wear. Needle talk cannot be avoided altogether, since the vibrating needle and the parts attached to it excite the surrounding air so that this also vibrates — there is no means of preventing this, of course, and in any case it does not endanger the record. Another cause of needle talk is that the needle vibrates not only horizontally, but also vertically (see page 41). The frequency of the vertical vibrations is twice that of the horizontal vibrations, and since precisely at high frequencies vibrations of small amplitude produce sound of considerable intensity, it is the vertical vibrations which are most likely to cause objectionable needle talk. It is highly important that the diamond or sapphire needle be as light as possible; the Philips pickups mentioned in this book operate with needles weighing only 0.4 milligram.

CHAPTER V
THE NEEDLE AND RECORD

Section 1. Tracking distortion

In recording, the cutter is guided by the lead screw in a straight line from the periphery, towards the centre of the disc. The reproducing needle does not move in a straight line, however, but describes an arc, since the pickup is attached to an arm rotating about a fixed point (fig. 32a). The result is that for most of the time the position of the pickup relative to the record grooves is different from that of the cutter during recording, and the direction of movement of the needle – at right angles to the axis of the pickup – does not coincide with the direction of the groove modulation, since this is invariably at right angles to the groove.

Fig. 32b shows what happens. The solid line in this diagram represents the actual groove, that is, the track of the cutter during recording. The tip of the reproducing needle also follows this track but its lateral velocity is not the same as that of the cutter, the result being that at a given moment the tip of the needle is at point 2, instead of at point 1, as it would be if the pickup were correctly positioned (in line with the chain-dotted line). At that particular moment, then, the needle tip is slightly too far to the left, and the instantaneous voltage is therefore correspondingly low. In this way the needle deviation at any given moment can be computed point by point, as indicated by the chain line in fig. 32b. Hence it is seen that whereas the original groove corresponds to a pure tone, the needle movement is distorted. This distortion produces even-harmonics, that is, vibrations at twice, four times, and six times the frequency. The second harmonic, whose frequency is twice that of the original tone, is the strongest.

There are several ways of avoiding such distortion. Firstly, by attaching the pickup to a parallel tracking arm, in other words, making it travel along a rod positioned, like the lead screw, along a radius of the record. This is not very practicable, however, partly because it is expensive and partly in that it makes the record player less simple to operate; moreover, it is almost impossible to combine a record changer with such a system. Another solution is to employ some means of rotating the pickup automatically and uniformly in relation to the pickup arm, in such a way as to ensure that the pickup is

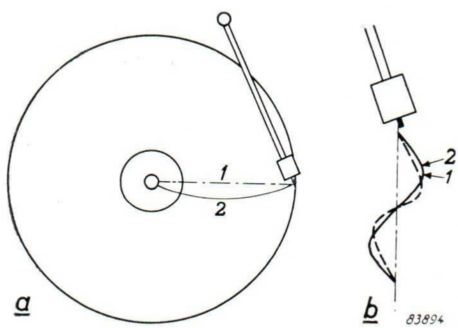


Fig. 32. The cause of tracking distortion

invariably in the proper position relative to the record groove. Although this system does not make the gramophone any more complicated to operate, however, it is not used in practice because it is so expensive. It would be possible to lengthen the pickup arm considerably and so bring arc 2 (fig. 32a) closer and closer to straight line 1, but the price objection again applies, the more so because a long pickup arm necessitates an extra large gramophone cabinet. Hence such arms are employed only in studio record players, where they are required in any case for another purpose, namely the playing of 16-inch acetate discs.

Calculations have shown that the tracking error can be reduced considerably by employing a curved pickup arm. Such an arm is shown in two different positions in fig. 33, from which it will be seen that the pickup is almost at right angles to a radius of the record (chain-dotted lines) in both positions. Accordingly, the curved shape of the pickup (or tone) arm is adopted not merely for the sake of its appearance, but for purely technical reasons; if properly designed it enables the tracking error and the associated distortion to be limited, even under the most unfavourable conditions, to less than 0.5%, which is small enough to be safely ignored.

The curved shape of the arm also has another desirable effect, as will now be explained. Owing to friction between the revolving record and the needle, the former exerts upon the tone arm a force in the direction of the groove at the point of contact of groove and needle. Since this force (arrow in fig. 33) is not in the same direction as the connecting line from the point of the needle to the tone arm bearing, it can be resolved into two components, the one in the same direction as the connecting line and the other at right angles to this line and in the direction of the centre of the record. Whereas the one force acts only upon the tone arm bearing, the other tends to pull the arm itself inwards; this centripetal force is very useful in several ways, as we shall now see.

When moving towards the centre of the record the pickup is impeded by friction in the tone arm bearing. The centripetal force helps to overcome this friction, thus enabling the pickup to follow the groove more easily and at the same time reducing the amount of pressure which the groove wall must exert upon the needle in order to move the pickup. This force also helps to guide the needle into the starting groove when the pickup is lowered on to the record. Again, the automatic switches in record players and the change mechanisms in record-changers are often actuated by the acceleration of the tone arm when the needle enters the run-out groove. A certain amount of force is required to operate the switch, and the frictional force acting upon the tone arm is very useful in this respect.

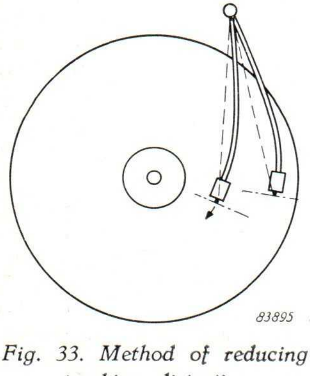


Fig. 33. Method of reducing tracking distortion

To minimize the tracking distortion, the tone arm is made long enough to bring the tip of the needle just in front of the turntable spindle when the arm is swung right over to the centre of the turntable; the centripetal force is then almost constant, and strong enough to be of practical value. The tone arms of Philips record players are so designed that the centripetal force is roughly 1 gram (0.04 oz.).

Where the automatic switch is operated by the return motion imparted to the tone arm by records with eccentric run-out grooves, a centrifugal force may be desirable; however, this is less advantageous from the point of view of tracking distortion and record wear.

Section 2. Tracing distortion and pinch effect

Fig. 34 shows a small portion of a record groove many times enlarged,, i.e. in plan view (a) and in two cross sections (b and c). It is seen from fig. 34b that modern needles have rounded tips which ride upon the sides, not the bottom, of the groove. The chain-dotted line in fig 34a indicates the centre-line of the groove. For completely undistorted reproduction the centre of the needle tip must follow the centre-line exactly. In reality, however, this centre follows another path which may be constructed geometrically by drawing circles at regular intervals all along the groove. Such a path is seen in fig. 34a (chain line).

A number of points in connection with this will now be explained. The width of the groove as measured in the vertical direction (d) is constant, since it is governed by the width of the cutter, which always moves at right angles to the groove as it would be without modulation. On the other hand, the shortest distance between the groove walls (a) equals d only at 90° and 270°, where, for a very short distance, the real groove is parallel to the groove as it would be without modulation. At all other points width a is smaller than width d, and the tip of the needle therefore rides relatively higher in the groove (fig. 34c); the effect of this is that the needle continually rises and falls during the playing of the record. Now, this vertical needle movement owing to the „pinch effect" of the groove is indicated in the diagram by a dotted line, from which it will be seen that the frequency of this movement is twice that of the horizontal excursion of the groove. It is clear that pinch effect becomes more and more noticeable according as the amplitude of the groove modulation increases and the wavelength decreases (that is, the higher the frequency). It is most noticeable on treble notes.

With pickups which do not respond to vertical movements of the needle, the pinch effect is troublesome only through being the principal cause of needle talk; as we have already seen, the mass of the needle should therefore be as small as possible. In pickups which do respond to such movements distortion occurs, especially of loud treble notes. Because the frequency of the pinch effect is twice that of the note as recorded, it is the even harmonics which are audible. Under certain conditions the amplitude of the pinch effect may be as much as 10% of the groove amplitude, so that, given a pickup as sensitive to vertical as to horizontal movements of the needle, the resultant distortion will also be 10% — in a properly designed pickup, however, such distortion can be limited to less than 1%. Since the highest notes

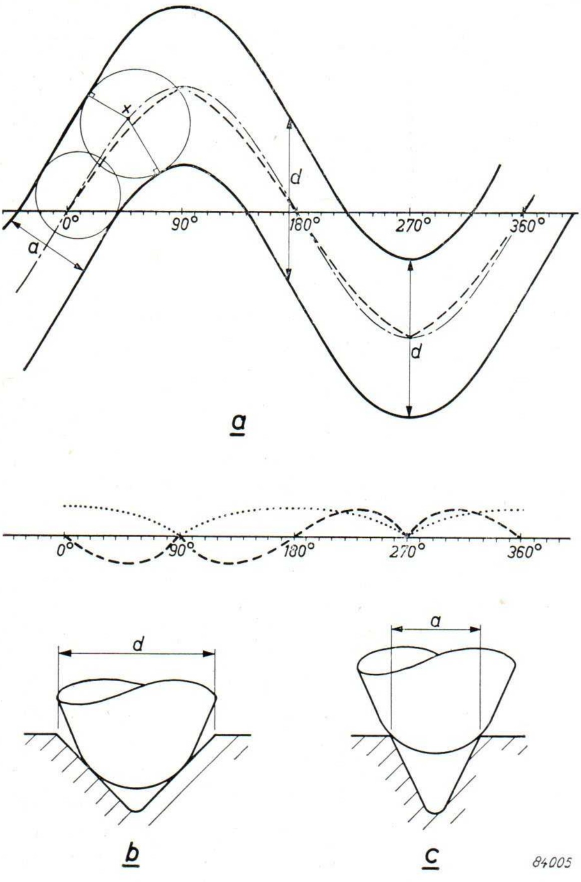


Fig. 34. The causes of pinch-effect and tracing distortion:
a) plan view of groove
b) cross-section of groove (straight part)
c) as b) (curved part)

are almost invariably overtones of lower fundamentals, and only a few of them are of any appreciable intensity, none of the distortion arising from pinch effect will be audible in practice provided that a good pickup is employed. In fig. 34a, the path of the centre of the needle tip is represented by circles drawn one after another in the groove and joined by lines from centre to centre. The diagram includes several of these circles and shows quite clearly that the needle centre is not on the centre-line of the groove, but below it. This is because the normal from point x to the upper wall of the groove is shorter than the normal to the lower wall. The difference between the centre-line and the path of the needle tip is again plotted separately on a larger scale; this is a graphical representation of tracing distortion. Analysis shows that such distortion consists mainly of odd harmonics, i.e. vibrations whose frequenties are 3, 5, 7 times the frequency of the note recorded. The distortion also contains the frequency of the note itself, but this vibration is opposed to the note and therefore attenuates it. It will be evident from fig. 34, in which the different ratios are exaggerated in order to be more easily understandable, that the tracing distortion increases according as the wavelength of the recorded vibration decreases and the amplitude of this vibration increases.

It is also clear that the distortion must decrease if a narrower groove and a thinner needle tip are employed.

The wavelength of a sound vibration is the velocity of propagation divided by the frequency. When such a vibration is "frozen" in a gramophone record, the velocity of propagation is superseded by the velocity of the record, that is: the rate at which the groove passes the needle, in feet or inches per second. This rate is not constant. The velocity of a 12-inch, $33^1/_3$ r.p.m. record is 20 inches/sec at the first groove and roughly 9 inches/sec at the last groove; the corresponding velocities for a 78 r.p.m. record are 48 and 16 inches/sec, respectively. This means that the wavelength of a given note on a gramophone record is not constant — the wavelength of the note A would be 1.2, 0.5, 2.7 and 1.1 millimetres in the above instances, and would assume other values at different points on the record or at different speeds (see fig. 35).

When all this is plotted to scale or when the distortion is computed, it is found that serious distortion occurs only at wavelengths very much shorter than 0.5 mm (0.02 inch), that is, at frequencies well beyond 400 c/s; since $33^1/_3$ r.p.m. records contain very much shorter wavelengths than 78 r.p.m. records, however, they should have a narrower groove, and be played with a thinner needle, in order to limit the distortion to the same level. Accordingly the tip-radius of microgroove needles is is usually only 25 microns as compared with 75 microns in the case of standard-groove records. At the same time, standard-groove records with what is known as a V-groove can also be played with a microgroove needle (noise and record wear

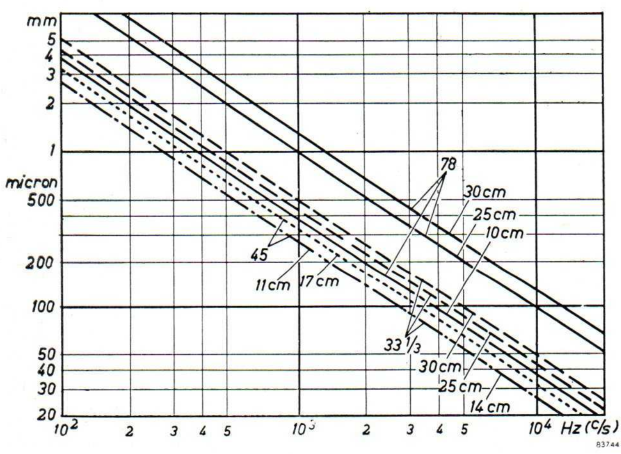


Fig. 35
Wavelength plotted against frequency for $33^1/_3$, 45 and 78 r.p.m. records

apart), and reproduce fortissimo passages with less distortion than when played with a standard needle.

The above also provides an explanation of the well-known fact that the quality of reproduction is better at the beginning than towards the end of a record, where fortissimo passages tend to sound distorted and the reproduction of treble notes is slightly attenuated. This is a point to be borne in mind by the composers of music for gramophone records, but in fact they, as also the arrangers of light music for recording purposes, are too prone to overlook it. The frequency limit beyond which even sound of relatively high intensity is liable to be seriously affected by tracing distortion is roughly 2000 c/s; hence the recording characteristic shown in fig. 25, if applied also to microgroove records, would give rise to distortion of about 20 or 30% at the highest frequencies. Now although the third harmonic of 10,000 c/s is far beyond the normal range of hearing, intermodulation between it and the harmonics of other notes may nevertheless produce distinctly audible and very irritating distortion.

Section 3. The modern recording characteristic

Accordingly, the following points should be taken into account in the plotting of recording characteristics:

1 The minimum amplitude should be large enough to ensure that it is not smothered by noise.
2 The maximum amplitude should not be so large as to cause any overcutting of adjaceit grooves; in other words it depends on both the groove width and the groove spacing.
3 Tracing distortion should be kept within bounds.
4 The dynamics of the orginal music should be preserved as far as possible.
5 Due consideration should be given to the maximum intensity of the different notes occurring in the music.

The combination of all these requirements has led to recording characteristics altogether different from that shown in fig. 25. That even smallest amplitudes should be large enough to escape "mashing" by noise and other unwanted sound in the record affects mainly the highest frequencies, since it is here that the amplitude is smallest for a given sound intensity, notwithstanding the fact that the velocitiy of vibration of the needle tip and the air particles is exactly the same as at lower frequencies. Noise is caused by grains in the record material or by surface irregularities in the groove, and the amplitude of the vibrations should be sufficiently large as compared with these irregularities; in principle, then, it is possible to drop the whole of the recording characteristic, with the exception of the highest frequencies, to a slightly lower level. We may take the portion of the characteristic below 2000 c/s, say, 10 decibels lower, keep the same level at 10,000 c/s, and slope the characteristic gradually between 2000 and 10,000 c/s (fig. 36). Of course, this affects the reproduction in several ways. If a disc thus recorded is played on a gramophone designed for records with characteristics as indicated by the thin lines in fig. 36, that is, by means of a pickup type AG 3012 or AG 3013 (see fig. 26), the volume control must be turned up in order to boost the intensity of the notes up to and including 2000 c/s. Also, the tone control must be turned towards more bass to cut down the surplus treble. (Given an amplifier or loudspeaker capable of little treble response, the tone balance will then be correct, although not on records with the old characteristic).

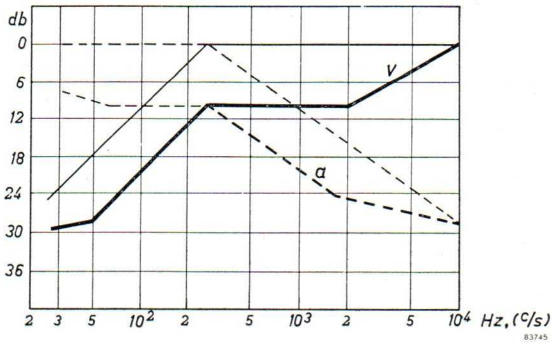


Fig. 36

The basic shape of the modern recording characteristic

In this way, however, we obtain an optimum music-to-noise ratio almost constant over the entire frequency range, and at the same time reduce the tracing distortion of frequencies below 10,000 c/s.

The most noticeable differences in dynamics and the most serious risk of overcutting during fortissimo passages occur at the lower frequencies. The above mentioned modification of the recording characteristic provides more space at these frequencies, thus improving the quality of fortissimo recording without affecting the dynamics. It is on these lines, more or less, that modern recording characteristics have been developed; for further details see section 4.

Instead of being angular, as in fig. 35, real characteristics are flowing curves; nevertheless, this diagram shows the basic shape of all modern recording characteristics. An incidental advantage is that the lowest frequencies, below 70 c/s, can now be recorded with even greater intensity than before, so that the quality of the bass notes is enhanced. So far we have only considered notes of variable pitch, but constant intensity. Tests carried out in concert halls have shown that what are normally the maximum sound intensities (that is, the highest velocities attained by the vibrating air particles) are very much the same for all frequencies up to 2000 c/s. Beyond 2000 c/s the maximum sound intensity decreases roughly 6 decibels per octave with increasing frequency. This means that the amplitude so far accepted as the maximum will probably not be encountered in practice at the highest frequencies, so that, in fact, the tracing distortion is very much less than would appear from the purely theoretical point of view. Often in gramophone recording the sound of certain high-pitched musical instruments is deliberately recorded slightly louder than is consistent with the ratios of sound intensity in concert halls; hence the distortion is not always quite so low. The final result is still that tracing distortion is no longer noticeable in properly recorded gramophone records.

As we have already seen (page 35), appreciable forces are required to move the needle to and fro. In the case of fig. 34 it is assumed that the record groove is perfectly rigid, but in practice, especially with microgroove records, this not so. Owing to the elastic properties of the vinyl the forces acting upon the needle also deform the groove slightly, especially on very high notes of considerable intensity. The result is that the actual treble attenuation is greater than is consistent with the theory of tracing distortion, whereas in fact the non-linear distortion is relatively smaller. The latter effect is very convenient, of course, and the extra treble attenuation can be compensated in the amplifier. This effect is strongest at frequencies above 10,000 c/s, therefore such frequencies, likewise important from the point of view of fidelity in the reproduction of music; can also be recorded and reproduced with so little distortion as to be below the threshold of audibility. This does not imply that every record is necessarily free from treble distortion; it only means that distortion-free reproduction is a theoretical, and practical, possibility.

Section 4. Recording and response characteristics.

It will be evident from the theory discussed in the preceding section that the „ideal" recording characteristic is not a thing that can be computed exactly, and that the choise of the shape of such characteristics is in many respects a matter of experience, so that the final decision of those responsible for plotting characteristics is inevitably influenced by their individual judgement and outlook. The result is that although all modern recording characteristics conform to the general pattern of fig. 36, we still find differences between the characteristics of records of different makes.

This means that in order to maintain good tone balance under all conditions, the tone control on the amplifier must be re-adjusted each time, and that, even then, it may not be possible to bridge certain differences entirely.

In fact, the whole concept of the recording characteristic is really more or less theoretical, except in the case of test records. Such discs are recorded by impressing a pure tone (sinusoidal voltage) of given intensity but variable frequency on the recording amplifier. Provided that the properties of the particular amplifier and recording-head are known, it is quite possible to cut records with a pre-determined characteristic and to check this cutting characteristic later on (see page 137). The characteristics of microphones employed in the recording of music are less easy to determine, but in any case the tone balance ultimately obtained is governed very largely by the acoustics of the particular concert hall. Bass intensity is often very much greater in a hall with a relatively resilient wooden floor than in one floored with cement. Cloth-covered walls in a hall attenuate the high tones. Every hall has its own specific acoustic characteristics, which cannot be reduced to any given acoustical standard by means of a tone control. Consequently the theory of the recording characteristic has become rather indeterminate; for instance, if the recording equipment be placed first in the Large hall, then in the Small hall of the Amsterdam "Concertgebouw" the recordings of a given violin or cello will sound quite different.

In recent years, therefore, the recording characteristic has been superseded by a standard response characteristic. It is in effect the inversion of the recording characteristic. If the recording characteristic shows, say, that a 10,000 c/s note is recorded at 10 db greater intensity than a 1000 c/s note, the one must be attenuated 10 db with respect to the other in the reproduction. During the cutting of the disc the tone control of the recording amplifier is so adjusted that when the record is played on a gramophone having a standard response characteristic, the tone of the reproduction is properly balanced.

In this way all records are made to sound well, instead of some sounding muted and others shrill, without it being necessary to confine all the recordings to one hall and one arrangement of the microphones. Thus it is now possible to use a location and microphone layout which are adapted to the requirements of the particular piece of music, and so bring out all the inherent acoustics of a given hall or studio without any risk of certain recordings acquiring an unacceptable tone balance. Owing to various causes such as the linear distortion arising from tracing distortion, and the fact that the rigidity of the records is not infinite, the final playback characteristic is to some extent distorted, the exact amount of this distortion depending also upon the properties of the particular pickup (needle dimensions, moving mass). Since the loudspeaker and the acoustic properties of the listening-room also have a certain effect on the ultimate response characteristic, differences between individual makes of records will still occur despite the plotting of electrical playback characteristics.

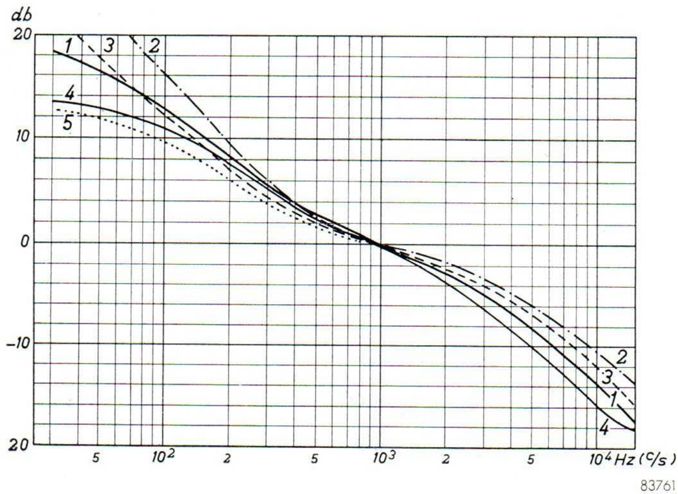


Fig. 37. Response curves for different makes of records

I.E.C	1—1	FFRR	4—2
A.E.S.	3—3	NAB	4—4

See also table under fig. 38.

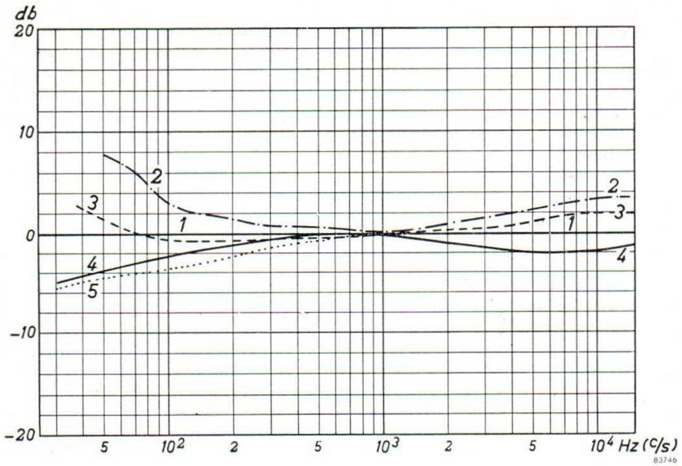


Fig. 38. Response corrections for different makes of records reproduced by means of an amplifier already corrected for records with the I.E.C. recording characteristic.

Make	Bass	Treble	Make	Bass	Treble
Capitol *)	3	3	Nixa, VLP & CLP	3	3
Capitol (Germany)	3	2	WLP & NLP	4	3
Columbia	4	4	Omega	5	3
Concert Hall *)	3	3	Pathe-Marconi	1	1
Decca (U.S.A.)	4	4	Philips *)	3	3
Decca FFRR	4	2	Polydor *)	5	2
Decca (Germany)	3	3	Remington *)	1	4
D.G.G. *)	5	2	R.C.A. (old)	2	1
H.M.V.	1	1	R.C.A. (new)	1	1
London FFRR *)	4	2	Telefunken	3	2
M.G.M.	1	1	Urania	1	4
Nixa approximate for most records	4	4	Westminster	1	4

The starred makes have either already adopted the I.E.C. characteristic or signified their intention of doing so in the near future.

However, these differences may well be smaller than if all manufacturers based their products on the same recording characteristic, and in addition, greater uniformity of tone balance amongst records of the same make is ensured. Unfortunately, the day when all record manufacturers make their recordings to the same response characteristic has still to come, although it certainly will before very long. Also, in view of the fact that all the thousands of records now on the market have been recorded with different characteristics, it will be many years before anything approaching real uniformity is achieved, even when a general standard is adopted, since it is not be expected that every firm will immediately remake all their pre-standard records. For this reason a number of the more important response characteristics are shown in fig. 37. In this connection it is worth mentioning that, owing to the complex international relationships in the gramophone world, it may well happen that certain records bearing the label of a particular firm (X) are really made by another firm (Y) employing their own characteristic; hence it is best to leave the final decision to the ear (as always with gramophone reproduction) instead of relying entirely upon the published characteristics.

The characteristics in fig. 37 are reproduced in fig. 38, but here they refer to the IEC response characteristic, which, it is hoped, will be the future standard. Accordingly, fig. 38 shows the tone compensations required for equipment which, with the tone control at neutral, has the precise response characteristic recommended by the IEC (International Electrotechnical Committee). Since the starting point for all the curves is near 1000 c/s, where all the characteristics are virtually straight, the differences between them can be smoothed out so effectively by means of an effective tone control that for all practical purposes any residual differences are too small to affect the quality of reproduction.

In this way a certain measure of standardization has already been accomplished, which is very much in the interest of the record lover.

The I.E.C. characteristic is also known under other names, such as: R.I.A.A., N.A.R.T.B. or C.C.I.R.

CHAPTER VI

THE CARE OF NEEDLE AND RECORD

Section 1. The gramophone needle.

It is seen from the cross-section of the groove shown in fig. 34b that the needle tip does not rest upon the bottom, but rides upon the side walls of the groove. Although pointed needles produce less tracing distortion, they should not be too sharp, otherwise both the needle and the record will wear too quickly. Moreover, if the needle is sharp enough to track along the bottom of the groove instead of being supported by the side walls, it tends to skid from side to side; such skidding, or „skating" causes distortion and extra noise. Again, the bottom of the groove is not always perfectly smooth and dust collects on it, which also affects the quality of reproduction. In all modern microgroove records, as also in certain standard records, the bottom radius of the groove is 5 microns, so that in theory a needle-tip radius of 10 microns would be ample; experience has shown however, that taking wear and variations of the groove into account, the optimum radius for needles for use with microgroove records is 25 microns. In the case of standard-groove records very much blunter needles are required, partly to keep the wearing on the relatively coarse shellac within reasonable limits and partly because the V-groove (standard groove with a bottom radius of 5 microns) in not yet employed for all such records. A tip radius of 55 microns is sometimes employed, but even this is too small for certain standard groove records; to be on the safe side, then, a still larger radius, 75 microns, is usually employed. Steel needles are no longer used in modern pickups, for the following reasons. Such needles have very sharp tips, which the record soon wears down to the shape of the groove. This was good enough when fitted to heavy pickups for use on shellac records, but with modern light-weight pickups the needle would take to long to wear down and would therefore be very hard on the record.

Vinyl records would suffer even more, steel needles damage them very seriously. The materials most widely used for needles nowadays are hard metal, sapphire and diamond. The shape of the needle is always the same: cylindrical with a 45° conical tip (the angle of the groove is roughly 90°). On the other hand the life of such needles varies considerably, as borne out by the figures given in various trade journals.

The following factors, amongst others, affect the life of a needle:

- the record material;
- the condition of the particular record: new or old, clean or dirty;
- the needle pressure;
- the properties of the pickup; compliance, moving mass, resonances;
- the shape and mass of the tone arm;
- the friction in the tone arm bearing;

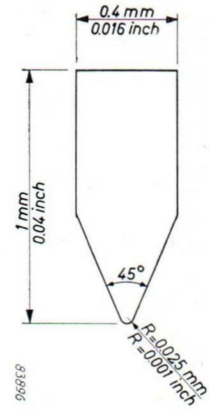
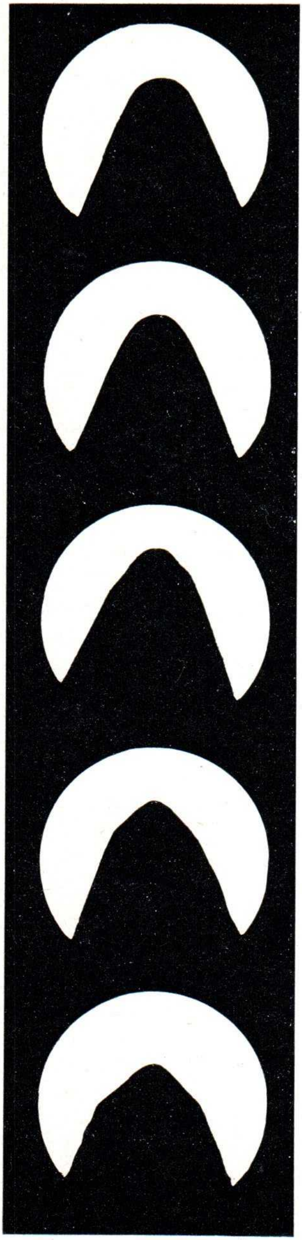


Fig. 39
Microgroove needle

the size and nature of the particular record collection. To judge the life of a particular type of needle accurately, then, we should keep these factors as far as possible constant. A record coming straight from the press causes less needle wear than one which is more than a month old. The greater the moving mass of the pickup, the greater the needle wear. Since quite a large part of the mass of the pickup is concentrated in the needle, the needle wear depends to some extent upon the dimensions of the needle itself. In the following we refer exclusively to needles having the dimensions shown in fig. 39, as employed in Philips pickups; as far as is known they are just about the smallest needles obtainable. For the tests on microgroove records we employed a pickup type AG 3013 and Philips record changer type AG 1000; the needle pressure was 11 grams in every case. All the discs used were orchestral recordings, unused and more than a month old at the beginning of each test, and the experiments were carried out in an atmosphere containing roughly the same amount of dust of very much the same composition as in an ordinary living room. Fig. 40 depicts the life story of a sapphire microgroove-needle tested for 225 hours on one side of a 10-inch microgroove record. Although needle wear can be seen through a microscope after only 25 hours, it is then so slight that it does not show very clearly in the stereotyped reproduction. If this needle were used with High Fidelity gramophone equipment, experienced listeners would already notice a slight falling off in the quality of the reproduction, especially towards the end of the record, where the shortest wavelengths occur; in most equipment, however, the effects of needle-wear do not become audible until it reaches the stage shown in big. 40b and, in certain cases, when a pickup with characteristic 3b in fig. 28 is employed, these effects would even then not be very noticeable; here, audible distortion occurs only when the needle is worn to the extent shown in fig. 40c. Since the increase in distortion is only very gradual, there is even a risk that the listener will become accustomed to it and therefore continue to use needles in the condition shown in fig. 40d and e. Needles so worn endanger the record, as will now be explained.

Fig. 41 shows how a worn needle rides in the groove. Diagram a is the end view; since the needle rides too deep in the groove the tip is liable to touch irregularities at the bottom. Diagram **b**, the top view, shows the needle in three different positions. In effect, the edges of the wear-facets work like chisels cutting away the walls of the record groove and severely damage it, the more so where notes of high intensity occur (2 and 3 in diagram **b**). The amount of damage the needle shown in fig. 40e is capable of doing will be seen from fig. 42, showing at the top a photomicrograph of a new record and at the bottom the appearance of part of the same record after having been played twice with a very worn needle. Plainly it is better for the records if the needles are not used too long.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It is not easy to prescribe an exact playing time for sapphire needles. Large-scale tests have shown that the records are not seriously endangered until the needle reaches the condition shown in fig. 40c, but that it is essential to renew the needle as soon as it reaches this condition. It will be evident from fig. 40 that the rate of needle wear increases the longer the needle is used. One explanation is that the needle continually deposits sapphire dust on the record and is itself attacked by this dust. The needle shown in fig. 40c was tested for 125 hours on one side of a particular record, so that all the sapphire dust from it accumulated on this record. In a similar test on 5 different records (10 sides), the needle took roughly 160 hours to reach the condition shown in fig. 40c. On the other hand, such tests are carried out under better conditions than would normally occur in practice, where a certain amount of jolting and jarring of the gramophone cannot always be avoided; therefore 125 hours is probably a safer limit for this kind of needle. It is just because the needle wears more rapidly the longer it is used that we must take care not to wait too long before renewing it.

At the same time this is by no means the whole story. Similar tests carried out near a factory workshop in which much abrasive dust is produced have shown that such conditions double the amount of needle wear. This illustrates the importance of keeping the records free from dust of this kind.

Wear data as given in various periodicals and books are very much less favourable than the above figures. This is partly owing to the use of inferior sapphire and inferior workmanship, which may increase the rate of wear in certain cases by as much as three times

Fig. 40. Sapphire microgroove needle, new and after being used for 75, 125, 175 and 225 hours

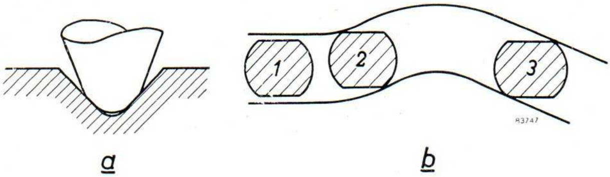


Fig. 41
How a worn needle damages the groove

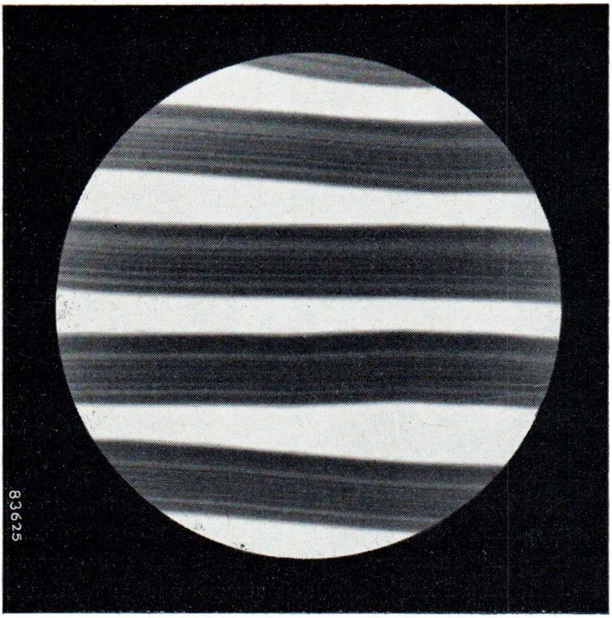
(condition shown in fig. 40c is then reached after only 40 hours); the effect of the quality of the sapphire needle on itself should therefore not be underestimated. Pickups with low compliance and large moving mass likewise produce a relatively greater amount of needle wear; too much needle pressure again results in much wear, but so also does too little pressure. This is a hint to record lovers who give preference to extra low needle pressure; moreover, a needle resting too lightly in the groove may overshoot and cause distortion during loud passages, thus accelerating both needle and record wear.

The record material may affect matters considerably; vinyl wears the needle least of all and mere colouring agents like carbon black have very little effect on needle wear. On the other hand certain microgroove record compounds containing not only vinyl, but also a fairly large amount of cheaper substances badly affect the amount of wear. We can quote one instance, although indeed an unusual one, in which the needle wore 5 times more quickly than with records made of a material of standard composition.

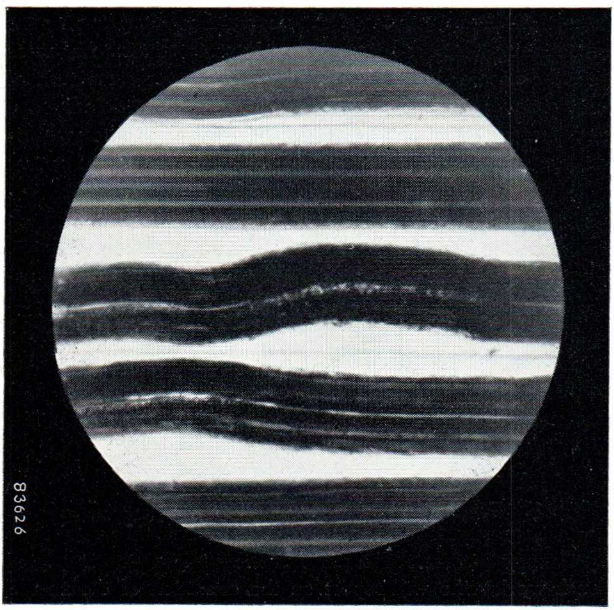
To reassure those who have heard dismal stories about needles becoming worn after only 10 hours it is worth mentioning that for reasons already stated the rate of wear increases considerably when the needle runs time after time along the same turn of the groove (e.g. the run-out groove). Tests carried out on a record with only one circular groove have shown that under such conditions the rate of needle wear is extraordinarily high; however, such tests have no bearing whatever on practical record playing.

Since gramophone owners cannot very well judge the different conditions, and usually have only a very vague idea of the number of hours that the needle has been used, various attempts have been made to find a suitable method of inspecting gramophone needles. It is best to employ a microscope of 80—200 time magnification; with a smaller magnification wear can be detected only when the needle is very seriously affected by it, and with a larger magnification too much of the needle is outside the field of view, so that many signs of wear go undetected.

It is also best to view the needle along the worn faces (as in fig. 40), since less experienced observers viewing such faces at right angles cannot recognize the signs of needle wear until it reaches a very advanced stage. Fig. 43 shows a special needle microscope with a pickup support to bring the needle into focus almost immediately. Another method of needle testing, sometimes employed in America, is to take an impression of the needle tip in very thin metal foil and have this impression inspected by a firm specializing in such work, who will report on the condition of the needle.



a)



b)

Fig. 42. a) Photomicrograph of a new record;

b) photomicrograph of the same record after it has been played with a damaged needle

However, this is reliable only when the needle wear is quite far advanced.
Still another method is to employ a special gramophone record made of very soft material, which suffers visible damage when played with a worn needle. The drawback is that materials soft enough to be visibly damaged by a not-too-badly worn needle are likewise damaged by new needles, whereas when a harder material is used only very badly worn needles — already far beyond the point at which they become a danger to music records — leave a clear trace on the test record.

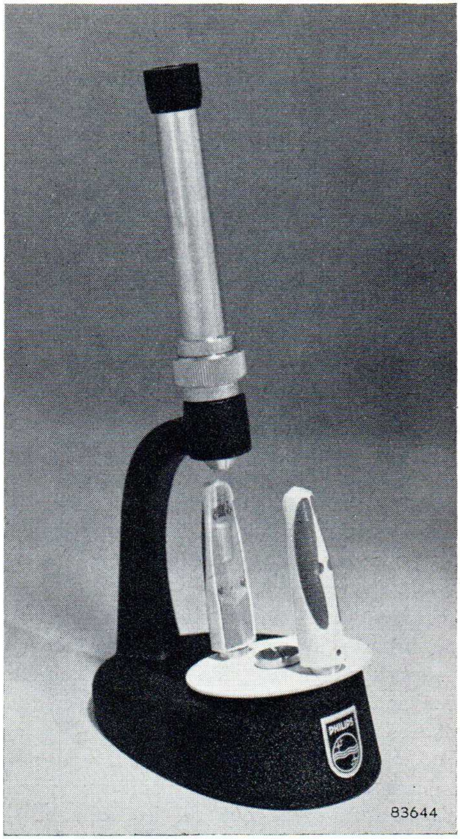


Fig. 43. Needle microscope

Section 2. Hard metal, sapphire and diamond

The material most widely used at the moment is probably sapphire. As stated in the preceding section there are appreciable differences in quality between sapphire needles of different makes; the cause of these differences lies in the specialized knowledge of, and the care taken by, the manufacturer and to a certain extent in variation in the quality of the material itself.
Sapphire is the transparent variety of corundum and it occurs naturally, but can also be produced synthetically by melting aluminium oxide under high pressure. Natural sapphire has a blue tint owing to the presence of certain impurities, whereas synthetic sapphire is almost colourless. Both varieties can be used for gramophone needles, but synthetic sapphire is more suitable for this purpose by virtue of its greater homogeneity.
After diamond, sapphire is the hardest of materials and it takes a very high polish. The specific gravity is low, therefore the mass of the sapphire needle is extremely small; this is a very important advantage from the point of view of needle talk, and of needle and record wear. Ruby is another variety of transparent corundum with a red instead of a blue tint; but it is rarely used for gramophone needles.

One of the metallic elements used in alloys for gramophone needles is osmium; others are cobalt and titanium carbide. Osmium is the heaviest of all substances — more than 1½ times as heavy as lead and 8 times heavier than sapphire. Usually only the extreme tip, instead of the whole needle, is made of osmium alloy; this tip is inserted in a shank of some lighter and more malleable metal. A good osmium needle wears about 4 times as quickly as a good sapphire needle and just a little more quickly than a sapphire needle of poor quality. Hence osmium needles should be changed after about 30 playing hours to ensure that the records will not be severely damaged by facets produced by wear. Although osmium is softer than sapphire, it is nevertheless very much harder than the record material, and a worn osmium needle causes just as much damage as any other needle in the same condition. An alleged advantage of osmium is that it is less likely to crumble at the edges, say, as a result of rough handling. It is the writer's experience that although this claim may hold good in theory it carries very little weight in practice, since with a properly ground needle of sapphire or diamond, even when it is roughly handled, there is very little likelihood that chips of the needle material will crack off. Other metallic needles, e.g. those made of tungsten carbide, are very much the same in this respect as osmium alloy needles.
Since there is nothing harder than diamond, diamond needles last a very long time. Such needles are very expensive, not only because the material is so costly, but also because it is expensive to cut. Whereas sapphire needles can be ground to the required shape relatively quickly by means of diamond dust, this dust being very much harder than the sapphire itself, the shaping of a diamond needle takes nearly twenty times as long. Synthetic diamond, although it has been made in laboratories, has not yet a practical value. Diamond needles are made either from chips of diamond or from whole diamonds. Whole diamonds, as used for the needles in Philips pickups, have this advantage that the needle can be cut in such a way as to ensure uniform wear, that is, not more on one side than on the other. Tests have shown that a diamond needle wears about 10 times more slowly than a good sapphire needle, which means that, for music at any rate, the period of absolutely undistorted reproduction is about 250 hours. As will be explained in the following section, however, the difference in practice is very much greater; here there is also less record wear, partly because the diamond needle retains its original shape far longer, and partly because it leaves less of its own dust in the groove. For all practical purposes needles of steel, wood, cactus and bamboo are no longer used. Since the life of such needles is shorter than the playing time of one side of a long-playing record, it would necessitate either changing or re-grinding the needle during the playing of such records. Moreover, the needles in modern pickups are so small that repeated needle-changing would be very inconvenient. The above kinds of needles were very useful when the needle pressure was 2 oz. (50 grams) or more, sapphire and diamond needles being unsuitable for such pressures, but the present-day pressures have so reduced record wear that the use of very soft cactus needles does not improve matters perceptibly, but merely results in a very much weaker treble response.

So far we have discussed only microgroove needles. It would be reasonable to suppose that standard-groove needles, having a blunter tip, would wear more slowly, since the needle pressure is then distributed over a larger area. This is true enough, but carries no weight in view of the fact that the shellac compound of 78 r.p.m. records is more abrasive than the material used for long-playing records. All the figures given in the above therefore apply equally well to standard-groove needles.
Another important question is what happens when records originally played with steel needles are afterwards played with sapphire needles — or when a sapphire needle is replaced by a diamond? It might be feared that the grindings deposited on the record by the old needles will damage the new needle. This is unfounded, however, since all the grindings, even those deposited by steel needles, are so small that there is no risk of chipping splinters from needles made of precious stone. Seeing that in either case the grindings from the old needle are softer than the new needle, the effect upon the needle wear will be very small. The writer has proved to his own satisfaction that sapphire and diamond needles used for playing records which had been used with steel needles in his younger days, wear quite normally. On the other hand, records already played many times with diamond needles should not be played with sapphire needles, as this may well result in increased needle wear.

Section 3. Gramophone records and how to take care of them

Many a record lover will at some time have estimated the amount of money tied up in his record collection and will have been appalled at the result of his calculations, but, however sincerely he may resolve to go easy in future, such resolutions are usually in vain. At the same time it is only logical that every possible precaution will be taken to ensure the safety of so valuable a record collection, and as part of this policy it is essential also to consider the question of needles from the point of view of the records themselves. It is believed in some quarters that record wear is largely due to the high temperatures occurring at the contact surface of needle and record. From this it is concluded that the needle should not be too hard, since a relatively softer needle becomes slightly compressed and thus presents a larger contact surface, giving a lower pressure per square inch, and a lower temperature. Again, a softer needle wears more quickly, thus enlarging the contact surface.
As a matter of fact, comparative tests with sapphire and diamond needles have shown that when a new record is played with a new needle, the sapphire produces the smallest amount of needle wear during the first few hours, mainly because it takes a smoother finish. Not so, however, when once the two different needles have been used for 5 to 10 hours. This is long enough for the sapphire needle to acquire facets such as to affect the wearing of the record. It is worth mentioning, however, that the record wear in question is very slight and is discernible only with the aid of the most sensitive equipment and carefully selected records, as also that brand new records were employed for purposes of comparison.
If the particular equipment or record contains a certain amount of distortion, or if the record has no fortissimo passages, the effects of such initial wear will not be noticed, even by the most experienced listener.

Further use of the needles shows that the facets worn on them, and the cutting edges thus formed are the principal causes of record wear, so that also from this point of view diamond needles are in the long run the most satisfactory, with sapphire as second choice and hard metal as third. Another important point to be considered in this connection is that diamond has the lowest coefficient of friction, this being even lower than that of sapphire. It is found that the harder the needle the smaller the amount of needle dust deposited in the record grooves and the less noticeable the effect upon needle and groove next time the record is played.
One difficulty is that there is no reliable method of measuring record wear. Although some idea of the amount of distortion occurring after a given number of playing-times could perhaps be obtained with the aid of certain test records, the practical value of such tests would be very small, since wear affects mainly the fortissimo passages and it is precisely these passages which offer no opportunity for measurement.
Record noise, also, can be measured only on special test records and during intervals in between the music, whereas it is in fortissimo passages that the greatest increases in unwanted sound occur. Since there is no alternative, then, it is necessary to employ subjective listener tests. A series of such tests has shown that needle wear and record wear are more or less closely related, that is, that if the particular needle begins to wear more quickly at a given moment or under certain conditions, so also does the record. Accordingly, many of the conclusions concerning needles mentioned in the preceding pages also hold good for records.
Well-preserved microgroove records are for all practical purposes everlasting. As a case in point, not one of 50 listeners present at a test in which a microgroove record already played sixty times was compared with a brand new disc was able to detect any difference between the two. It is true that if this test had been repeated several times some of the listeners would probably have noticed a slight variation in the fortissimo passages, but this does not alter the fact that the record wear after sixty playings is obviously very slight. Experience has shown that the records employed to demonstrate High Fidelity equipment (ordinary commercial records) can be used up to 200 times. Even then, although it is advisable to replace them, the loss of quality is so slight that it does not make such records any the less enjoyable when played on High Fidelity equipment, apart from which, given an amplifier with a more limited range of response or with a certain amount of distortion (that is, not quite up to Hi-Fi standards) the difference with respect to a new record is barely perceptible.
After 500 playings records lose much of their original quality; nevertheless, when subsequently played on equipment with only a limited response characteristic they are by no means as bad as might be expected as regards distortion and noise. In fact, such „old-timers" are still playable even on High Fidelity equipment. After 1000 playings, however, the quality is so bad that the records are really unusable, although new records played on a bad gramophone often sound no better and are nevertheless listened to with enjoyment by the proud owner.
No doubt many readers will consider the above figures altogether too flattering, and

it is true that without adequate care the records will lose their original quality very much more quickly. The principal condition for the lasting enjoyment of a record collection is that worn needles are never used. Also, of course, the tone arm must be able to move freely. The needles shown in fig. 40 are worn more on one side than on the other, the most severe wear occurring at the side nearest to the centre of the record. This is owing to the centripetal force acting upon the pickup, which has already been explained with the aid of fig. 33. If the tone arm bearing is too tight, however, the needle will wear most rapidly on the outside, and the groove will also suffer more. Here we have one more point to be considered, for the safety of the records, when needles are inspected.

Another condition is that the records are kept free from abrasive dust. Part of the dust floating about in living rooms is soft — fluff from plants or fabrics — and part of it is hard — grains of sand and chalk, metal particles and very fine grit from various sources.

The danger lies in this hard, abrasive dust, which is also the most difficult to remove. Records should therefore be kept in their covers when not in use. Brushing the records before playing them does not help very much, since it only removes some of the dust that has penetrated into the grooves and at the same time charges the records electrostatically, so that they attract more dust. Although there are a number of quite satisfactory microgroove record brushes on the market, a slightly moistened or a specially treated cloth like the Discleaner is still the best dust remover. Cast-off nylon stockings are also excellent for this purpose. Any relatively large dust motes, chalk for instance, should be blown off the record first. Before putting a record down either on the turntable or elsewhere, see that the receiving surface is clean. Whereas occasional hard granules are less dangerous when they occur on turntables covered with felt or velvet, since they then tend to sink into the soft covering rather than into the record, on rubber or plastic covered turntables, which do not absorb such particles, it is invariably the record which suffers. The inside of the dust cover should also be free from dust; glued seams tend to scratch the record and should be avoided. Dust-covers with folding inner covers are best because they enable the record to be placed in, instead of pushed into, the inner cover, so that any abrasive dust which has settled on the record is not ploughed into the latter. Covers made of material capable of absorbing abrasive particles, e.g. filter paper, are better than plastic covers, which do not eliminate the risk of record-scratching. Even though fluff might come from the filter paper itself, this fluff is too soft to damage the records. Any fluff on the record clings to the reproducing needle and should be removed from there carefully from time to time with the finger or with a soft brush. The same applies to fibres coming from fabric covered turntables. Dust-covers without an inner cover should be opened out slightly before the record is inserted or withdrawn, so that the latter slides easily in or out without scraping against the paper. Patent record cleaners, with or without anti-static properties, should not be expected to do the impossible; in any case, a few drops of water on a cloth are just as effective as most of the highly specialized cleaning fluids. Liquids which are sprayed on to the record and left to dry gradually sometimes attack the covering or the

metal parts of the turntable; hence they should be used very carefully.
Records which have been played at a party sometimes bear traces of the refreshments served on that occasion. Although relatively harmless in themselves, residues of butter, cream, liqueur and gin on a record attract dust, which is dangerous to the record. Such spots and splashes can be removed by sponging the record carefully with a very weak, almost cold solution of some soapless detergent and then rinsing it thoroughly with clean water.
Nearly all stain-removers attack gramophone records, that is, with the exception of ethyl alcohol, and even this should not be used indiscriminately since it may happen, of course, that a certain manufacturer will change the composition of his records at a given moment to include a constituent soluble in ethyl alcohol.
In general it is advisable to touch only the non-playing parts of the records in order to avoid finger marks on the grooves. If the particular record is too large to be handled conveniently in this manner, however, there is a danger that it will be dropped; hence it is sometimes wiser to risk leaving a few finger marks. Provided that the records are kept free from dust and played only with needles in good condition they will last a very long time, and as a rule any falling off in quality in these circumstances is caused mainly by scratches resulting from careless handling. This can also be avoided, and all that is then necessary is to store the records properly. As far as possible, gramophone records should be stored upright; they should never be stored at a slant, and stacking is permissible only on condition that the stacks are not too high and that all the records in each stack are the same size, otherwise they will certainly warp.
If a particular compartment of the record cabinet does not contain enough records to fill it properly and the records therefore topple over, the surplus space should be packed with books or boards to hold the records perfectly upright. It is best to keep long-playing records in their dust-covers, and shellac records, if possible, in albums. Records which nevertheless become warped to such an extent that they slip on the turntable, or wear unevenly owing to being played with a rather heavy pickup, can be straightened by warming them very carefully on a flat glass plate. This can be done in summer by placing the record in the sun, and in winter by placing it near, but not too near, to an electric stove. The record should be warmed until it flattens under its own weight; on no account should any attempt be made to speed-up this process by bending the record. Another possible cause of warping is stress set up in the material during the pressing process. Although this can be remedied in the above-mentioned manner, there is no means of ensuring that the particular record will not warp again for the same reason at some time after the treatment. Records which are warped, but so slightly that this does not affect the playing of them are best left alone. It is always advisable to store gramophone records in a fairly cool place, that is, not close to a stove or a radiator. At the same time a very dry atmosphere is likewise undesirable, since microgroove records then attract dust more readily.

CHAPTER VII

RECORD PLAYERS AND RECORD CHANGERS

Section 1. The motor

From the point of view of construction there is no essential difference between record players and record changers; both contain a tone arm, a motor-driven turntable and a controller, operated in a certain manner by the tone arm, to actuate the switch or changer mechanism when once the reproducing needle reaches the run-out groove of the record. It is precisely the elements common to both which have the greatest effect upon the performance of such machines. Possible differences in quality between record changers and record players have nothing whatever to do with the fact that the former have changer mechanisms and the latter have not, but can usually be traced to the turntable, the motor, or the tone arm.

Almost all modern gramophone motors are cage-wound and employ inductive phase shift. Commutators are now used only in D.C. motors, so that they alone are apt to produce interference which may be amplified into a noise in the loudspeaker. The speed of the induction motors is so steady that they do not require a centrifugal governor. Experience has shown that, provided the load does not vary too much, and that the mains voltage variations do not exceed 10—20%, the speed remains constant to within some tenths of one percent.

In fact, the speed of the motor is noticeably affected only by mains frequency variations, to which it is proportional. Since the frequency of the lighting mains is usually constant, with only occasional, and very slight, variations during peak hours, however, this is also a minor problem.

The principle of the squirrel cage armature gramophone motor will now be explained with the aid of fig. 44. Here we have an armature rotating between the pole pieces of an electromagnet energized by a coil (L) connected to A.C. mains. Both pole pieces of the stator, as it is called, are slotted and provided with short-circuiting turns passing through the slots in the manner shown in the diagram.

The current flowing in the coil produces an alternating magnetic field from the one pole piece, through the armature, to the other piece; the direction of this field is indicated by dotted lines. Fig. 44b shows the variation in field intensity with time. Beginning at the zero line, the flux from the larger parts of the two pole pieces (chain lines) increases to a certain maximum, then falls off again, changes direction, and so on. The flux from the parts of the pole pieces provided with shorting rings varies in the same way, but with a slight delay owing to the two rings.

Accordingly, the overall flux is supplied by the larger parts of the pole pieces after ¼ cycle, and by the ringed parts after ½ cycle when it is usually slightly weaker. After ¾ cycle the magnetic field, then in the opposite direction, is again supplied by the major portions of the poles, and at the end of the full cycle it proceeds, likewise in reverse, from the smaller parts. At intermediate times both parts of the

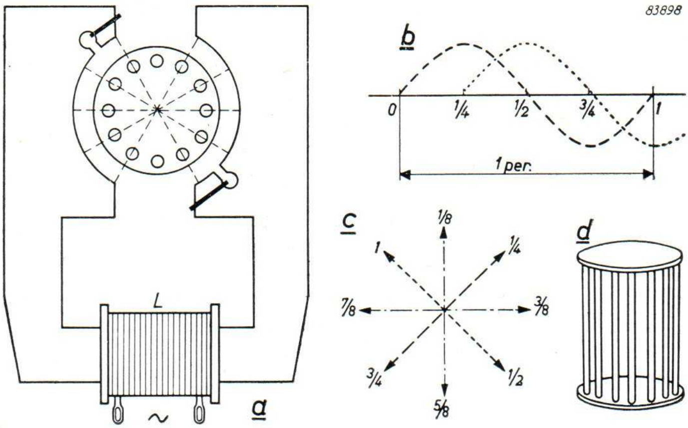


Fig. 44. Principle of the 2-pole gramophone motor

pole pieces contribute to the overall flux. The sum of the two fields then produced is roughly equal to the field intensity after ¼, ½ or ¾ cycle, but the direction has then changed with respect to the direction after ¼, ½ or ¾ cycle. The full cycle takes place 50 times per second when the mains frequency is 50 c/s, and 60 times per second when this frequency is 60 c/s, as will be seen from fig. 44c. It can be shown that very much the same result would be obtained by rotating a bar magnet 50 (or 60) times per second.

Now, the armature, or rotor, of our gramophone motor is shown separately in fig. 44d. It consists of a number of copper rods (e.g. 13) soldered to copper discs and covered with iron. The iron enables the magnetic flux to pass freely through the rotor and ensures that it does so in the right direction. Since the magnetic field rotates, is moves in relation to the copper rods and therefore induces voltages in them in accordance with the principle already explained in connection with pickups. The copper discs connecting the rods enable currents to flow in them, and these induced currents, in turn, produce another magnetic field. If we imagine that this field is also associated with a bar magnet it will be easy to understand what happens: this hypothetical bar magnet is attracted by the one representing the stator field. This rotates 50 times per second, and the rotor magnet tends to follow it. In effect the armature magnetic field is connected to the armature itself, so that this rotates with the field. In practice the speed of the armature is not exactly 50 revolutions per second, or 3000 r.p.m.; intead it is slightly lower, say, 2950 r.p.m., the reason being that if the armature did rotate at exactly 3000 r.p.m. the stator magnetic field, whose exact speed this is, would not move in relation to the rotor rods and therefore would not induce any voltages in them.

Apart from this two-pole motor there are other models with four poles (fig. 45). Both types operate on the same principle, but the speed of four-pole motors is only

half that of two-pole models, therefore it is slightly less than 1500 r.p.m.
The relative advantages and disadvantages of four-pole and two-pole motors are not always properly understood. For instance it is considered in some quarters that the one should run more steadily than the other because its armature receives twice as many thrusts from the magnetic field during each revolution as that of the two-pole motor.

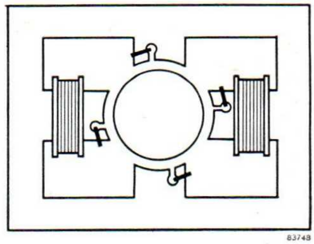


Fig. 45
4-pole gramophone motor

In fact, steady running depends upon the number of force-impulses per second, which is the same for both types. Although with internal combustion engines an eight-cylinder model always runs more smoothly than one with only four cylinders, this is because the speed of rotation is practically the same in both cases, so that the eight-cylinder engine receives roughly twice as many impulses per second as the smaller model. In electric motors, however, this difference does not exist; on the contrary, it is the two-pole motor which runs more steadily thanks to its greater speed. Given equal rotor-weight, the two-pole motor accumulates four times as much energy as the four-pole model (this energy is $\frac{1}{2}$ m ω^2, where m is the mass and ω the speed of rotation). Four-pole motors are usually larger and have a heavier armature than two-pole motors. On the one hand this boosts the power in the rotor, but on the other hand heavier pulses are required to turn the rotor, and they affect the rumble-level unfavourably. Another point to be considered is that even a small amount of armature-wobble owing to play, however slight, in the bearing, will cause the heavier armature to thrust harder against the idler wheel than would the lighter armature of a two-pole motor.
Uniform rotation of the armature is of course most important, and for this reason as many as 10 short-circuit rings are included in the motor shown in fig. 46. This ensures a more uniform variation of the magnetic field, so that in effect the hypothetical magnet now remains of a constant strength and rotates smoothly instead of jerkily, thus causing very much less rotor vibration. Play in the bearings must be kept to a minimum. This is sometimes accomplished by making the shaft a little on the thick side and running the motor in until the shaft just turns easily in the bearing. However, one serious disadvantage of this method is that shaft and bearing wear is not confined to the running-in period, so that play in the bearing, as also an increase in motor rumble, occur after the motor has been in service for a few months.
The shaft of the motor shown in fig. 46 is ground in the ordinary way and then given a special treatment (known as "superfining"), which removes all the scratches caused by the grinding and produces a finish so smooth that the shaft does not wear even when it has been in use for a very long time. Of course this necessitates great accuracy in the manufacture of the bearings, therefore the machining processes carried out on the shaft and the bearing are very much the same as those employed by watch makers.

To ensure reliability in service it is also necessary that the rotor be properly balanced. This can be done by resting the motor shaft upon two parallel blades, so that the heaviest part of the rotor swings down (in the same way that with a bicycle wheel, for example, the valve gravitates to the lowest point). Holes are then drilled in the underpart of the rotor to restore the balance. On the other hand it is very much better to determine the centre of gravity of the rotor by rotating it at the normal speed on a dynamic balancing machine. During this test the rotor is spot-lighted by means of a stroboscopic lamp; the point of overweight is indicated by a number appearing each time in front of a pointer (see fig. 46). Unlike static balancing, dynamic balancing can ensure exact equilibrum at both ends of the shaft.

Another important question is how powerful should the motor be? The power required to rotate a turntable carrying a 12-inch record at the proper speed is roughly 50 milliwatts; slightly more when there are 10 records on the turntable, because this increases the friction in the bearing. Supposing the efficiency of the transmission between motor and turntable to be 50%, the output of the motor should then be 100 milliwatts, or 0.1 watt. Although the efficiency of gramophone motors generally is low, we may safely conclude from the above that motors consuming only a few watts are quite powerful enough.

The motor driving the Philips battery gramophone is rated at less than 1 watt. It is usual to employ a motor developing more than the minimum amount of power required, however, to ensure that the turntable does not take too long to reach normal running speed after switching on, and, in the case of changers, to provide extra power for driving the changer mechanism.

An 8-watt motor is ample, seeing that the power of the motor makes very little difference to the constancy of the turntable speed, as will be explained later, and that too heavy a motor not only generates too much heat, but also gives rise to excessive motor-rumble. Heavy motors are no more suitable for gramophones than heavy diesel engines are for private cars.

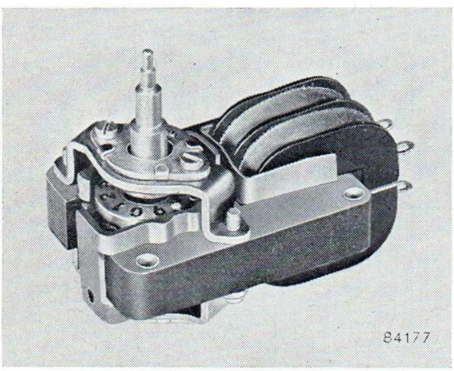


Fig. 46. Motor of record-player AG 2004

In almost all cases what are known as idler- or intermediate wheels are used to transmit the motion of the motor shaft to the turntable. For all practical purposes worm and wormwheel transmissions in gramophones are a thing of the past, partly because they are difficult to use in conjunction with three turntable speeds, partly owing to the fact that much of the unavoidable motor rumble is then transmitted to the turntable, and finally because they are expensive.

Fig. 47 illustrates the basic

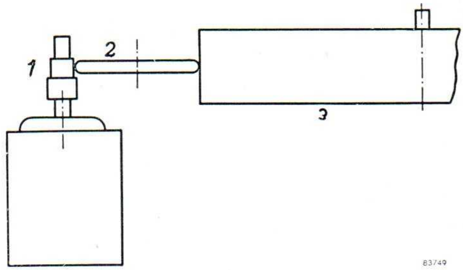


Fig. 47. Three-speed turntable-drive: 1) roller; 2) idler wheel; 3) turntable

principle of a three-speed drive involving an idler wheel. Note that the roller (1) on the motor shaft has three different diameters. The idler wheel (2) presses against the roller (1) on the one side and against the turntable (3) on the other; hence it rotates with the motor shaft and transmits the motion of this shaft to the turntable. The speed of the turntable is governed only by the speed of the shaft and the diameters of the roller and the turntable in the particuler position of the idler wheel; it does not depend upon the diameter of the idler wheel itself.

It is possible to demonstrate the above in the form of a simple calculation. If d_1 is the diameter of the roller, d_2 the diameter of the idler wheel and d_3 that of the turntable, and the speed of the roller be n_1 r.p.m. then the speed of rotation of the idler wheel is: $n_2 = \frac{d_1}{d_2} \times n_1$ (as with gear wheels, the speeds of rotation are inversely proportional to the diameters).

The speed of the turntable is:

$$n_3 = \frac{d_2}{d_3} \times n_2 = \frac{d_2}{d_3} \times \frac{d_1}{d_2} \times n_1 = \frac{d_1}{d_3} \times n_1.$$

The diameter of the idler wheel is therefore eliminated from this formula. Supposing that the diameter of the turntable is 10 inches and the speed of the motor is 2950 r.p.m., the required pulley diameters for the three turntable speeds will be:

0.112 inch for $33^1/_3$ r.p.m.
0.1512 inch for 45 r.p.m.
0.2624 inch for 78 r.p.m.

As the speed of the turntable depends to a very large extent upon the diameter of the roller, this must be very accurate, and the tolerances are therefore of the order of 10 microns. Even with the most accurate finish, however, it is sometimes necessary to give scope for a certain amount of adjustment; this can be done by tapering each of the three running faces of the roller slightly, so that the speed can be regulated even more accurately by varying the position of the idler wheel. Another method is to apply the drive to the bottom face of the turntable instead of to the side (see fig. 54).

The idler wheel is a most important part of the transmission. The supposition that this wheel does not affect the speed of the turntable holds good only on condition that it is perfectly circular. If the wheel is not quite round its speed of rotation varies. Such speed variations are transmitted to the turntable, therefore the pitch of the

music fluctuates at a fairly fast rate, very much to the detriment of the quality of reproduction; another possible consequence is transmission-rumble.
Since idler wheels even when they are not made almost entirely of rubber are at any rate provided with rubber running faces, it is by no means easy to make them perfectly circular. The final result depends very much upon the composition and quality of the rubber. As the rubber rim cannot be moulded accurately enough, it is ground down on a special high-speed grinding machine until the periphery is a perfect circle. Rubber is used for two reasons:
Firstly because it does not transmit the unavoidable background rumble of the motor to the turntable. For example, the transmission shown in fig. 47 interposes two layers of rubber between motor and turntable, the one between the pulley and the idler wheel and the other between this wheel and the turntable.
Secondly, the rubber running face for all practical purposes eliminates slipping, both of the idler wheel against the roller and of the turntable against the idler wheel. Although from this point of view it would be best to make the idler wheel as far as possible of the softest of rubber, in practice it must not be too soft, since the rubber then gives way too much to elastic deformation by the forces acting upon it, which also causes the turntable to run unevenly.
The choosing and fashioning of the material are therefore very difficult, the more so because rubber of a composition not suited to the purpose tends to harden.
Again, it is no simple matter to fix the idler wheel. It must ride against the roller and the turntable with a certain, fairly slight, pressure, so that rigid mounting of the idler wheel bearing is out of the question; moreover a lever-operated retractor must be provided to shift the idler wheel to different parts of the pulley, and at the same time, this wheel must not be given unrestricted play. Fig. 54 shows a typical solution to this problem.
Another difficulty is that rubber under pressure for a long time becomes permanently deformed. In a playback machine long unused, the motor roller constantly presses against the same part of the idler wheel and may therefore flatten the rubber permanently at this point, causing an unpleasant rumbling in the machine next time it is used. Afterwards, when the machine has been in use for some time, this rumbling dies down or may even disappear altogether; at any rate the extent to which it persists depends upon the quality of the rubber. On the principle that prevention is better than cure, the speed selectors of the latest Philips record players and changers are provided with a fourth position enabling the idler wheel to be drawn away from the motor roller and the turntable.
Although the turntable bearings of players and changers require only occasional greasing, it is nevertheless as well to mention that the grease used for this purpose must not on any account be allowed to come in contact with the idler wheel, as this is certain to be attacked by it; oil is not recommended for the lubrication of the turntable bearings.

Section 2. The turntable

The turntable must be driven at an accurate and perfectly uniform speed. Even if the speed of the motor is perfectly constant, it does not necessarlily follow that the above requirement will be fulfilled. Fortissimo passages have a certain braking effect on the record, and therefore also on the turntable, which causes the rubber part of the idler wheel to be suddenly deformed and the speed of the turntable to be reduced for a time. After a moment or two the rubber returns to something like its normal shape, and, although the needle still presses rather heavily on the record, the turntable resumes its original speed. Exactly the opposite takes place after the particular fortissimo passage. This is very much the same as what happens when a toy on wheels is dragged across a table on the end of a piece of elastic. Given a steady pull on the elastic, this will stretch to a certain length. If the wheels of the toy happen to encounter a slight obstruction, say, a fold in the tablecloth, it will slow down for a moment, during which the elastic stretches again, but as soon as the initial shock is absorbed the toy will move forward at its original speed. Exactly the opposite, that is, a momentary acceleration, takes place when once the toy has passed the obstacle. On the other hand, if the particular toy is heavy enough it will be carried over the obstacle by its own impetus and will therefore travel at a very much more constant speed. So also with turntables; experience has shown that if they are heavy enough the speed of rotation remains sufficiently constant despite any sudden variations in the friction between needle and record.

In general there can be no precise definition of what is „heavy enough". The greater the compliance and the smaller the moving mass of the pickup, the lighter the turntable required. It will be evident that this also depends on the turntable bearing; with a heavy-running bearing it is all the more difficult for the turntable to maintain constant speed.

Properly designed turntables also smooth out speed fluctuations originating in the motor, say, owing to sudden, transient variations in the mains voltage, and others caused by imperfections in the idler wheel.

Other conditions which must be met in order to avoid speed fluctuations are, of course, the smallest possible amount of play in the turntable bearing, that the surface of the turntable to which the driving force is applied be perfectly smooth, the rim of the turntable (if it is rim-driven) perfectly circular, and that the turntable as a whole does not wobble.

Even if all these conditions are satisfied, however, slight speed-fluctuations will nevertheless occur. The result is that the pitch of the music reproduced likewise fluctuates. When the pitch fluctuations are relatively slow (frequency 1 to 3 per second) the effect, if audible, is referred to as „wow". On the other hand high-speed pitch fluctuations produce what is known as „flutter" in the music. Tests have shown that wow and flutter resulting from speed fluctuations smaller than 0.3% are very difficult to detect. It is also found that these effects are audible mainly during the reproduction of relatively low tones and, other than by exception, only when the particular note is fairly long-drawn-out. Slow piano-music is probably affected the most; lovers of quick, rhythmic music, on the other hand, are not likely to be disturbed by wow.

Although the above-mentioned limit, 0.3%, is fairly stringent, it has the advantage, as borne out by many discerning listeners, that the effect of such speed variations usually is not noticeable enough to be annoying even in the reproduction of critical music.

As it is at low speeds that the weight of the turntable does least to keep the speed constant (the value of $\frac{1}{2}$ m ω^2 is then low), audible wow is most difficult to prevent at $33^1/_3$ r.p.m. Nevertheless, on an average the 0.3% limit can be maintained even in mass-produced equipment, provided that the general design and method of manufacture are planned with this in view.

To satisfy more stringent requirements special measures are necessary; the turntables are then cast, turned on a precision lathe, and balanced. Moreover, the castings for such turntables must be quite free from blow-holes. Owing to these special precautions the price is very much higher than that of ordinary turntables.

Now, turntables are not necessarily perfect merely because they are heavy; the omission of any of the precautions which should be taken during the manufacture may result in unsatisfactory performance. It is worth mentioning in this connection that on an average the amount of wow and flutter in Philips High Fidelity equipment is about 0.2%.

Although, as already stated, the disturbing effect of wow and flutter depends upon the type of music, as also upon the capacity of the particular listener to detect them, the following figures will nevertheless be useful as a general guide.

	Just perceptible	Intolerable
Slow piano-music	0,2%	1,0%
Violin music	0,5%	1,5%
Symphony orchestra	0,5%	1,5%
Dance music	0,5%	1,5%
Dance music with piano	0,5%	1,2%
Jazz	1,0%	2,0%

These are not arbitrary values; a slightly greater amount of wow is permissible in piano-music with a faster tempo, whereas a slow waltz occurring in the violin parts, or a horn solo in a symphonic composition, may impose above-average requirements.

The writer has noticed that the playback machine often gets the blame for wow really caused by something else. As a case in point, it may happen that the record does not lie at the exact centre of the turntable, say, owing to the hole in the record being too big, to eccentricity of this hole or of the record grooves, or to the spindle of the turntable being off-centre; this also causes wow.

This may be demonstrated with the aid of very simple mathematics. If the distance from a certain groove to the centre of the record be r cm and the speed of the turntable N r.p.m. the groove velocity will be 2π rN cm/min. Supposing that the centre of rotation, instead of coinciding with the centre of the record, is d cm away from it, then the groove velocity at a given moment will be $2\ \pi\ (r + d)N$ cm/sec, whereas a moment later it will be $2\ \pi\ (r - d)N$ cm/sec. The speed therefore varies 2π dN cm/sec on either side of the correct value, so that the record rotates first too fast, then too slowly, and so on. Expressed as a percentage, then the speed fluctuation is:

$$\frac{2\pi\, dN}{2\pi\, rN} \times 100\% = \frac{d}{r} \times 100\%.$$

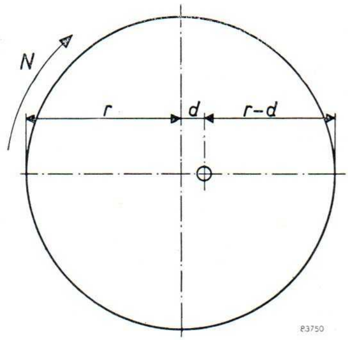


Fig. 48. Explanation of fluctuations owing to eccentricity

The wow caused by eccentricity is therefore the same at all records speeds, and it is worst when r is small, that is, at the innermost grooves of the record. Assuming that r is here 5 cm (2 inches) an eccentricity of only 3/10 mm (0.012 inch) would cause 0.6% wow. The wow at the first groove of a 12-inch record is then 0.2%. It is evident, then, that the tolerance on eccentricity is very narrow; nevertheless it does not often happen in practice that the eccentricity of the record hole falls outside this tolerance. Groove eccentricity is likewise rare, but at the same time there is very little unity as regards hole diameters.

The turntable spindles of studio playback machines are usually slightly thicker than those of domestic models. Some record manufacturers base the diameter of the centre hole on the studio spindle-size, therefore it may happen that the hole is slightly too large for the spindle of an ordinary record player or changer and the record does not lie exactly at the centre of the turntable. Most record firms take account of the smaller spindle-size, however, and their records do not fit well, if at all, on studio equipment. Wow from a record with too large a hole can be remedied by wrapping thin paper round the spindle before lowering the record on to the turntable. This is not practicable, of course, in the case of a record changer with a long changer spindle; all that can then be done is to try patiently to centre the record exactly.

In the same way, warped records and wobbling turntables may also cause wow.

Given a fairly good record player, the reproduction of a record whose hole is slightly on the large side may be good or bad, depending on the extent to which the fluctuations of the turntable-speed and those caused by the eccentricity either cancel each other out, or aggravate each other.

Measuring the speed-variations of the turntable is no simple matter and requires special apparatus. The stroboscopic discs used to determine the actual speed of the turntable cannot be used for this purpose, since any wobbling of the bars on these discs is almost always attributable to slight inaccuracies, say, owing to stretching of the paper on the discs, and has nothing whatever to do with the wow. In any case, tests with very accurate, metal stroboscope- discs have shown that only very serious wow can be demonstrated in this way, and that the human ear is a very much more reliable guide.

Some aspects of the turntable covering have already been considered in connection with the care of gramophone records. A good fabric covering is best, because the fibres shut off hard dust-particles; moreover. such coverings are more resilient than rubber, unless it foam rubber, and this requires a covering of its own to prevent it from becoming dirty too soon. Plastic materials have the same disadvantages as rubber; moreover, such materials must be chosen very carefully to ensure that the plasticiser giving the particular substance its mechanical properties does not react with the plasticiser in the record.

The most widely used covering consists of fibres of felt, rayon or nylon. This is applied by spraying the turntable with glue and then blowing the fibres on to the glue. There is a better method, however, which is to "draw" the fibres on to the turntable by means of an electrostatic field (voltage 25.000 — 50.000 V). In this way all the fibres are drawn on to the turntable at right angles, so that they penetrate further into the glue and also provide a more resilient covering. The covering can then be treated in a special way to make the fibres adhere even more firmly to the turntable.

Section 3. The tone arm

Since we have already seen something about the tone arm on pages 35 and 40 a word or two on the subject will now be enough. The tone arm bearing must be so designed that the needle can move unhampered towards the centre of the record, without overshooting the groove at any point: this is important also from the point of view of needle wear.

With a properly designed tone arm bearing it is possible to obtain 10 to 20 g cm of friction torque, consistent with less than 1 gram (0.04 oz.) of lateral pressure on the needle. Since nearly all records are slightly warped and undue stiffness in the vertical movement of the tone arm therefore causes them to wear more quickly, the arm should also move lightly up and down. On the other hand, too much play in this tone arm bearing affects the quality of reproduction.

Another important point is what is known as tone arm resonance. The tone arm, also, has a certain natural frequency at which it vibrates most readily. If the particular record happens to contain a tone at this frequency, the needle will transmit vibrations at the same frequency to the tone arm. Pickups operate on the principle that the needle moves in relation to some voltage generating device clamped in the tone arm. At the same time, any vibration of the tone arm itself, and therefore of the voltage generating device, causes the amount of needle movement relative to this device to be either increased or reduced, depending upon the mode of vibration. In the one case the particular tone to which the tone arm is sensitive is amplified, and in the other case it is attenuated, in the reproduction. However, the natural frequency of properly designed tone arms is low enough to ensure that it

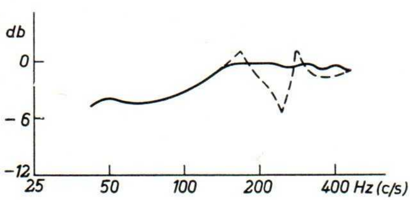


Fig. 49
Effect of tone arm on bass response

does not occur, or at any rate occurs only very rarely, in recordings; also, special measures are taken to damp the natural resonance of the tone arm as much as possible. This is done to a large extent through the tone arm bearing. Part of the response characteristic of a particular pickup mounted first on one tone arm and then on another is shown in fig. 49, from which it is seen that such characteristics are not very much use unless particulars of the tone arm are also given.

Section 4. The switch mechanism

Almost every modern gramophone contains a trip of some kind to actuate a switch or a changer mechanism after the playing of each record. In principle there are three different systems.

a) The distance switch

This operates when the tone arm reaches a certain position. Since the diameter of the last music groove is not the same for all records — it varies considerably as between $33^1/_3$, 45 and 78 r.p.m. records and to a smaller extent also as between individual records of the same type — such switches have to be readjusted every time. Although it is possible to simplify matters by coupling the switch and speed controls and ignoring the smaller differences between individual records, this is not very satisfactory; hence distance switches are now being used less and less.

b) The eccentric-groove switch

On many modern records the final groove is a closed, eccentric circle. When the needle enters such a groove, the tone arm swings to and fro.

In some cases the outward swing of the tone arm is used as a means of operating a switch, on the principle which will now be explained. An auxiliary arm carrying a pawl is attached to the tone arm. As the tone arm swings inwards the auxiliary arm swings with it and the pawl slides past the switch. When the arms swing outwards, the pawl engages with the lever of the switch, thus turning it to the "on" position. In practice the system is more complex and includes one or two other levers, amongst other things to enable the gramophone to be switched on again by swinging the arm further outwards. Changer mechanisms can be operated in the same way. The disadvantage of this method of switching is, of course, that is does not work on records without an eccentric groove. An analysis of the international record market has shown that although many records are provided with such grooves, records with different stopping grooves, including the 45 r.p.m. records, are also quite common. One advantage of this switch system over that which will be described in the nexth paragraph is, that although it is not specially designed for the purpose it enables the pickup to be lowered on to any part of the record by hand without difficulty.

c) The velocity-trip switch

On records with eccentric final grooves, as also those with spiral run-out grooves (that is final grooves in the form of a concentric circle), the inward swing of the pickup accelerates as soon as the needle passes from the last music groove to the stopping groove.

This acceleration can also be used to operate a switch. Fig. 50 illustrates the

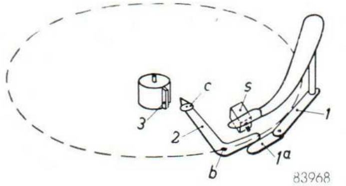


Fig. 50. Principle of the velocity trip-switch

principle of such systems. Here we have an auxiliary arm (1) attached to the vertical spindle of the tone arm, and a plate (1a) pivoting with a certain amount of fricton at the free end of the auxiliary arm. When the tone arms swings towards the centre of the gramophone record, 1 and 1a move with it, and at a given moment plate la engages with a tripping hook (2) pivoted at point b. Hook 2 turns very readily and is therefore pushed, in anti-clockwise direction, by the plate. Since the tone arm traverses the record very slowly, hook 2 also rotates slowly, but after a time the hook swings so far round that a roller (C) attached to it collides with a lug (3) on either the hub of the turnable or the turntable itself, which pushes the hook back a little way.

Because plate la pivots on the auxiliary arm, it does not block the hook; as the tone arm swings further inward, however, the hook turns again and again in anti-clockwise direction, only be knocked back every time by the lug. At this stage the hook does not travel more than a few thousandths of an inch. When once the needle reaches the run-out groove, however, the tone arm carries the hook along very much faster (roughly a quarter of an inch per revolution of the turntable). Instead of striking roller C, lug 3 then hits the tip of the tripping hook itself, thus swinging it round to operate the switch (S) or the changer mechanism. This method of switching works with various records irrespective of the type of run-out groove; the only disadvantage is that it also operates when the pickup is moved towards the centre of the record by hand, which would make it difficult to select a given passage of the recording. Provided that the different components are suitably dimensioned, however, the tripping mechanism does not function until the pickup is only about 2½ inches from the middle of the record; since the run-out groove is nearly always less than 2½" from the middle of the record, the needle can be lowered into any of the music grooves without too much difficulty and without making the tripping mechanism any the less reliable.

Section 5. The record player

The preceding sections of the present chapter have been devoted as far as possible to general principles. In practice, however, there are often considerable differences between one piece of equipment and another. As it is not within the scope of this book to discuss such differences at all fully, we must be content with a single example.

Fig. 51 shows a record player type AG 2004. The motor, of the type already described, has a vertical shaft with a „stepped" roller at the upper end. A lever is provided to raise and lower the idler wheel and lock it in any of 4 positions. In the bottom position this wheel is clear of the roller and the rim of the turntable, and in the next higher position it rides against the thickest part of the roller to drive the turntable at 78 r.p.m. The motor with the idler wheel, and the lever with its socket are secured to a spring-mounted metal plate, to ensure that the vibration of the motor is not transmitted through the „Philite" mounting plate to the turntable and the tone arm.

The turntable has what is known as a gyroscope bearing, that is, a steel ball resting on the end of a pin, fixed to the mounting plate, which fits into the hollow boss of the turntable; such bearings are very accurate and smooth-running. The lug to operate the tripping mechanism is on the bottom face of the turntable, and the trip slide is pushed towards this lug by a lever attached to the pivot of the tone arm. As long as the needle runs in the music grooves, the slide moves very slowly and is pushed back every time the lug strikes the plastic buffer. When the needle enters the run-out groove, however, the slide moves more quickly and reaches the tip of the buffer before the lug has time to reach it; the slide is then shifted far enough to trip the switch. The lever also trips the mains switch on again when the tone arm is swung outwards, and at the same time operates a pickup switch to short the pickup when the motor is not running. In this way unwanted sound, e.g. the hiss of the needle in the run-out groove, is suppressed.

When this record player is built into a radiogram or other equipment incorporating a speaker, it is mounted on springs to ensure that the vibrations of the speaker do not reach the pickup. The latter would re-convert such vibrations into voltages, which would be amplified, fed to the speaker, and so on in a vicious circle. The effect of such acoustic feedback is that the gramophone oscillates and the speaker produces a howl whose pitch depends upon, amongst other things, the acoustic properties of the cabinet.

Of course, this happens only when the amplification goes beyond a certain limit; with the volume control turned down, the vibration from the speaker is too weak to sustain a prolonged howl, although strong enough to cause a certain amount of distortion on certain notes.

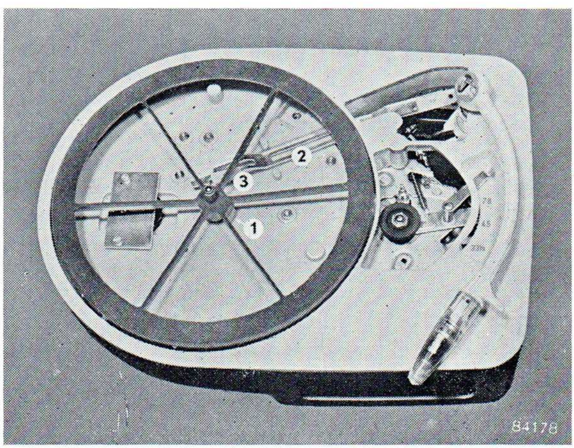


Fig. 51. Record-player AG 2004 (section)

Whereas in some cases microphony occurs only in response to a particular tone in the music, in other cases noise from the amplifier, even if it be slight, is enough to start this effect. Mounting the record player on springs ensures that the sound vibrations do not reach the pickup, provided, of course, that the springs are flexible enough.

The problem of microphony is not confined to record players; all playback equipment housed in the same cabinet as a speaker should be spring-mounted. Moreover, microphony may also occur when the speaker and the record player or changer are not in the same cabinet, say, if the loudspeaker is standing on fairly slack floor-boards and the playback equipment on a rather flimsy table. Again, microphony may occur in any player or changer close to a radio receiver on the same table or sideboard.

At worst, it may cause the gramophone needle to overshoot the groove: this can be very annoying, especially when the latest records are being played to guests and the amplifier, without it being noticed, is allowed to play more loudly than usual. If groove jumping takes place under these conditions, although it has not happened before, it is advisable to test for acoustic feedback as a possible cause by turning down the volume slightly.

Section 6. Record changers

With record changers, also, there are considerable differences between different models. There is always more than one way to reach a goal, and although it would be reasonable to suppose from the number of patents now filed that every possibility has already been explored, experience shows that there is still plenty of scope for development. Record changers differ from record players in that they include a mechanism to swing the tone arm outwards after the playing of the record and then guide the needle back to the starting groove of the next record, and another to hold a stack of records and release them one at a time from the bottom of the stack so that they drop on the turntable at exactly the right moment.

Record changers differ in many particulars, which will now be discussed.

a. Record-size selection. The tone arm must be governed in such a way that it stops at the precise moment when the needle is over the first groove and ready to be lowered on to the record. This is simple, of course, if the diameters of all the records are the same, e.g. 7 inches. Since nearly all record changers are designed to operate with records of different diameters, however, it is necessary to provide some means of adjusting the tone arm to suit the particular record. Some tone arms are adjusted by means of a knob (manual operation) and others by a feeler responding to the record itself. Systems which adjust the tone arm automatically to the record size fall into two categories, namely those in which any two records played one after the other must be the same size and those which also enable records of different sizes to be played at random one after another. One advantage of what is known as automatic intermixed playing is that it simplifies the operation of the changer enough to make it fool-proof.

b. Reject. The reject knob actuates the changer mechanism, so that the playing of the particular record is interrupted and another record is lowered on to the turntable and played instead. In most changers the reject knob is also the starting knob.

c. **Repeat.** When the repeat knob is operated the particular record being played is repeated. Here we have two possibilities, namely that the pickup returns to the start of the record as soon as the button is pressed, or that it does so only after the record has been played.

d. **Interval.** Some changers are provided with an interval knob to introduce a certain interval, usually 1 to 5 minutes, between the playing of two consecutive records.

e. **Stopping.** Many changers stop automatically after playing the last record. Some models repeat the last record until switched off by the listener, whereas in others the tone arm stops in the neutral position, although here the listener must operate the mains switch.

In certain record changers a switch controlled by a special knob stops the changer automatically as soon as it has played the particular record on the turntable, even if this record is not the last. With other models the same effect is obtained by guiding the tone arm to the stopping pawl by hand.

f. **Changing cycle.** The time required to lift the pickup from the record, drop another record on to the turntable and lower the pickup on to this record is roughly 4 seconds for 78 r.p.m., 7 seconds for 45 r.p.m., and 10 seconds for $33^1/_3$ r.p.m. records. Faster operation is undesirable from the point of view of reliability, and slower, although it happens in some cases, is in principle quite unnecessary. In some systems the changing time is the same for all three turntable speeds — usually about 5 to 7 seconds.

The most important characteristic of any record changer is its reliability. Changers which tend to become temperamental when it is least expected are a source of continual annoyance. At the same time it is impossible to define any general rules for assessing the reliability of record changers. Given such rules, in fact, the manufacture of good record changers would not present much difficulty; not only the actual design of the changers is important, but also the care taken in manufacturing and dispatching them, and the condition of the particular factory plant. Transport tests, shock tests and life tests can provide manufacturers with much information, which, properly applied, will ensure that their products continue to function unerringly for many years. It will be evident, then, that the question „record player or record changer" need not be considered from the point of view of reliability. As regards sound reproduction, also, there is no essential difference between properly designed record changers and record players. On the other hand the changer offers more comfortable and undisturbed listening, especially in the playing of 78 and 45 r.p.m. records. Another advantage of the changer, which also holds good for long-playing records, is that the mechanism lowers and raises the pickup more gently than is possible by hand, so that for all practical purposes there is no risk whatever that the record will be scratched. Although the same advantage is of course obtained with fully automatic record players, the price of record players with automatically raised and lowered pickups is very much the same as that of

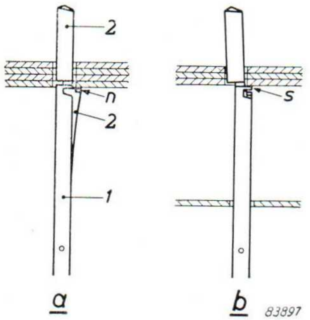


Fig. 52. Changer spindle for records with small hole

a changer. The question is sometimes raised, whether a changer is not likely to break the records or at any rate damage the centre hole. With a properly designed record changer, however, this risk is extremely small; during laboratory tests dozens of records were „changed" more than a thousand times before suffering any damage, although, of course, such tests resulted in a considerable amount of groove wear. In our experience records remain practically unworn in normal use, any serious loss of reproduction quality being far more likely to be caused by scratches, amongst other things owing to careless or clumsy lowering of the pickup, or by dust. With a changer, the risk of scratching, at any rate, is almost entirely eliminated. The problem of designing a changer so that it will hold a stack of gramophone records and release them one by one would be relatively simple if all records were the same thickness, but as it happens they are not. In many of the older types of records changers, the record pile rested on a number of blades, protruding from "posts". To release a record these blades were retracted and other blades, slightly higher than the first-mentioned ones, were fed forward. These higher blades were inserted between the bottom record and the one above it to support the rest of the pile at the moment when the blades supporting the bottom record were withdrawn.

A moment later, when the blades were rotated back to their original positions, the stack of records again rested on the lower blades. One disadvantage of this method is that such changers cannot be made easily adjustable to different record diameters, let alone suitable for automatic intermixed playing.

Another method, depending less upon the thickness, but more upon the tolerances on the record diameter, is the one involving a turntable-spindle with a bend in it. The records are supported at the centre by the bend in the spindle and at the periphery by a special ledge, so that in effect they lie at a slant. When the changer is operated, a blade presses down upon the bottom record, forcing it past the bend in the spindle so that it drops on to the turntable. Although this method is simpler from the point of view of adjustment to different record diameters, it does not solve the problem of intermixed playing.

With the most up-to-date models, the records rest upon a lug on the spindle and are usually held level by what is known as a record loading disc. How the spindle of the Philips changer operates will be seen from fig. 52. This spindle comprises a fixed part (1) and a moving part (2). In the neutral position (fig. 52a) the records rest upon the expanded tip, or nose (n), of the moving part. To change the record, part 2 of the spindle moves to the left; the bottom record of the pile is prevented from following this movement by the shoulder (s) of spindle 1 (fig. 52b). The tip of the moving spindle being fully retracted, the bottom record is deprived of its

support and slides down the spindle on to the turntable. An air cushion beneath the (horizontal) record prevents it from falling too heavily. The other records in the stack are carried along by the upper part of the spindle until they rest upon shoulder (s). At the end of the changing cycle, when the moving part of the spindle returns to the rest position, it pushes the pile of records off shoulder s and back on to nose n, thus restoring the situation shown in fig. 52a. The movements of the spindle are governed by the changer mechanism.
Changer spindles of this type must be withdrawn from the turntable to enable the records to be removed after being played. In other changers the moving part swings aside when the records are lifted up the spindle; hence the latter need not be wihdrawn. On the other hand, the records must then be lifted quite level in order to avoid damaging the centre holes. With all changer spindles the thickness of the records is very important. The limits of tolerance on record thickness suggested for international standardization are 1.6 and 2.3 mm (0.064 and 0.092 inch). With two relatively thin records, say both 1 mm thick, there is a risk that the changer spindle will release both of them at the same time, whereas records which are too thick are apt to „hang" on the spindle. Therefore it is important that the tolerances on record thicknes be adhered to. The very nature of the pressing process necessitates that these tolerances are rather wide; the smaller firms manufacturing shellac records are prone to make them too thick, and some of the cheaper long-playing records on the market are too thin. Any records suspected of being outside the limits of tolerance should be checked with a gauge. In the case of records with a large hole (1½ inches, the 45 r.p.m. records) the problem of tolerances is less acute thanks to the special form of such records. The special changer spindles for records of this type (fig. 53) are provided with two sets of blades one above the other to support and release the records in the manner described on page 76.
Another method is to employ a system of "buffers" which emerge from the pin to clamp the records — this method is less suitable for records with a small hole, owing to the amount of force involved as also the tolerances on the centre of the hole.

Section 7. The changer mechanism

Changer mechanisms in general are so many and varied that it is altogether beyond the scope of this book to describe them all. We shall therefore consider only what are, in our opinion, the more noteworthy models. Fig. 54 is a phantom view of the Philips changer type AG 1000. The turntable (3), here shown in faint outline, is driven through an intermediate wheel (2) by the motor (1). Note the system of levers to shift the wheel. The hub of the turntable (4) is cogged to drive the control disc (6). In the neutral position these cogs rotate freely in a slot at (5) in the edge of the control disc.

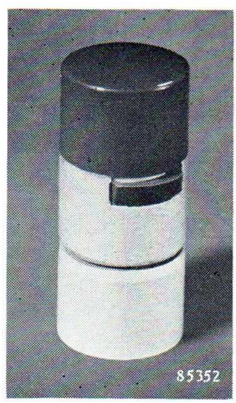


Fig. 53. Changer spindle for records with large hole

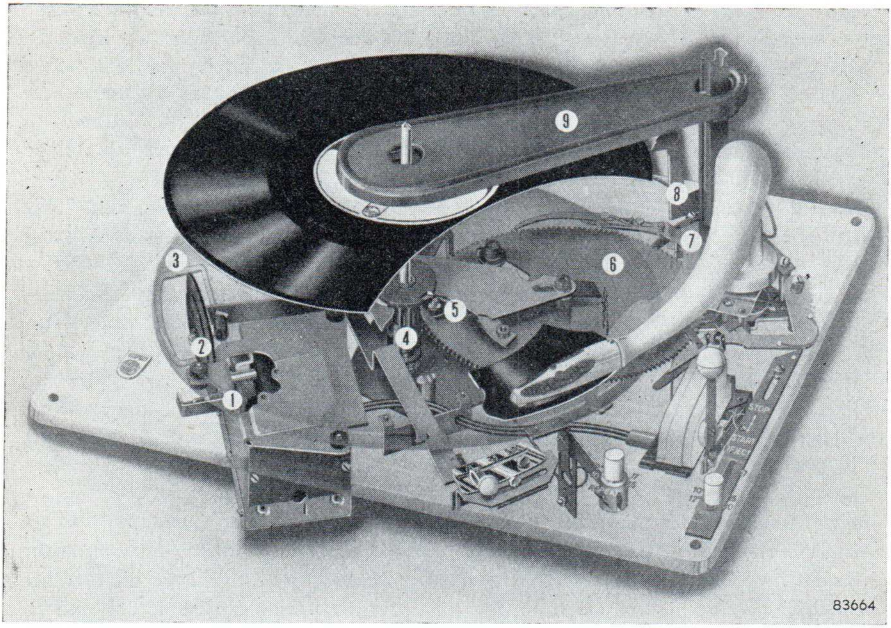


Fig. 54. Record-player AG 1000

On receiving a slight push, however, either from the impulse hook coming into operation after the playing of each record (see also fig. 50), or through the movement of a bracket connected to the starting handle, the control disc makes one complete turn. In so doing it actuates a pin, passing through the vertical tone arm bearing, which pushes the tone arm up. At the same time a stop (6a) presses against the right-hand part of the control hook (7), pushing it and the tone arm, to which the hook is connected, outwards. Next, the control disc actuates the moving part of the changer spindle, and a moment later stop 6a presses against the left-hand part of the control hook, thus guiding the pickup back to, and lowering it on to, the record. The different components are so designed that with a 10-inch record the needle is automatically lowered into the starting groove. When a 12-inch record drops on to the turntable it slides along a feeler (8), which recoils slightly, thus raising a lever at the bottom of the feeler to block the tone arm at the precise moment when it reaches the starting point for the playing of 12-inch records. Since part of the left-hand section of the control hook is resilient, this blocking system operates without damaging the mechanism. For the playing of 7-inch records a special hook is provided, to be moved into position in front of the ordinary control hook by means of a knob and slide.

The changer is stopped by preventing the tone arm from swinging inwards again

when once it has reached its outermost position. The tone arm then descends, not upon the record, but upon a switch which stops the motor. This blocking system can be actuated either by the loading disc (9) descending after the playing of the last record, or by means of the operating handle.
Pressing the repeat button disengages the changer spindle from the mechanism, thus preventing the release of the next record. A lug on the control disc pushes the repeat button back to the neutral position as soon as the needle is lowered on to the record.
Record changer AG 1004 also operates with a control disc, but the needle of this changer is lowered into the starting groove in a different manner, as will now be explained (see fig. 54a). When a 10-inch record drops on to the turntable it touches a hook (1), which is thus pressed down slightly to remain fixed in a certain position. A projection on the tone arm then engages with this hook and so prevents the arm from swinging inwards. The hook is so dimensioned that when the tone arm descends the needle drops exactly into the starting groove of the 10-inch record.
When a 12-inch record is released, the hook is pressed further down and fixed in a lower position. Hence the inward swing of the tone arm during the second phase of the changing cycle stops even sooner, that is, just above the starting point for 12-inch records. Only 7-inch records clear the hook completely when dropping on to the turntable, thus enabling the tone arm to swing freely inwards until the control hook — beneath the mounting plate — is blocked and the needle descends at the starting point for 7-inch records.
The mechanism of record changer type AG 1003, shown in fig. 55, operates on entirely different principles.
In this changer, which can also be employed as an automatic record player, the movements of the tone arm are governed by a control drum (1 in fig. 55; partly visible). This drum makes one complete turn during the full changing cycle, and in so doing lifts a bracket provided with a pin to raise the tone arm from the gramophone record.
At the same time it moves a lever, whose one end rests in a slot in the drum and whose other end is attached to the vertical pivot of the tone arm. This causes the tone arm firstly to swing outwards from the record, and then to swing inwards again to the starting groove of the next record as soon as this drops on to the turntable. The control drum is driven, through spindle (2), by a worm wheel engaging with a worm attached to the hub of the turnable. A gap made by remov-

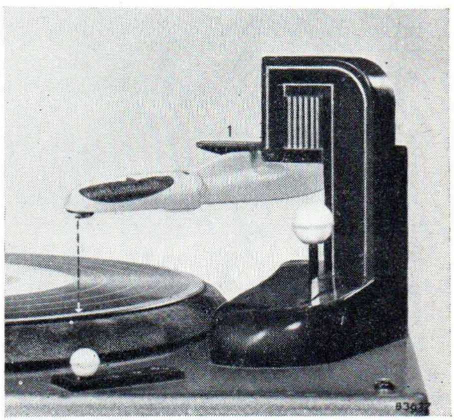


Fig. 54a. Indexing mechanism of record-changer AG 1004

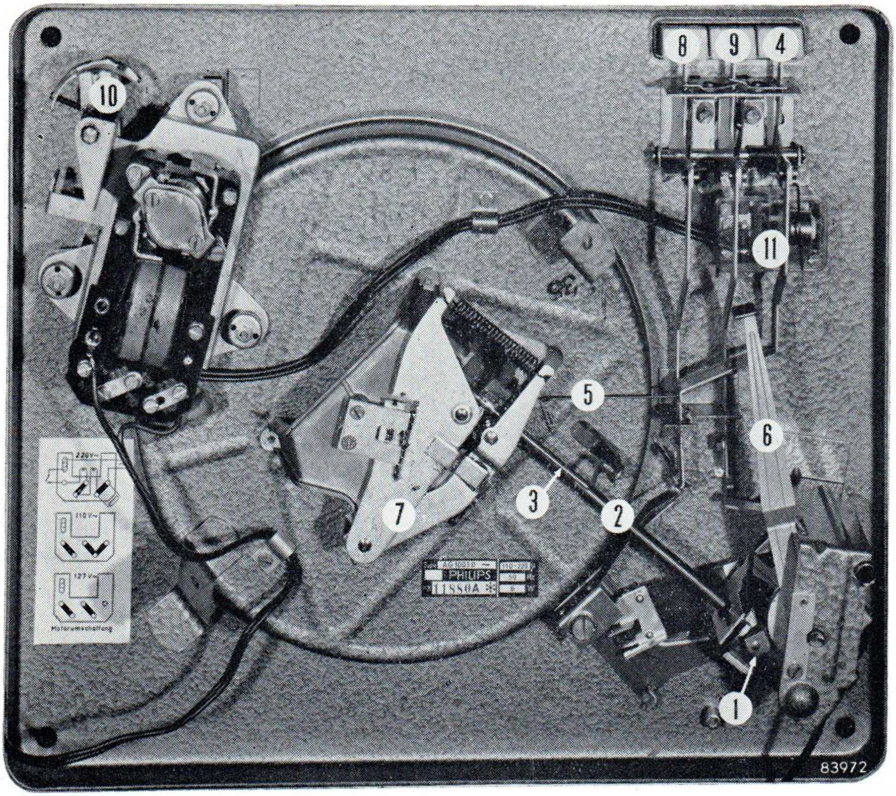


Fig. 55. Changer mechanism of record-changer AG 1003

ing two teeth from the worm wheel enables the worm to rotate freely during the playing of the record. This gap can be closed by means of a special coupling plate provided with two teeth, which is moved into position in front of the gap at the beginning of each changing cycle to couple the worm wheel to the worm; when the wheel has completed a full turn, the coupling plate drops away from the gap, thus disconnecting the worm drive and stopping the rotation of spindle 2. The coupling plate is controlled by rod (3), as will now be explained. When starter button (4) is pressed, hook (5) moves forward to press against rod (3), thus moving the coupling plate into position to cover the gap in the worm wheel. At the end of the particular record, the impulse hook (6) connected to the tone arm touches rod 3. As long as the tone arm is moving gradually inwards, rod 3 is continually repulsed by a ridge on the turntable. When once the needle reaches the run-out groove of the record, however, the movement of impulse hook 6 is accelerated and rod (3) is then pushed far enough to bring the coupling plate over the gap in the worm wheel. During the changing cycle a pinion on spindle 2 moves the two blades of the scissor-shaped support (7) of the changer spindle, thus releasing the bottom record of the stack so that it slides down on to the turntable (see also fig. 52).

As it slides down the spindle, the record touches a feeler projecting from what is known as the control column, and this feeler "signals" the diameter of the record to a bracket which stops the inward swing of the tone arm as soon as the needle is over the starting groove of the record. In this way, the pickup is always guided automatically to the correct starting point for the particular record, regardless of whether this is a 7, 10 or 12-inch model.

The tone arm can also adjusted by hand by pressing button 8, the adjustment then depending on how far down the button is pressed. This is because automatic changing, although very useful for records which play for only 3 to 9 minutes, is less so for long playing records, which in many cases provide quite enough music for one sitting. Although the changer lowers the pickup on to the record more gently than by hand, it is going the long way round to use this mechanism also to place the record on the turntable. With this record-playing equipment, however, the record can be placed on the turntable by hand, and the pickup then lowered on to it automatically; hence the equipment gives sufficient scope without any risk of damage to the records. The changer also lifts the pickup from the record automatically as soon as the music ceases. Again, of course, the pickup can be lifted or lowered by hand, so that it is also possible to play only certain parts of records. Changer AG 1003 therefore combines the most convenient features of automatic and ordinary record players.

The changer can be switched off at any moment by pressing push-button 9. Lever 10 shifts the intermediate wheel to change the speed of the turntable; the motor switch is seen at 11. When the tone arm returns to its support it presses down a pin to operate the motor switch; the spring seen on the right (11) turns the switch on when the starter button (4) is pressed.

CHAPTER VIII
AMPLIFIERS

Section 1. Power and distortion.

The output of the pickup is far too low to drive the loudspeaker and must therefore be amplified. This is done by means of amplifiers equipped with valves or transistors. For information as to the working of these components, we refer to the special literature on the subject. It is already evident that the transistor is destined to play an important part in the future development of the gramophone. Its most notable advantages are that it does not require a heater voltage and that it can be operated with only a low d.c. voltage, e.g. 6 to 12 volts. In the opinion of the writer there can be no doubt that portable gramophones with clockwork motors and heavy acoustic reproducers will be superseded in the very near future by electric gramophones equipped with transistor-amplifiers and powered by flash-lamp batteries, which will be suitable for long-playing, as well as for standard, records. Such gramophones will keep record-wear to a minimum and provide high-quality reproduction even in places without mains-electricity. Transistors offer obvious advantages for use in pre-amplifiers, because the required D.C. voltage is low and because the possibility of amplifier hum is very much smaller owing to the fact that there is no filament. Transistors suitable for use in amplifier output stages, where the output must be high to ensure good-quality reproduction, are also obtainable.

All amplifiers contain the following divisions:

An **output stage** to generate voltages and currents strong enough to drive a speaker. The output transformer, matching the impedance of the speaker to that of the output valve, acts in very much the same way as the transmission of a car; it ensures that energy is transferred as efficiently as possible from the output valve to the speaker. Cars would not move very fast if their wheels were coupled direct to the rapidly rotating shaft of the motor, nor can they travel very fast if the gear-ratio is too low.

A **pre-amplifier,** to boost the output voltage of the pickup enough to drive the output stage.

A **tone-control,** to correct imperfections in the tone balance.

A **volume-control,** or loudness-control, to vary the sound level.

A feedback circuit, to reduce non-linear, and in some cases also linear, distortion.

A **supply unit,** to provide the necessary a.c. and d.c. voltages for the two amplifying stages.

Usually these are not separate units, but are all mounted on one chassis, although in some cases part of the pre-amplifier, with the tone and volume controls, is designed as an individual unit. This is then referred to as the pre-amplifier, the remainder of the circuit being described as the output amplifier.

All the above-mentioned parts are important from the point of view of the quality of reproduction; imperfection of any one of them affects the performance as a whole. Gramophone amplifiers have to satisfy many requirements; the two best-known features of such amplifiers are the frequency characteristic and the output power, which are closely inter-related. As the largest vibration amplitudes occur at the lowest frequencies, the amplifier also develops full power at such frequencies. To be suitable for reproducing the very lowest frequencies, amplifiers must not only have a frequency characteristic which is reasonably straight at such frequencies, but must also be capable of supplying enough undistorted power to drive the speaker. As a case in point, an amplifier with a maximum output of 3 Watts is quite powerful enough for a large room, provided that the bass response of this amplifier begins at 80 c/s; with bass reproduction from 30 c/s, however, the output must be at least 10 watts. Supposing that the response range of the 3-watt amplifier also continues as far as 30 c/s, and that both amplifiers are adjusted to the same volume during a passage containing no bass, the difference will become noticeable only when very low notes occur in the music; then, however, it is usually very distinct. This explains the fact that amplifiers which are entirely free from distortion with a particular pickup, suddenly emit audible distortion when operated with another pickup having better bass response. Although this is well-known, it often happens that the better pickup is blamed for the distortion really owing to the limitations of the amplifier. When choosing a good-quality amplifier, then we should make sure that it is not too small; although the above figures are only approximate and depend also upon the efficiency of the particular speaker, they will be useful as a guide.

Amplifiers are often rated according to the power which they are able to deliver subject to, say, 5% distortion as measured at 1000 c/s. Such specifications are not really adequate, however, since it is quite possible that at higher frequencies the threshold of distortion will occur at a very much lower level, perhaps because negative feedback is not effective at high frequencies, or owing to a loss of sensitivity in the amplifier, or because it becomes necessary to increase the input voltage of the amplifier to such an extent as to cause distortion in the initial stage. If the output transformer is too small it reduces the amount of power that can be delivered without distortion, the more so on low notes. Fig. 56a shows power characteristics consistent with 5% distortion for a good amplifier and a bad one, both with the same output power at 1000 c/s; fig. 56b shows similar characteristics plotted in a different way, the lines in this diagram indicating the amount of distortion at different power levels and frequencies. Here the difference will be seen quite clearly.

As we have already seen in connection with pickups, intermodulation distortion is a more reliable measure of the properties of the particular reproducing equipment than distortion as measured with the aid of a single tone. It is usual to measure intermodulation distortion by applying two voltages to the amplifier at the same time, the one at a high, and the other at a low frequency, and the one with four times as much amplitude as the other.

Voltages at 100 and 4000 c/s are usually employed, although for High Fidelity amplifiers the frequencies are often taken further apart, say, 25 and 20,000 c/s, both of which occur in the most critical ranges and therefore reflect the characteristics

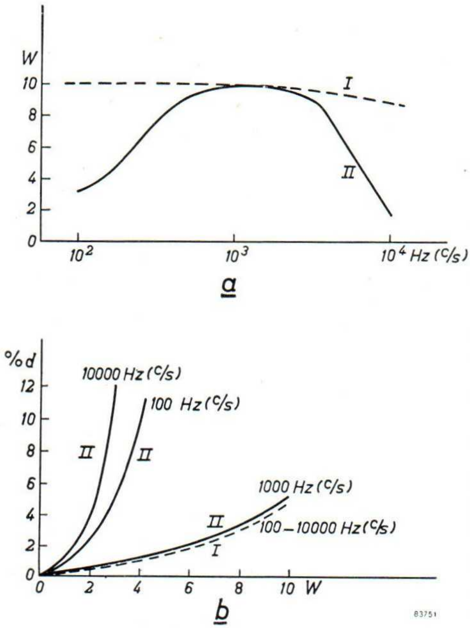


Fig. 56. Power and distortion of two different amplifiers

of the amplifier more clearly. Intermodulation distortion is measured by determining the strength of the sum and difference tones resulting from such distortion, with a circuit very much the same as the detector circuit in a radio receiver. The ratio of the distortion tones to the highest test-tone is known as the intermodulation figure. Fig. 57a shows the two test signals separately, and fig. 57b the sum of them as applied to the amplifier. Fig. 57c is the distorted signal at the output of the amplifier.

The tone of the lowest frequency is filtered out of this signal to leave the sound vibration shown in fig. 57d. The actual distortion will be seen quite clearly when figures 57a and 57d are compared – in ordinary radio-phraseology, the treble is modulated by the bass to the accompaniment of frequency doubling. In effect, intermodulation distortion denotes the depth of modulation of the distortion occurring in the signal at the highest frequency; therefore:

$$\text{I.M.} = \frac{A - B}{A + B} \times 100\%$$

where A is twice the largest, and B twice the smallest, of the amplitudes shown in d. It is usual to specify the intermodulation curves of amplifiers, i.e. the relationship between intermodulation distortion and output power. At the same time, scientists are not entirely in agreement regarding this matter. One point of view is that the most serious distortion occurs at moments when the sum signal is strongest, and that the peak power should therefore be computed at such moments. The speaker voltage is then $V_l + V_h$ and the computed power

$$\left(\frac{V_l + V_h}{\sqrt{2}}\right)^2 : R_l,$$

where V_l and V_h are the bass and treble amplitudes and R_l is the impedance of the speaker.

As $V_l = 4\ V_h$, the peak power is therefore $12\tfrac{1}{2}\ V_h^2 : R_l$

Another point of view is that the average output power should be computed, that is:

$$\left(\frac{V_l}{\sqrt{2}}\right)^2 + \left(\frac{V_h}{\sqrt{2}}\right)^2 : R_l.$$

Substituting $V_1 = 4\ V_h$, we find that the average power is: $8\frac{1}{2}\ V_h^2 : R_l$.

The usual course is to specify intermodulation as a function of peak power.

Fig. 58 shows the intermodulation curve (full line) of a High Fidelity amplifier type AG 9000. One part of this curve, almost horizontal, merges fairly abruptly into another, almost vertical, part. It is generally accepted that intermodulation distortion becomes noticeable only when it goes beyond 2%; in High Fidelity amplifiers, therefore, this point is taken to the highest possible power level, regardless of the fact that a very marked increase in distortion then takes place beyond it. With ordinary amplifiers the first part of the characteristic is less straight and the second part less steep (dotted line (b) in fig. 58).

As a general rule intermodulation distortion in good amplifiers, whose distortion curve is virtually straight (see fig. 56), is roughly 3—4 times the non-linear, single-tone distortion. This means that to ensure undistorted reproduction the single-tone distortion must not exceed ½%, which is very much less than the amount tolerated in conventional amplifiers (5 to 10%).

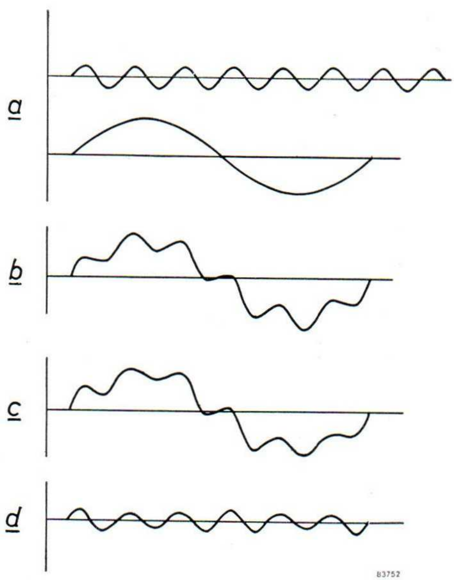


Fig. 57. Intermodulation distortion:

a) The two test-signals

b) The composite test-signal

c) The distorted output-voltage

d) The distorted high tone

Apart from intermodulation distortion there is also what is known as beat note distortion; the two are closely related, but with beat note distortion the frequencies are variable, the starting point being a tone varying continuously between, stay, 3000 and 20,000 c/s, and another of the same intensity, at all times 1000 c/s lower than the first. Therefore the beat note occurring as a result of distortion is invariably 1000 c/s; the sum tone varies from 5000 to 39,000 c/s. It is the former which is measured, and although this is done in very much the same way as the measurement of intermodulation, the two tests may produce entirely different results at a given frequency. Through this test, although it is not very widely

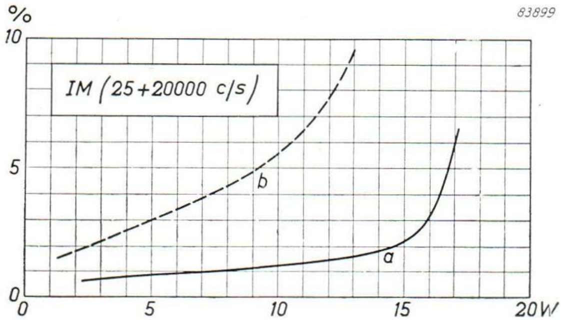


Fig. 58. Intermodulation curves:
a) amplifier type AG 9000 b) ordinary amplifier

used so far, certain defects not revealed by the intermodulation test (although perceptible enough to the ear) may become quite apparent.

Section 2.

Transients and ultrasonic vibrations.

The statement that the frequency characteristics of High Fidelity amplifiers should be straight even outside the audible range is not always accepted without query. This is understandable, since it is not altogether obvious that variations in response at frequencies high enough or low enough to be inaudible, and even beyond the range of music, nevertheless affect the reproduction. An abrupt falling-off of the response characteristic from, say 25000 c/s sharpens the tone of the reproduction slightly in very much the same way as any steep drop in response occurring at high frequencies within the range of hearing. The explanation of this effect, based on the phase rotations which take place within the fall-off range, is too complex to be discussed here; of course, the effect itself takes place only when the pickup responds to the highest frequencies.

The effect of this, although it may seem surprising at first glance, is that record noise is fainter and sounds less sharp with an amplifier whose response characteristic is more or less straight up to 50,000 c/s, than with one which falls off sharply, or exhibits a resonance peak in the response curve, at 20,000 c/s. Resonances at sub-audio frequencies are particularly troublesome in the bass. With a pickup which is sensitive to such frequencies, vibrations at frequencies too low to be audible, owing, say, to people walking about in the room or to a passing car, may be so enhanced as to overload the amplifier and therefore distort the music from time to time in a seemingly inexplicable manner. Moreover, the sudden starting or ending of a tone may cause the resonant circuit to oscillate at the resonant frequency, which also tends to overload the amplifier.

For High Fidelity reproduction, the characteristic should be straight to within 2 db from 20 to 20,000 c/s, and to within 5 db from 10 to 50,000 c/s. The whole range from 10 to 50,000 c/s must be kept free from noticeable resonances. Although this is by no means simple, it is nevertheless practicable. It is also necessary to ensure that impulse sounds are properly reproduced. The notes of many musical instruments start very abruptly and then die away gradually. This applies to various percussion instruments and also to the violin, since what is heard as a single violin note is really a whole train of separate sound impulses or transients.

There are several factors which affect the reproduction of such transients. If part of the amplifier has a certain resonance and therefore imparts extra boost to signals in a given frequency range, the response characteristic will have peaks at these frequencies. When transients enter such amplifiers, these not only reproduce the sound itself (fig. 59a), but also oscillate at the resonant frequency, thus mixing a disturbance with the desired signal (fig. 59b). The reason is that the resonant part of the amplifier is excited by the abrupt starting or ending of the particular impulse, and, like the pendulum of a clock, which goes on swinging for a long time after a single push, such resonant vibrations may last longer than the impulses which caused them. Resonances occur in circuits containing not only capacitances, but also self-inductances (coils). Bad wiring may cause an inductance which, in conjunction with the capacitance of, say, a decoupling capacitor, oscillates at audio-frequency; in many cases such oscillation is due to the unavoidable self-inductance of the decoupling capacitor itself.

An inductance of only 1 millihenry, combined with a capacitance of 0.5 μF, resonates at 7000 c/s, and values of this order of size are by no means uncommon. Parasitic capacitances in the output transformer in conjunction with the self-inductance of the transformer windings are another possible source of undesired resonances. In fact, one of the most difficult tasks associated with the development of an amplifier is to design a good output transformer.

Now, the signals considered so far are really only test signals; true impulse sounds are more like the one shown in fig. 59c. Although it would be reasonable to suppose from fig. 59b that the parasitic vibrations at the tail of the impulse are the most troublesome, as these are not masked by the desired vibrations, it will be clear that in the case of music the real culprits are the effects occurring at the beginning of the particular impulse tone, since such tones tail off more gradually in music than in test impulses. Resonance peaks of two amplifiers of the same power and with identical frequency characteristics, may affect the actual music quite differently, and if the violin notes reproduced sound unnatural the cause is sometimes to be found in the transient characteristic.

Resonances in the ultrasonic range are another possible source of trouble. Random vibrations resulting from stray impulses die away gradually. It can be shown that such vibrations are invariably accompanied by others at frequencies below resonance, which in many cases are audible. Something of the kind also happens in spark transmitters, which are audible not only on the proper wavelength but also throughout a whole range of frequencies. So also with radio interference. It is worst on a particular wavelength, but also affects a whole range of other wavelengths, usually the more so at low frequencies (long waves).

Section 3. Hum and noise.

Quite rightly, amplifiers are expected to reproduce only what is put into them, and to cut down hum and noise as much as possible. In some cases, however, this is easier said than done, for reasons which will now be outlined.

Firstly, all amplifier valves and transistors, and all resistors, generate a certain amount of noise. If such noise is amplified enough it becomes audible, as in very

sensitive amplifiers, say, for use with electrodynamic pickups.
Secondly, amplifier valves are also a source of hum, caused partly by the heater in response to the A.C. supply voltage, and partly by the slight ripple always present in a D.C. supply obtained by rectifying A.C. mains voltage.
Again, magnetic and electric fields from the mains transformer or from the mains itself may produce small A.C. voltages at the input of the amplifier. Experience has shown that 10-watt amplifiers with a noise-level 60 dB below this rated value are for all practical purposes free from interference. With such amplifiers the hum or noise is completely drowned by the music, and becomes audible only in a quiet room when the pickup is lifted from the record, so that needle-hiss also stops; even then, these unwanted sounds are very faint and it takes a very sharp pair of ears to hear them.
In this case the power of the hum or noise reaching the speaker is 0.00001 watt, or 0.01 milliwatt. Supposing that 0.1 V must be applied to the input of the amplifier to give an output of 10 watts, then, the input voltage to limit the interference to 0.01 milliwatt will be one thousand times less than 0.1 V, that is 0.0001 V or 0.1 mV. Since all amplifiers carry A.C. voltages of 200 V or more they should be

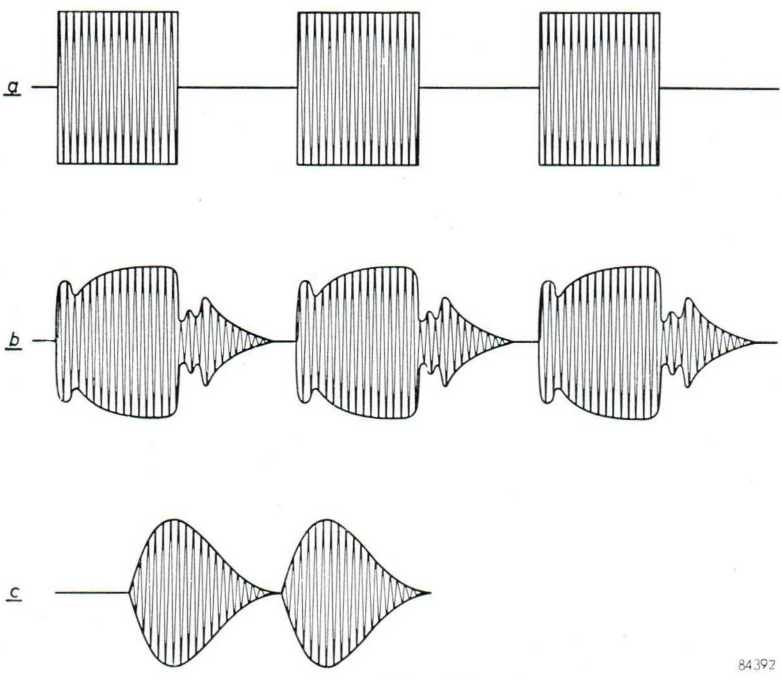


Fig. 59. Distortion of impulses:
a) The signal without distortion *b) The distorted voltage* *c) Musical tone*

so screened as to ensure that less than half of one millionth part of this voltage reaches the input of the amplifier. Amplifiers should therefore be wired very carefully and provided with suitable earth leads, because alternating currents in the chassis, say, from a smoothing capacitor to another point, may otherwise induce an intolerable hum voltage in the input leads of the amplifier. Of course, to avoid going beyond the above values it is necessary to smooth the anode voltage very effectively.

Another point worth mentioning is that with too much hum in the amplifier the hum voltage may easily become strong enough to partly load the output stage, so that this is more likely to be overloaded by strong signals than it would be without the hum. Although very much less smoothing and screening is necessary with speakers which do not respond to the lowest frequencies, too much hum nevertheless produces distortion, which is usually all the more difficult to trace because the hum itself is inaudible.

Section 4. Tone control

Since the discs themselves are not recorded to a straight frequency characteristic, it is necessary to effect certain corrections in the amplifier, and even when these corrections are fully applied, further adjustment of both the treble and the bass is nevertheless desirable in many cases. Such adjustments may be simply a matter of taste; many listeners are partial to a slight (or sometimes considerable) accentuation of the bass, whereas others prefer to boost, or to cut down, the treble to some extent. The ultimate quality of the sound also depends upon the acoustic properties of the room, and, last but not least, experience has shown that differences sometimes occur as between records allegedly made to the same recording characteristic. Tone control, then, is a supplement to playback correction. Both can be procured with the aid of circuits comprising resistors, capacitors and coils.

Coils are not used in good-quality amplifiers, however, because in conjunction with the inevitable capacitances they very soon cause audio-frequency resonance.

Accordingly, only resistance-capacitance circuits are employed for this purpose. The frequency-characteristic can be given the required shape in several ways. One of these is to include in the negative feedback circuit a combination of resistors and capacitors whose object is to reduce or increase this feedback, and thus boost or cut down the amplification of a certain range of frequencies relative to others. This method is not used very much in really first-rate amplifiers, however, because it leads to a certain amount of instability and also because it affects the transient characteristic. Filters are therefore added to the ordinary amplifier circuit, preferably at points where the amplification-level is low, since a tone filter coming immediately after an amplifier valve of relatively high gain affects the transient characteristic.

Tone control is more costly than would appear, as the actual effect of the resistors and capacitors is to cut down certain voltages more than others. What is known as bass or treble boost really means that the particular signals are not attenuated as much as others at higher, or lower, frequencies. To boost the bass, say 12 dB above the treble and the middle register, the tone control simply cuts the last-mentioned tones down 12 dB, therefore the amount of amplification required to load the amplifier fully is then 12 dB more than without tone control.

The most difficult aspect of the matter is the correction of the playback characteristic. As will be seen from fig. 37, this involves cutting down the amplification by roughly 18 dB, that is, a loss of roughly 8 times as compared with „straight" response.

When the correcting filter is connected direct to the input of the amplifier, it very soon cuts down the original output voltage of the pickup to such an extent as to give undue prominence to hum and noise from the amplifier. It is therefore desirable to pre-amplify the pickup voltage; this should not be carried too far, however, or it will affect the transient characteristic.

It is for this reason that in good amplifiers the gain in the correction stage does not go beyond 3 or 4 times. Fig. 60 shows two correction stages a) for use with triodes and b) for use with pentodes. Both are designed in accordance with the I.E.C. characteristic (1 in fig. 37 and fig. 38). It will be clear that electrical playback correction is not always necessary; such correction can be dispensed with when the pickup itself provides a certain amount of mechanical and/or acoustical correction. As a case in point we have the response curve of a pickup type AG 3013, for long-playing records; here, all that is necessary is a certain amount of treble attenuation; no bass-boost is required.

Tone control in its simplest form is treble control effected by means of a capacitor and a variable resistor in series, in parallel with the speaker transformer. This is only a means of cutting down the treble response of the amplifier more, or less. Given a pickup whose characteristic rises steeply towards the higher frequencies, maximum resistance will produce a rise in the overall response curve, and minimum resistance a certain falling-off in treble response. For ordinary amplifiers 50,000 μF and 0.1 Megohm may be employed, but this method of tone control is less suitable for quality amplifiers. Also, not every pickup has a characteristic rising towards high frequencies, and in any case some form of bass control is desirable in High Fidelity reproduction; in fact, it is essential as a means of compensating the differences between individual recording characteristics. A circuit enabling bass and treble to be controlled independently and providing both amplification and attenuation is shown in fig. 61. This tone control causes roughly 20 dB of attenuation at 1000 c/s;

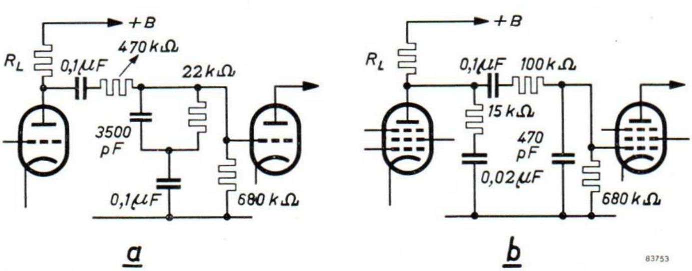


Fig. 60. Filters for playback-correction: a) With triode b) With pentode

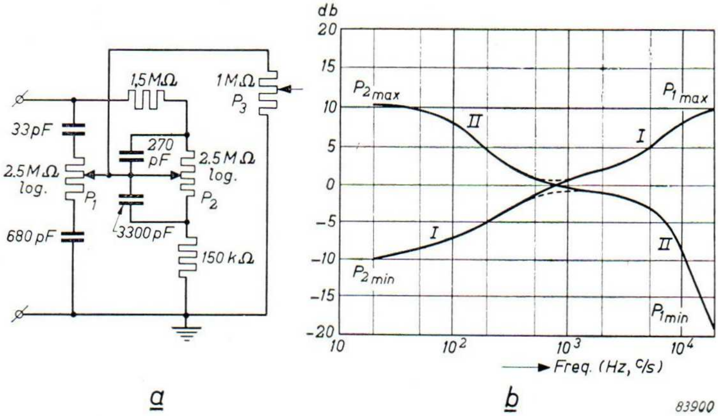


Fig. 61. Circuit for treble and bass control

the curves in fig. 61b refer to such a control operated in conjunction with a crystal pickup having an internal capacitance of 2000 pF.

Accordingly, the total loss of amplification owing to tone control and playback correction is about 38 dB, or nearly 1000 times, which in many cases necessitates adding another amplifier valve. The best way of doing this is to combine the tone control and correction circuits. Such a combination, giving an overall amplification of 4 times, is shown in fig. 62. Further amplification, although it can be obtained with this circuit, involves a risk that the quality of reproduction will be affected by undesired decay effects.

Section 5. Volume control

There are also one or two things to be said on the subject of volume control. The human ear is not able to perceive all frequencies equally well. It is most sensitive to the middle frequencies, and less so to the relatively higher and lower frequencies. This variation is shown diagrammatically in fig. 63, from which it will be seen that with sound of high intensity (1), as produced by playing a gramphone loudly (at the risk of trouble with the neighbours), the differences in sensitivity are relatively small. With low sound-intensity, that is, when the gramophone is being played as softly as possible (3), the ear is very much less sensitive to the lower frequencies. The middle line of the three (2) indicates what is more or less the normal listening level. It is seen from this diagram that turning down the volume of a gramophone not only reduces the intensity of the sound, but also affects its timbre. Too preserve the original timbre, then, it is necessary to adjust the tone control whenever the volume is turned up or down. It is not necessary, and in fact it is not desirable, to make good all the loss of aural sensitivity to the lowest frequencies occurring when the sound intensity is reduced. Our hearing of "live" music is similarly affected, as we notice for instance when a band is passing by; the music sounds flatter when the band is in the distance, and becomes fuller as the band draws nearer. Our ability to estimate how far we are from a particular sound source is partly attributable to this effect. Since it is physiological, any attempt to compensate it fully will produce an unnatural, and therefore to us unpleasant, sound. Taking curve 2 of fig. 63 as the norm, since it is plotted at normal sound intensity, we see that a slight bass attenuation, say, 5 db for 100 c/s, is permissible at high sound levels, and a correction not exceeding 10 db at 100 c/s for sound reproduced very softly.

In some radio-receivers and amplifiers, what may be described as physiological volume control or loudness control is employed. This means that when the volume control is turned down, the bass is attenuated slightly less than the treble and middle register. The disadvantage of such loudness control as applied to gramophone amplifiers is that it only works properly with one particular type of pickup; with any other pickup physiological volume control gives either too much, or too little compensation, depending on whether the output voltage of the pickup is above, or below the optimum. The final result also depends, of course, on the room in which the particular equipment is played. A given setting of the volume will produce a higher sound level in a small room than in a large one; whereas it is possible to say more or less accurately in what size room a particular radio receiver will be used, it is very much more difficult to make a similar prediction for gramophones. Therefore, loudness control is not always used in gramophones.

Section 6. Negative feedback

One important feature of amplifier circuits, which, although it is sometimes mentioned in advertisements, does not convey very much to the layman, is the negative feedback circuit. The principal purpose of such circuits is to suppress non-linear

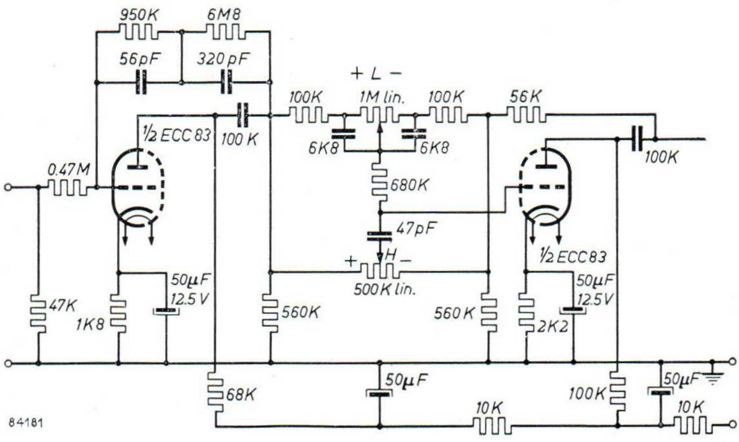


Fig. 62. Pre-amplifier with tone control and playback correction, suitable for crystal, magnetic and dynamic pickups

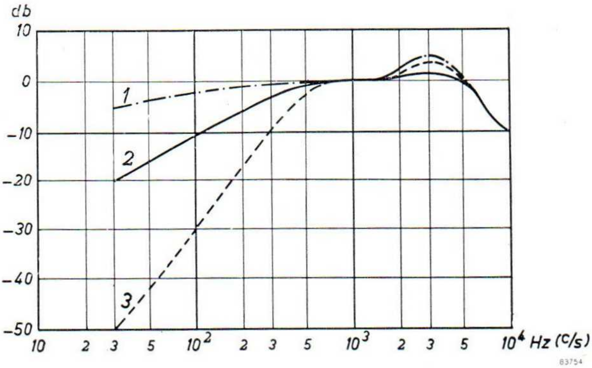


Fig. 63. Auditory-sensitivity characteristics:
1) High sound-level 2) Normal sound-level
3) Low sound-level

distortion, and the manner in which they do so will now be explained with the aid of an example. Suppose that an amplifier is fed with 0.1 Volt at 1000 c/s. The amplification is 100 times and the second-harmonic distortion 10%. We then have, at the output of the amplifier, 10 V, 1000 c/s plus 1 V, 2000 c/s of distortion. Now, 1/200 of the overall output voltage is fed back to the input in such a way as to oppose the original input voltage. At the particular moment, then, the voltage at the input of the amplifier is : 0.1 V — 0.05 V = 0.05 V, 1000 c/s, plus 0.005 V 2000 c/s.

It is thus evident that the negative feedback halves the output voltage; to keep this voltage at the proper level, the voltage applied to the amplifier must be increased to 0.15 Volts (see fig. 64), making the effective voltage again 0.15 — 0.05 = 0.1 V. The 0.005 V, 2000 c/s component of the feedback voltage is amplified 1000 times to give 0.5 V at the output in such a way as to oppose the original distortion voltage in the output; in this way the distortion voltage itself is cut down by 1 — 0.5 = 0.5 V, making the component fed back to the input of the amplifier 0.0025 V instead of 0.005 V. The amount of distortion-compensation acquired is then smaller than as computed above, and the ultimate distortion is $6^2/_3$% instead of 10% without negative feedback. At the same time, the overall amplification is similarly only $^2/_3$ of the original amplification; therefore extra pre-amplification is required. In this case the distortion is not very much reduced, but $^1/_{20}$ of the output voltage as feedback to the input of the amplifier, instead of only $^1/_{200}$, cuts down distortion to 1.6%, which is a considerable improvement.

On the one hand, this also reduces the amplification to only one sixth, but on the other hand, as distortion occurs for the most part in the output stage and not very much in the pre-amplifier, the loss of amplification can be compensated, without affecting the overall performance, by employing a more powerful pre-amplifier.

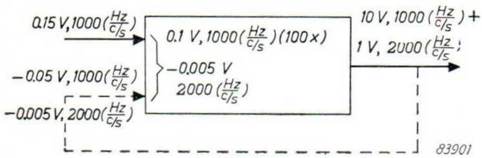


Fig. 64. Negative feedback

The above does not imply that this method of suppressing distortion involves no problems whatever. We have seen that part of the output voltage should be fed back to the input in such a way as to oppose the original input voltage. So also for all the frequencies at which the amplifier operates. With the aid of the above method it can be shown that if for some reason voltages at certain frequencies happen to produce positive, instead of negative, feedback, this does not reduce, but instead actually increases, the amplification, as also the distortion.

This not only distorts the frequency characteristic, but, at any rate if there is too much positive feedback, also involves a risk that the amplifier will become unstable and therefore oscillate. Such oscillations may take place at audio-frequency and then are easily discovered, or at a frequency high enough to be inaudible. In the latter case the oscillation may partly load, or even overload, the amplifier so as to distort the music. It is therefore fairly difficult to design negative feedback circuits suitable for High Fidelity amplifiers to cover a wide range of frequencies, and in which a considerable amount of negative feedback is required to keep non-linear distortion to a minimum. When the negative feedback is made dependent on the frequency, it affects the frequency characteristic, and in this way certain imperfections in the characteristic can be corrected. Given variable feedback the frequency characteristic of the amplifier can be varied as much, or as little, as necessary; at the same time this method of tone control is not suitable for use in High Fidelity amplifiers, since with an adequate range of tone control there is then too much risk of instability (oscillation) and the transient characteristic is also affected. One advantage of negative feedback is that if the speaker happens to produce decaying vibrations of its own after reproducing a certain tone, the voltages generated in the voice coil by these vibrations are carried by the feedback to the input of the amplifier and then return, amplified, to the speaker; here they suppress the original decay effects and thus improve the reproduction of impulse sounds.

Section 7. Amplifier circuits.

The principle of negative feedback can be applied in several ways, which cannot be discussed in detail within the scope of this book. Properly applied, this principle produces remarkable results. This is illustrated by fig. 65, showing a) the frequency characteristics, and b) the distortion characteristics, of an amplifier:

1. with negative feedback, and
2. without negative feedback.

These curves refer to the circuit shown in fig. 66. Here, the feedback voltage is taken from the secondary of the speaker transformer and applied to the input of the amplifier. Resistances R3 and R5 are such that about $1/_{200}$ of the output voltage reaches the input of the amplifier. The amplification is 1250 times without negative feedback, and only 170 times with this feedback; in other words it is reduced to less than one seventh, as also the distortion. Capacitor C 5 gives the feedback voltage the correct phase shift at high frequencies.

For all practical purposes, only push-pull amplifiers are employed in high-quality reproduction. Such amplifiers offer certain advantages, principally less distortion, and more output with a smaller transformer, than when a single output valve is employed.

The circuit shown in fig. 66 is unusual in that most of the amplifier stages are coupled direct, instead of across capacitors or transformers. This is to ensure the best possible reproduction of the very lowest frequencies. Capacitors and transformers not only attenuate the lowest frequencies, but also cause phase rotations, which would interfere with the negative feedback.

Although in this circuit the grid potential of the left-hand triode in the ECC 83 is positive, the very high cathode resistance of this valve (68000 Ohms) provides enough bias to ensure that the grid is nevertheless negative with respect to the cathode. So also with the right-hand triode. The left-hand system of the ECC 83 operates as an ordinary amplifier. The grid of the right-hand system is earthed across capacitor C8 in order to be free from A.C. voltages. Such voltages do occur on cathode-resistor R9, as it is not bypassed. This increases the overall negative feedback, but at the same time provides an effective A.C. voltage between grid and cathode of the right-hand triode. This voltage is in counter-phase with the other A.C. voltage on the grid of the left-hand triode; hence the voltages across R10 and R11 are likewise in counter-phase, as required to drive the push-pull output stage. This is coupled to the pre-amplifier through capacitors to prevent the anode voltage of the ECC 83 from reaching the grids of the output valves.

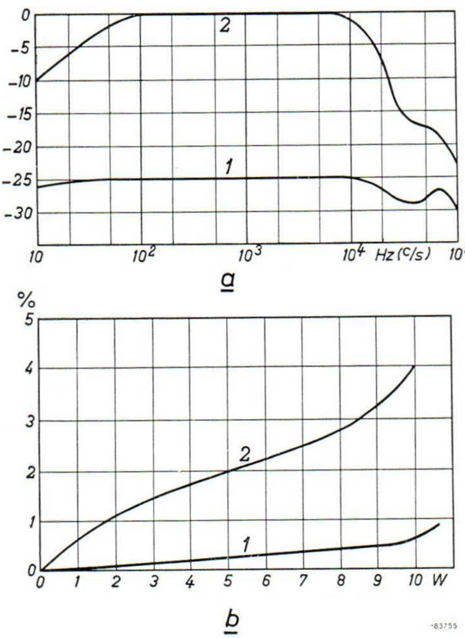


Fig. 65. Effect of negative feedback:
a) Frequency characteristic (1) with and (2) without negative feedback. b) Distortion (1) with and (2) without negative feedback

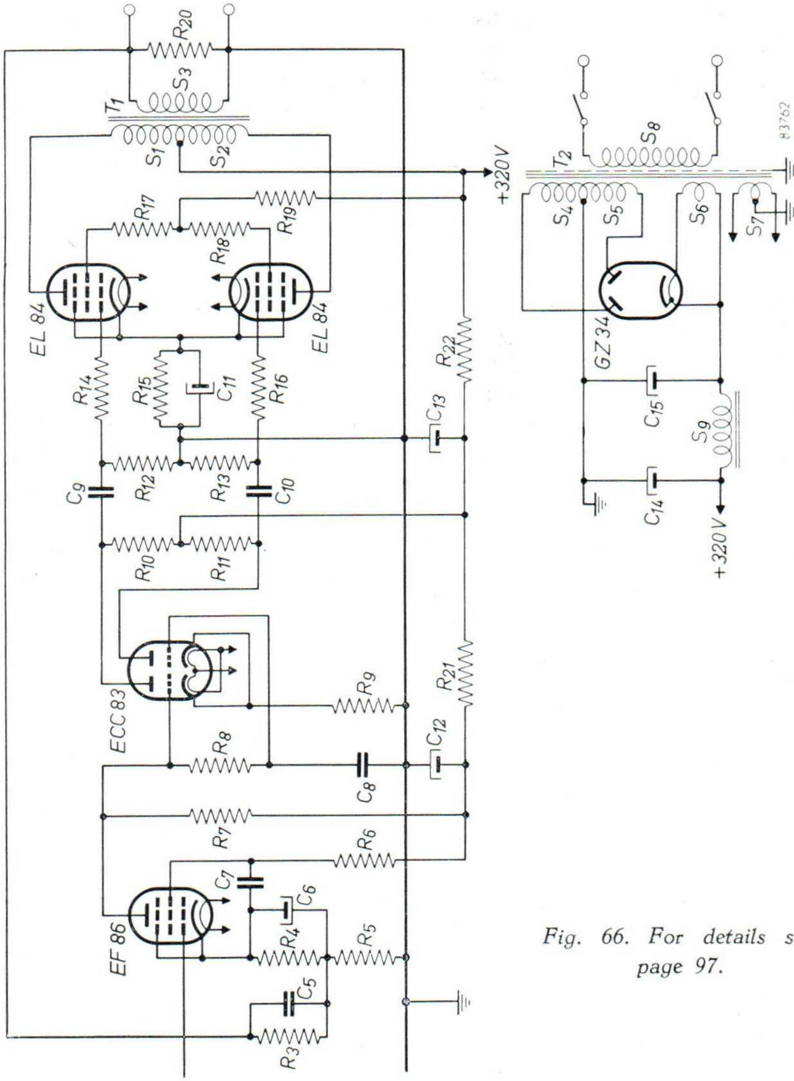


Fig. 66. For details see page 97.

Details of simple quality-amplifier (see fig. 66).

R_3	2.2	kΩ	¼ W	R_{19}		3.9	kΩ	1 ,,
R_4	2.2	,,	¼ ,,	R_{20}		1	kΩ	¼ W
R_5	10	Ω	¼ ,,	R_{21}		47	,,	½ ,,
R_6	1	MΩ	¼ ,,	R_{22}		27	,,	½ ,,
R_7	180	kΩ	1 ,,					
R_8	1.2	MΩ	¼ ,,	C_5	mica	1500	pF	
R_9	68	kΩ	½ ,,	C_6	electrol.	100	μF	12,5 V
R_{10}	0.1	MΩ	½ ,,	C_7	paper	47000	pF	400 ,,
R_{11}	0,1	,,	½ ,,	C_8	,,	0.1	μF	400 ,,
R_{12}	0.33	,,	¼ ,,	C_9	,,	0.1	μF	400 ,,
R_{13}	0.33	,,	¼ ,,	C_{10}	,,	0.1	μF	400 ,,
R_{14}	1	kΩ	¼ ,,	C_{11}	electrol.	100	μF	25 ,,
R_{15}	130	Ω	3 ,,	C_{12} }		50 + 50	μF	355/400 ,,
R_{16}	1	kΩ	¼ ,,	C_{13} }				
R_{17}	220	Ω	¼ ,,	C_{14} }		50 + 50	μF	355/400 ,,
R_{18}	220	,,	¼ ,,	C_{15} }				

Transformers:

T_1 Output transformer type AD 9000

T_2 Mains transformer: 2 × 280 V, 130 mA; 6.3 V, 2 A; 5 V. 1.9 V.

S_9 Choke type 7833 : L = 8 H; R = 2000 Ω; I_{max} = 115 mA.

Fig. 67 is the block diagram of a more ambitious type of amplifier, the AG 9000. This has two inputs, i.e. R for Radio or Tape Recorder, and ⌕ for gramophones. ⌕ is connected to a special amplifier valve in series with a filter for playback correction. A potentiometer is provided to enable the next valve to be connected either to the radio input or to the corrected gramophone-input.

The network between this valve and the next is the tone control, both stages are designed in such a way that transient distortion is negligible. Next, we have an amplifier — phase inverter, similar in principle to that included in fig. 66. The push-pull output stage is unconventional in that the output transformer is part of the cathode circuit. One advantage is that this simplifies the design of the transformer. The anode voltages are obtained from three separate rectifiers, partly in order to keep hum to a minimum and partly because the use of more than one rectifier makes the amplifier all the more stable.

The response characteristic of the amplifier can be varied by means of two controls (see fig. 68); also, the actual response curve has been made visible as a line behind the frequency dial which is bent into the shape of the characteristic when the controls are turned. This not only helps the listener to adjust the tone balance when playing records with recording characteristics different from that for which the amplifier is corrected, but also provides him with immediate evidence of any tendency of his own to give too much boost (or cut) to bass or treble. Ultimately, this adds to the pleasure of listening. Behind the small window seen on the left in the picture is a lamp, which lights up whenever the output of the amplifier approaches the highest level consistent with undistorted reproduction. This warning enables the volume to be turned down on the approach of a loud passage in the music.

A striking feature of this amplifier is the size of the output transformer, always an important component in amplifiers. When it is too small, the lowest frequencies are reproduced too faintly and are soon distorted. At the same time it is a mistake, too often made, to think of this matter only in terms of the bass, since a number of problems associated with the higher frequencies are also involved. The principal difficulty here is that the transformer coils have capacitance as well as self-inductance. Such parasitic capacitance in conjuncton with what is known as the leakage inductance of the transformer forms a tuned circuit, which not only affects the frequency characteristic, but also interferes with the negative feedback, sometimes completely upsetting it. Leakage inductance has the same effect as a coil in series with the transformer; it arises from the fact that, even in the best transformers, the coupling between primary and secondary is never quite perfect. In good transformers, however, the amount of leakage inductance associated with, say, 50 H of primary self-inductance is less than 50 mH.

With this amount of leakage inductance, the parasitic capacitance of the transformer and associated amplifier valves must be less than 180 pF to ensure a resonant frequency of at least 50,000 c/s. It is therefore necessary to keep parasitic capacitance and leakage inductance to a minimum, and this necessitates very accurate winding, as also a fairly large transformer. Moreover, push-pull amplifiers impose another

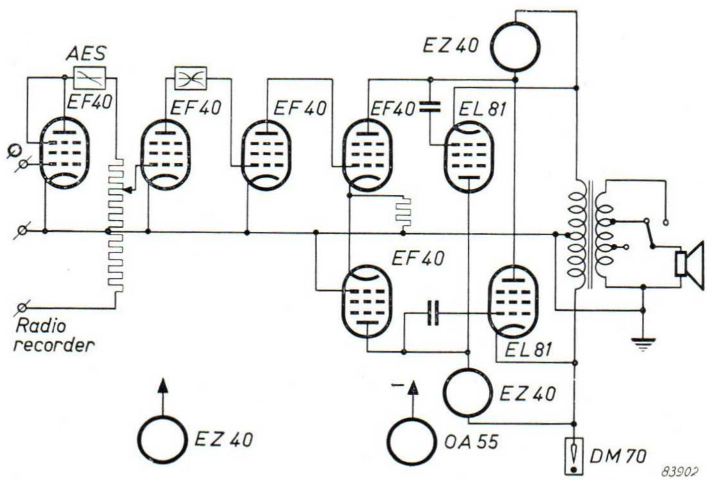
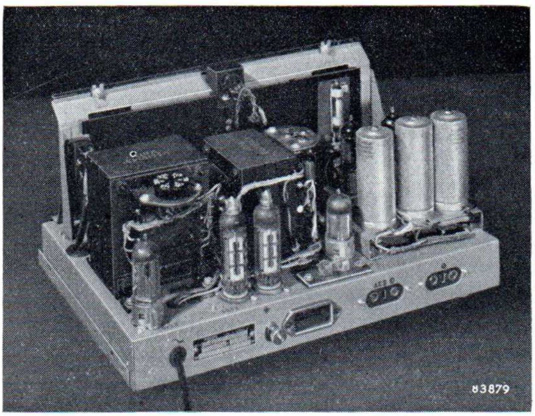


Fig. 67. Block diagram of High Fidelity amplifier type AG 9000

requirement on transformers, namely that the two half-sections must be identical. Such special features add so much to the cost of "High Fidelity" transformers that some of them are almost as expensive as small radio receivers. It is also worth mentioning that several circuits which do not require output transformers have been developed — however, they can be used only with special high-resistance speakers.

Fig. 68. Amplifier AG 9000, with and without protecting cover.



This is impracticable in the case of ordinary circuits because it necessitates matching-resistances of 5000 Ohms or more, and voice-coil resistances of this order of size cannot be obtained in electrodynamic loudspeakers.

Load-resistance values for the transformerless circuits are usually between 400 and 1000 Ohms. The advantages offered by transformerless output stages are less linear and non-linear distortion over a wide frequency range.

CHAPTER IX-A

LOUDSPEAKERS: OPERATION AND CHARACTERISTICS

Section 1. Electrodynamic speakers

The task of the loudspeaker is to convert the amplified electrical vibrations into sound vibrations. The requirements imposed on it are the same as for every other link in the gramophone chain; high efficiency, minimum distortion and an extended, and as far as possible straight, frequency characteristic. Such a characteristic, in particular, is by no means easy to obtain. As with pickups, many different methods of meeting these requirements have been tried; at the same time there is one important difference between pickups and speakers, namely that whereas the former can only be used one at a time, the latter are quite often used in groups, for various reasons connected with the reproduction.

Before dealing with the main subject of this chapter, we shall discuss the different speaker systems employed to convert electrical vibrations into sound vibrations.

Taken in order of importance, these systems are:

1. electrodynamic speakers;
2. electrostatic speakers;
3. crystal speakers;
4. electromagnetic speakers.

The „Ionophone", the hot-wire speaker, the singing arc speaker and some other systems which are not used in gramophone reproduction with not be discussed here. The systems which we shall discuss are similar in principle to the various types of pickups, i.e. electrodynamic, condenser, crystal and magnetic.

Electromagnetic speakers were the most widely used about 25 years ago, but have since been pushed right out of the picture by the electrodynamic speaker; therefore they are no longer used nowadays, at any rate in quality reproduction-equipment. Crystal and electrostatic speakers have become rather popular lately as special treble-units, or tweeters, for use with electrodynamic speakers, but recent developments suggest to the writer that this popularity may be only temporary. All these speakers operate on the principle that a vibrating diaphragm makes the air around it also vibrate; the differences are mainly in the manner in which the diaphragm is made to vibrate. In electrodynamic pickups voltages are induced in a coil by the movements of the coil in a magnetic field, in very much the same way as in a generator. Conversely, the coil may be made to move by applying a voltage to it; this is the principle of the electrodynamic speaker, as also of the electric motor. As regards appearance there is very little difference between electric motors and generators. Not so with pickups and loudspeakers; in the former the forces moving the coil are quite weak, whereas in the latter very much stronger forces are available, enabling very much sturdier components to be employed. From the other point of view, sturdier components are required precisely because it takes relatively strong forces to make the air vibrate. It is seen from fig. 69, showing an electrodynamic speaker in cross-section, that the principal parts of such speakers are a magnet system (I) and a diaphragm (II). The magnet system comprises a permanent magnet (1), a soft-iron core (2) and two pole-plates (3 and 4). The magnet, aided by the pole plates and the core, generates a strong magnetic field in an air-gap between the top pole-plate and the core. The diaphragm, or cone (5), carries a small coil (6) which fits into the air-gap. A flexible centring ring (7) and a rim, both attached to the speaker chassis, hold the cone in position; the magnet system is also attached to the chassis. When current flows in the voice coil (6), this moves forward or back, depending upon the direction of the current, therefore alternating current causes this coil to move to and fro, carrying the cone with it. These movements take place in frequency with the alternating current, and the cone-displacements are proportional to the strength of the current. Provided that the alternations of the current correspond exactly to the original needle-movements, then, the movements of the cone will be identical with those of the needle, as also the sound vibrations produced by the cone. As often happens, however, certain natural laws intervene at this point to make life difficult for the designer (and at the same time justify his existence) by presenting him with one or two very awkward problems. The cone moves by virtue of the current in the voice coil, and, as we have seen, the movements must be as far as possible proportional to this current. Now, it can be shown by means of a certain formula that the force acting upon the voice coil is $K = 0.1\ Hil$ dyne, where H is the magnetic flux density, i is the current, and l is the length of wire in the coil. The length of the wire, of course, is constant, and to keep force K proportional to current i, flux H must also be constant. Fig. 70 shows the air-gap, the top pole plate and the core enlarged. The dotted lines are magnetic lines of force, from which it will be seen that the flux is densest at the centre of the air gap and thins out to right and left of the centre. It follows that if the voice coil moves too far in either direction it will move through zones of different flux density, that is, no longer in a constant field, therefore the forces acting upon the coil, instead of being exactly proportional to the current, will be weaker. The effect is that the movements of the voice coil are also out of proportion to the current. Owing to this effect distortion would occur in electrodynamic speakers operated at above a certain sound level, but for the fact that special

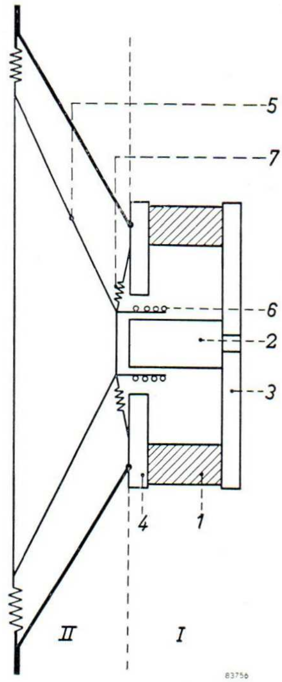


Fig. 69. Electrodynamic loudspeaker: 1) Magnet. 2) Core. 3) and 4) Pole-plates. 5) Cone. 6) Voice coil. 7) Centring ring.

measures are taken to prevent it. Such distortion can be suppresed in two ways. Firstly, the voice coil can be very much extended, so that both ends of it emerge from the field. The flux in the coil then remains constant through the strongest vibrations occurring under practical conditions. One disadvantage is that extending the coil adds to the overall weight of the cone and thus affects the treble response. Another is that the coil-turns beyond the strongest part of the field add nothing, or at any rate not very much, to the forces moving the cone, and at the same time the resistance of these turns causes a loss of power.

Another way of solving the problem is to make the air-gap longer than the coil itself, so that the latter moves always in that part of the field which is practically constant. This method affects neither treble response nor efficiency, but necessitates a larger magnet and therefore makes the speaker more expensive. Nevertheless, it is most suitable for quality speakers such as types 9758, 9762 and 9710. The magnet is highly important, in the first instance, in that it governs the sensitivity of the speaker, as will now be explained. The efficiency of a given speaker is proportional to the square of the magnetic flux in the air-gap; therefore it is desirable to have this flux as high as possible. At the same time it is fairly obvious that a given magnet will produce a stronger field in a narrow air-gap (relatively short distance from core to inner edge of pole plate) than in a wide one. The width of the air-gap depends in the first instance upon the thickness of the voice coil, and, as this must never touch either the core or the pole plate, much depends also upon the precision of the coil and the centring ring. Given enough accuracy in the making of the coil and the ring, and the means to ensure that they will not be distorted when in use, it will be possible to ensure a certain speaker-efficiency with a narrow air-gap and a correspondingly small magnet. Otherwise, the wider air-gap will necessitate a heavier magnet. Another important factor, of course, is the quality of the magnet steel. This is measured in terms of what is known as the BH_{max}, which in pre-war steels was roughly one million. Modern „Ticonal" steels have BH_{max} values up to seven million. It will be evident from this, and from what was said in the preceding paragraph, that it is impossible to judge the sensitivity of a speaker merely from the size or weight of the magnet. The weight is a basis for comparison only when the type of steel and the size of the air-gap are also known, and provided that the magnet is properly designed. Moreover, the properties of the soft-iron poleplates and core also affect matters. Because these properties become less and less effective with increasing flux

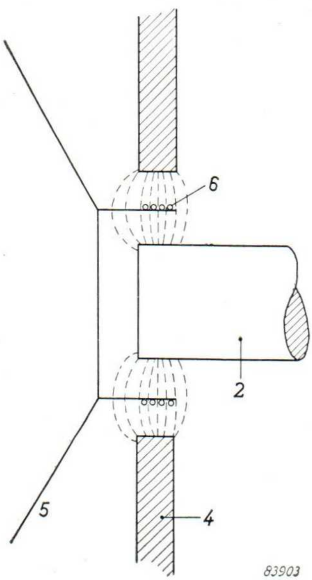


Fig. 70. The magnetic field in an electrodynamic speaker: 2) Core. 4) Pole-plate. 5) Cone. 6) Voice coil.

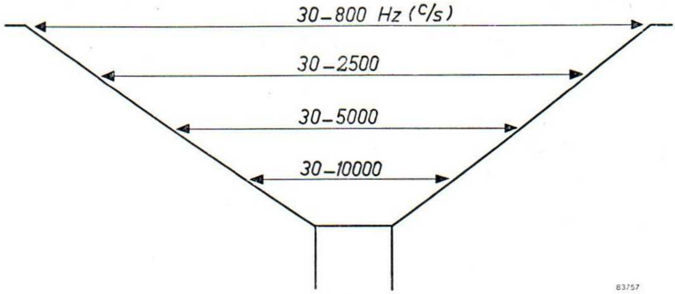


Fig. 71. The distribution of sound over a large cone

density, it is difficult to increase the field strength very much beyond about 13,000 gauss. The cone presents problems even more difficult to solve. Many of the magnet data can be computed, whereas cone design, apart from the scientific principles, is largely a matter of experience. The cones of electrodynamic speakers are of moulded paper treated with certain lacquers to make it damp-proof; these lacquers affect the acoustic properties. Now, the vibrating cone sets the surrounding air in vibration. As explained in Chapter IV, section 6, low-frequency vibrations are large and relatively slow, and high-frequency vibrations small and fast. The larger the cone the more readily it makes the air vibrate; in other words, the cone is then more efficient. This applies especially to the bass; with a small cone the relatively slow movements do not give it enough „grip" on the air, therefore speakers with a large cone have the best bass-response. Although in theory good bass-response can also be obtained with a small cone in practice the amplitude of the cone vibrations would then have to be so large that they would cause distortion. Very much smaller cones can be used for treble reproduction; in fact large cones are quite unsuitable for this purpose, being too heavy and therefore too slow to follow the extremely fast treble-vibrations. The higher the frequencies to be reproduced the smaller the maximum permissible cone diameter; at the same time, nature, in an unusually generous mood, comes to our assistance by so arranging matters that a complete range of frequencies can be reproduced without the aid of many different speakers.

It can be shown that at high frequencies vibration in large cones is confined mainly to the middle part of the cone. Up to roughly 1000 c/s the whole cone vibrates as a single, rigid unit, whereas beyond 1000 c/s only part of the cone vibrates and the vibrating part

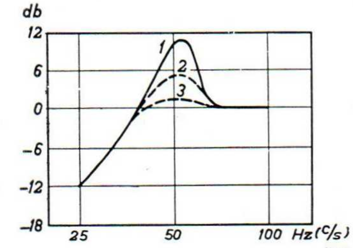


Fig. 72

The fundamental resonance of: 1) An ordinary loudspeaker. 2) A speaker provided with a cover. 3) A speaker with air-cushion.

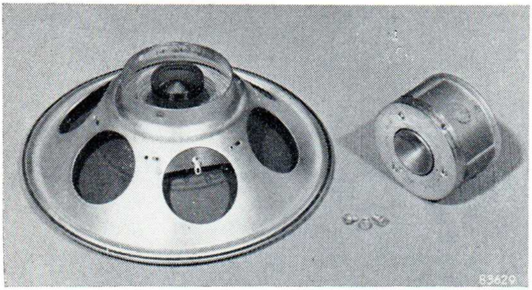


Fig. 73. Speaker type 9710; note the tip of the cone in the voice coil, and the recess in the core

becomes smaller and smaller as the frequency increases. The response ranges of certain parts of the cone are shown diagrammatically in fig. 71. This diagram is only approximately correct, however, since in practice treble vibrations are propagated all over the cone; at certain frequencies, therefore, it may happen that part of the cone moves forward just at the moment when another part is moving back, with the result that the particular tone is attenuated. Other tones are over-accentuated in very much the same way, and the overall effect is to produce a series of peaks and dips in the response characteristic. Such effects can be suppressed by providing the cone with reinforcing rings; also, they depend very much on the composition of the cone paper.

By and large, it is found that properly designed speakers 10 to 12 inches in diameter can be used to cover the range from 40 to 10,000 c/s, 8-inch speakers for from 70 to 12,000 c/s and 5 or 7-inch speakers for from 100 to 15000 c/s. In certain speakers the frequency range is deliberately cut down, and they are less suitable for use in gramophones; other speakers, however, cover an even larger frequency range than that given above.

Another point to be considered in connection with speakers is what is known as the fundamental resonance. Experience has shown that the combination of the cone, the centring ring and the surrounding air vibrates most readily at a certain bass frequency; tones associated with such resonances are reproduced with extra intensity, and below resonant frequency the sensitivity of the speaker decreases 12 dB per Octave. Such resonance peaks, like all resonances, affect the quality of reproduction, partly by over-accentuating certain tones and partly because they reverberate too long and thus make the music sound „boomy". Many different methods of suppressing the peaks are known; one of them is to provide the speaker with a tight-fitting cotton cover. Fig. 72 shows the response characteristics of two speakers, one with (solid line 1), and one without (chain line 2), such a cover. A very effective method of damping (see fig. 72, line 3) is to make a suitable recess in the core so that the apex of the cone can be taken into the voice coil instead of being cut off. This method is employed in speaker type 9710, shown in fig. 73. On the other hand, resonance peaks are sometimes useful in speakers enclosed in small cabinets, as we shall see in section 5. Of course, it is impossible to mention all the different types of electrodynamic speakers in the present discussion.

Section 2. Electrostatic loudspeakers

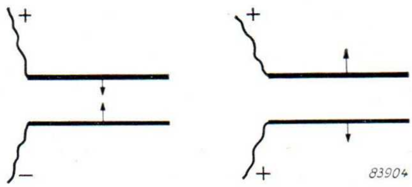


Fig. 74. Principle of the electrostatic speaker.

Electrostatic speakers are used only rarely in gramophones, but quite often as special treble-reproducers in radio receivers. Such speakers operate on the principle that a D.C. voltage applied between two parallel metal plates causes these plates to attract, or repel, each other (see fig. 74). The amount of push or pull depends upon the voltage. If one of the plates is flexible, it bends slightly; the exact amount of bend also depends on the voltage. As luck will have it, however, the attractive force is not proportional to the voltage; instead it is proportional only to the square of the voltage. The result is that with only A. C. voltage applied to the two electrodes, the movement of the flexible electrode would not be proportional to the voltage, but would increase very much more than the latter, thus distorting the reproduction. To reduce this effect, not only a A.C. voltage but also a very much larger D.C. voltage is applied to the two plates. It can be shown in the form of a simple calculation that with a D.C. voltage ten times the A.C. voltage, the distortion is less than 5%. As most amplifiers supply 250—300 V D.C., this means that the A.C. voltage should not exceed 30 V. The electrostatic forces of attraction depend not only on the voltage, but also on the space between the plates; this space must not be too small, or shorting and distortion will occur; hence the forces developed are not strong enough to move the diaphragm very much. The large amplitudes necessary for bass-reproduction in turn necessitate A.C. voltages far beyond the limit imposed by distortion; the alternative is to boost the D.C. voltage to more than 1000 V, which has many practical drawbacks. In view of these limitations, electrostatic speakers are employed only as treble-reproducers; although distortion can be reduced by placing the diaphragm between two fixed, perforated plates, this does not improve matters nearly enough to enable the bass, or even the middle register, to be reproduced effectively. Nevertheless, recent experiments with large electrostatic loudspeakers to reproduce low frequencies have attracted a good deal of attention.

Details of one of the different models now in use will be seen in fig. 75. Here we have one fixed plate (1), perforated so that is does not obstruct the sound waves and separated from the diaphragm (3), a layer of metal (usually gold or silver) only a few microns thick, by a thin layer of elastic insulation: the diaphragm (3) and the cover are earthed. The metallised insulation is kept taut by a spring (5), insulated from the fixed plate, which presses upwards against the actual loudspeaking-element. The insulation prevents shorting between 1 and 3, which might otherwise occur owing to dust between the two or to overloading of the speaker. One drawback of this system is that it makes the speaker rather sensitive to temperature variations; moisture is another possible source of trouble. Because electrostatic speakers have a rather high capacitance, which affects the frequency and transient characteristics, they are not easy to incorporate in a circuit.

Section 3. Crystal loudspeakers

Crystal speakers (see fig. 76) are really crystal pickups in reverse. In principle it is possible to make a loudspeaker by attaching a paper cone to the needle of such a pickup. In pickups the movements of the crystal are very small, however, whereas in loudspeakers they have to be very much larger. In fact, the movements necessary to reproduce the bass are large enough to crack the crystal; hence crystal speakers are used only as treble-reproducers (above 5000—10,000 c/s). Moreover the response characteristic is by no means straight, therefore such speakers are not used very much.

Section 4. Speaker characteristics

Speakers or groups of speakers should satisfy the following general requirements:

a. Efficiency as high as possible.
b. As far as possible uniform response over as wide a range of frequencies as possible.
c. Minimum distortion.

As with any other product, moreover, the price must be as low as possible, and in view of the amount of material that goes into every loudspeaker, some of it very expensive, this sometimes makes things very difficult for the designer.

The difficulty about speaker-efficiency is that it varies according to the frequency, owing to the fact that the response characteristics are never quite straight. It is usually measured somewhere between the fundamental resonance and the highest frequency consistent with overall cone-vibration (see fig. 71). The efficiency of a loudspeaker that is, the ratio of acoustical output to electrical input, depends upon the flux density in the air-gap, the size of the cone, the properties of centring ring and cone rim, and, last but not least, the properties of the cone itself. In practice it varies from 1½% for the smallest speakers with correspondingly small magnets to 15% for 12-inch speakers with very large magnets. Not long ago the average efficiency was 1%; nowadays speakers are more likely to be from 4 to 6% efficient, and some with efficiencies up to 50% are made for use in laboratories. These very efficient speakers are so expensive, however, that it is more economical to use a less sensitive speaker with a very much more powerful amplifier.

As the cost of the magnet adds considerably to the overall price of the speaker, sensitive speakers are fairly expensive. With an

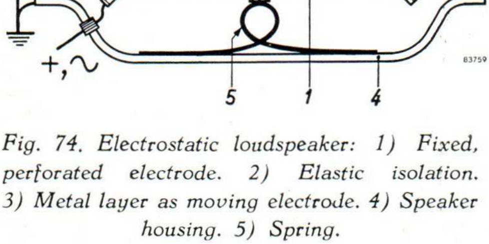


Fig. 74. Electrostatic loudspeaker: 1) Fixed, perforated electrode. 2) Elastic isolation. 3) Metal layer as moving electrode. 4) Speaker housing. 5) Spring.

extra-wide air-gap to improve the quality of reproduction, a larger magnet is necessary to ensure the same amount of efficiency. To avoid over-spending on the speaker it is sometimes necessary to choose between efficiency and freedom from distortion.

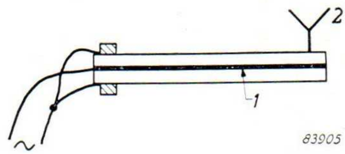


Fig. 76. Principle of the crystal loudspeaker: 1) Bimorph. 2) Cone

For example, we have speaker type 9710 combining maximum quality with average efficiency, as against speaker type 9762, representing the optimum in both quality (freedom from distortion) and efficiency. The choice between these alternatives is governed, then, by the size of the room in which the speaker is to be used, and the power of the amplifier; in a small room, or with a powerful amplifier, the speaker need not be very sensitive.

The response characteristics of speakers are rather difficult to plot, as they depend very much on the nature of the baffle, if any, and on the acoustic properties of the particular test-room. When a speaker-cone moves forward the air in front of it is compressed (+) and the air behind it rarefied (—). The cone then moves in the other direction and produces exactly the opposite effect. The sound waves thus occurring at the back of the speaker also travel, round the baffle, to the front. Therefore a listener sitting a certain distance (d) from the front of the speaker hears sound vibrations from the back, as well as those from the front, of the speaker (fig. 77). The vibrations from the back lag behind the others to the extent of the extra distance (a) which they have to cover. When this extra distance is very short, however, the lag is negligible, and the two waves then virtually cancel each other out — the compression at the front is for all practical purposes eliminated by the rarefaction from the back of the cone.

When distance (a) is not small enough to be ignored, there is a certain interval between two waves, so that they do not damp each other quite as much. The damping effect at high frequencies is small enough to be safely ignored in all baffle-calculations. It can be shown, in the form of a calculation as well as experimentally, that no damping takes place when distance a is $\frac{1}{4}$ of the wavelength. Formulated: critical wavelength = 4a, or: critical frequency = 343/4a (a in metres, 1 metre = 3.17 ft). Below the critical frequency the response goes by halves, that is, 6 db per octave. The importance of all this will be evident seeing that the wavelength is approximately 1 metre at 300 c/s, 3 metres at 100 c/s, 6 metres at 50 c/s, and so on. As the final result depends not only on the dimensions, but also on the shape and certain other properties, such as sympathetic vibration at given frequencies, of the baffle, loudspeakers can be judged very much more accurately by testing them without a baffle. The characteristic then falls steeply towards the bass, which is considered a drawback from the point of view of advertising; on the other hand, frequency characteristics are no use whatever in practice, unless full particulars of the baffle are also given.

Response characteristics are plotted by means of a calibrated microphone placed a certain distance, say, roughly two feet from the front of the loudspeaker. The frequency of the speaker current is then varied from 0 to 20,000 c/s without varying its strength; as the output voltage of the microphone is a measure of the sound pressure, this enables a characteristic to be plotted, as shown in fig. 79. It is necessary to see that the microphone is not affected by sound from any source other than the speaker, as also that it does not intercept other than direct radiation from the speaker itself, i.e. sound reflected from the walls of the test-room. For this reason, the speaker and the microphone are placed in what is known as a "dead" room. Such a room is shown in fig. 78. It is a thick-walled, cube-shaped compartment supported on heavy springs inside another, similar cube. Both are provided with heavy, sound-proof doors to give perfect silence inside the inner cubicle. All six walls of the test-room are studded with 5-foot wedges of sound-absorbing material, making it completely free from reverberation and echo. The silence in this room is awe-inspiring; after 15 minutes of it, shut off from the rest of the world, sounds like the beating of the heart, air rushing through the nostrils, and all manner of at other times inaudible noises caused by the digestive system are heard quite clearly. In fact such perfect silence is rather frightening and, after a short time, is apt to be too much even for those who are not naturally of a nervous disposition; nevertheless, very accurate tests can be carried out in this test-room. To eliminate background noise all the measuring equipment other than the microphone and what goes with it is placed outside the dead room. The signal generator and a roll of paper tape to record the microphone voltage automatically are driven by a motor. As a check, the signal generator is switched off for a moment at 1000 c/s; the resulting dip in the characteristic is reproduced in all the subsequent response curves and, of course, has no bearing whatever on the characteristic of the particular speaker.

It is seen from fig. 79 that between 800 and 200 c/s the characteristic falls about 6 db per octave. This is because the speaker is tested without baffle; except for the cone which acts as a baffle. The cone of the particular speaker (2a) is 8 inches in diameter, therefore tones with a wavelength of 16 inches (0.42 metre, corresponding to 800 c/s) are not attenuated by it, but all tones further towards bass are. Very much greater sound pressure occurs in the region of 90 c/s, owing to the extra sensitivity of the speaker at resonance; below this frequency the characteristic falls 18 dB per octave, that is, 6 dB per octave owing to the absence of a baffle and 12 dB per octave as the normal loss of sensitivity in all speakers below resonance.

Note that above 1000 c/s the curve rises instead of being flat; this is no accident, but part of the design of the particular speaker. The cone acts as a reflector of high notes, thus beaming them in very much the same way as a spot-lamp beams light.

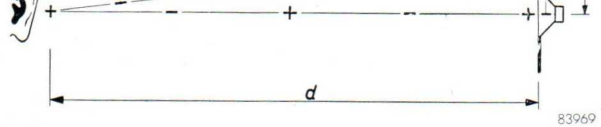


Fig. 77. Effect of baffle

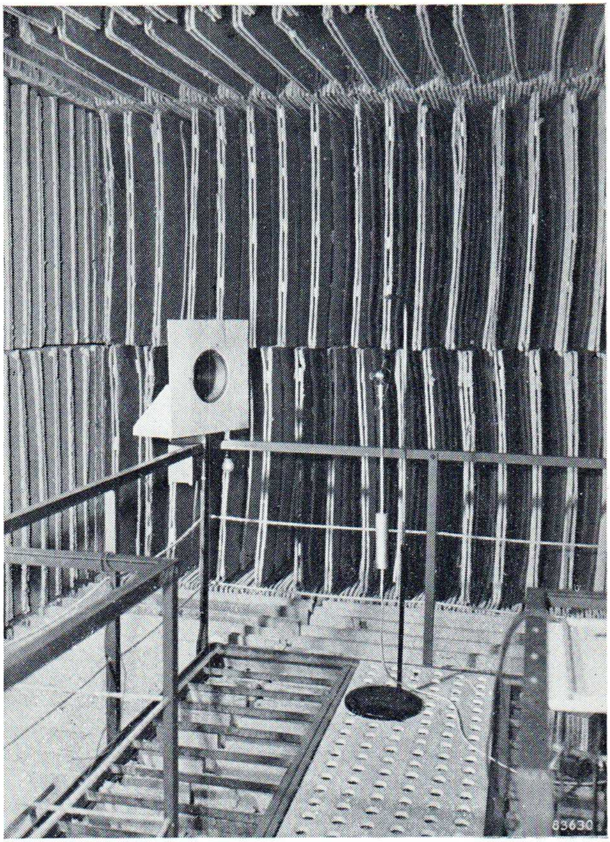


Fig. 78. Echo-free room for acoustic tests. Note the spherical microphone on the right of the speaker, here mounted on a baffle

Such notes sound very much stronger from straight in front of the speaker than from an oblique angle; because this effect is partly compensated by reflection from the walls and ceiling of ordinary rooms, it is the more noticeable in a dead room. If the overall sound energy be constant, then, as it should be, there will be a little too much sound-intensity straight in front of the speaker at the higher frequencies. On the other hand, with constant sound pressure straight in front of the speaker, the overall sound energy decreases towards higher frequencies, and the reproduction is therefore marred by too much bass, especially when it is heard from an oblique angle. Fig. 80a is the radiation, or directivity, pattern of speaker type 9770 (as plotted in the dead room shown in fig. 78) the length of a line drawn from the centre of the diagram to a point on one of the curves is a measure of the sound intensity towards that point. Such patterns can be improved, as shown in fig. 80b (speaker type 9770 M), by placing a sound-diffusing cone in the mouth of the cone or by attaching another, smaller cone to the voice coil. As we shall now see, the second cone does more than improve the radiation pattern. Ordinary cones large enough to have good bass response are less suitable for the reproduction of the highest frequencies. In some cases, therefore, an extra speaker with a small cone (tweeter) is used as a means of reproducing the treble.

The reponse characteristics of such tweeters must match those of the associated large speakers exactly; such matching is obtained with the aid of special filters. It is more logical to employ the other method, that is, attaching a special treble-cone to the voice coil. All that is then necessary to ensure that the characteristics are properly matched is to shape the treble cone correctly. Fig. 81 gives an example. Here, the solid line (a) refers to speaker type 9710; the other speaker referred to in this diagram, type 9710 M, contains an extra cone which keeps the response characteristic practically straight up to very much higher frequencies (chain-line b). Compare these diagrams with fig. 82, showing the response characteristic and radiation pattern of an electrostatic tweeter. It will be seen that the bass response is very low, but this does not matter very much in a speaker specially designed for high frequencies. At the same time, such speakers are liable to be damaged by low-frequency signals and are therefore usually connected to the amplifier across a simple filter.

Speaker-impedance has been mentioned in the previous chapter in connection with output transformers, and will now be discussed in more detail. The voice coils of electrodynamic speakers have a certain amount of D.C. resistance, depending on the length, the thickness and the material (usually copper) of the wire. Now, the resistance as measured with A.C. voltage is higher than the D.C. resistance. One reason is that voice coils, like other coils, have a certain amount of self-inductance, which causes the impedance (A.C. resistance) to increase with the frequency. Another is that A.C. current moves the voice coil. As explained in connection with electrodynamic pickups and microphones, voltage is induced in a coil when this moves in a magnetic field. So also with the voice coil; the voltage thus induced opposes the voltage applied to the speaker and therefore weakens the current in the voice coil.

This becomes more and more noticeable towards higher frequencies; therefore, given a constant voltage, the current will become weaker according as the frequency is raised. In effect, then, the apparent resistance of the coil increases with the frequency, to an extent depending

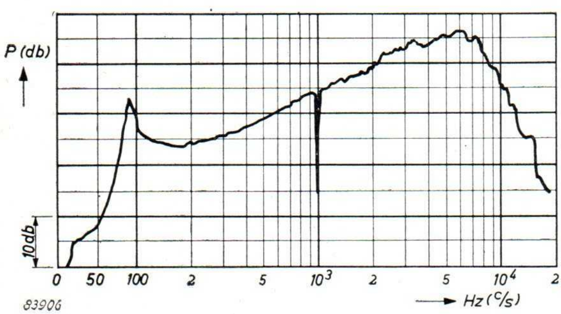


Fig. 79. Sound-pressure curve of speaker type 9770

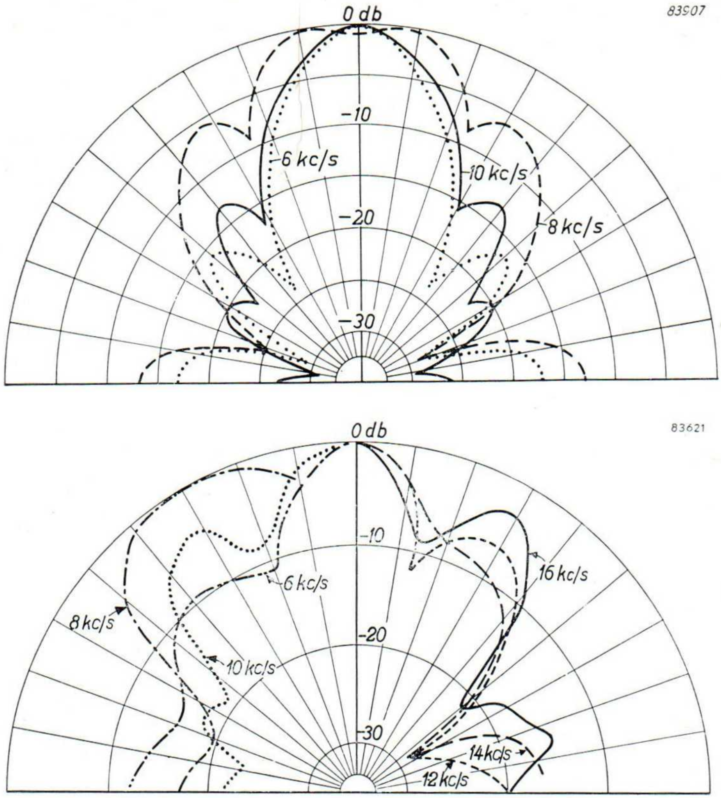


Fig. 80. Radiation patterns:
a) Speaker type 9770 b) Speaker type 9770 M

on the design of the speaker. The result is that perfect matching of speaker to output stage can be obtained only within a certain part of the overall frequency range. Mismatch between the two means a reduction of the maximum amount of power which the output stage can transfer to the speaker without distortion. As the reproduction of the lowest frequencies takes the most power, it is necessary to ensure that the speaker is properly matched at these frequencies. The full line in fig. 83 shows that perfect matching at 1000 c/s ensures reasonable matching over a wide range of other frequencies.

Again, it will be evident from the above that response characteristics plotted with constant voltage instead of with constant current give less treble response. This is also the case with certain amplifier circuits, where it is therefore necessary to provide a certain amount of treble boost, say, by cutting the negative feedback;

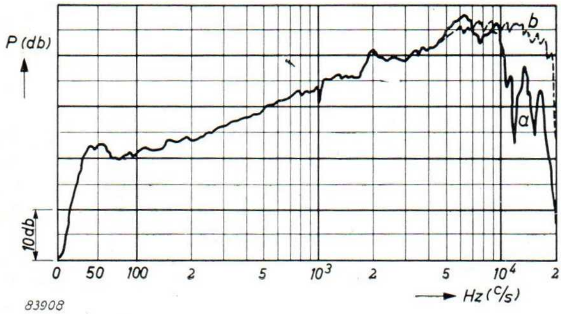
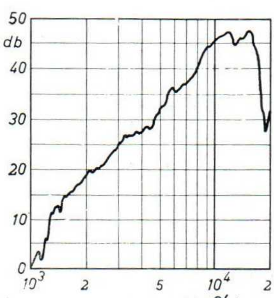


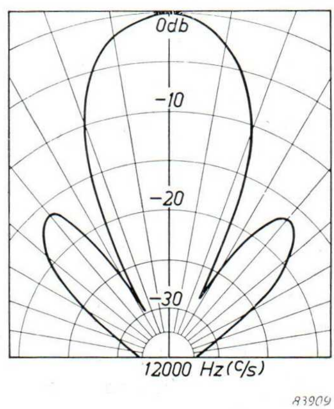
Fig. 81. Sound-pressure curves of:
a) Speaker type 9710 b) Speaker type 9710 M/88

such cutting, combined with the inevitable high-frequency mis-match, does much to mar the quality of reproduction. With this in view, certain quality-speakers are provided with a copper ring round the core at the point where this passes through the hole in the top pole-plate (fig. 69). The magnetic flux produced by current in the voice coil induces current in this copper "lining" and thus reduces the above-mentioned "counter"-voltages considerably. The result is a marked improvement in the impedance characteristic, as we see from the chain-line (b) in fig. 83. It is also worth mentioning in this connection that similar counter voltages occur at lower frequencies and affect the size of the resonance peak to some extent; by and large, the fundamental resonance is very much more pronounced in characteristics plotted with constant current than in those plotted with constant voltage.

Now a word or two on the subject of distortion in loudspeakers. Strictly speaking, the requirements are the same as for amplifiers and pickups, but distortion in speakers, whether it be non-linear, as in the reproduction of a single tone, or intermodulation, is very much more difficult to measure; hence the figures are not published, or at any rate are published only very occasionally in highly technical articles. Seeing that with good speakers it is possible to hear small differences in the measured distortion of amplifiers quite clearly, it is reasonable to suppose that the speakers themselves do not cause very much distortion.

Fig. 82. Electrostatic speaker:
a) Sound-pressure curve
b) Radiation pattern at 12,000 c/s





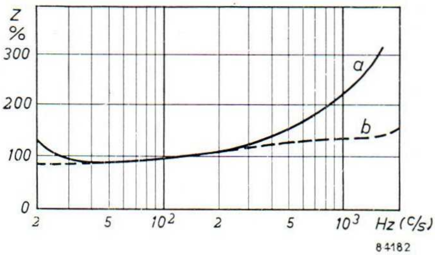


Fig. 83. Variation of speech-coil impedance:
a) Ordinary speakers. b) Speakers whose impedance is almost independent of the frequency

Much more than in amplifiers, distortion in speakers depends upon the frequency. Distortion near the fundamental resonance has the peculiar effect of apparently enhancing bass response. Distortion of a low note lacking intensity owing to insufficient baffle area produces overtones. Instead of recognising them at once as products of distortion, the ear "invents" a spurious fundamental to go with the spurious overtones; hence the distorted tone produces a very much stronger, and at the same time "woollier", impression than it would if it were not distorted. When this impression is compared with the original sound, the latter, of course, sounds best. Intermodulaton distortion in speakers is accompanied by the very closely related Doppler effect. It is a well-known fact that sound from a rapidly approaching source is more high-pitched than when the source is receding; for example, the whistle of a passing train.

The explanation is that in one case the apparent frequency of the sound waves is increased, whereas in the other it is reduced. Something of the kind also happens when a loudspeaker produces a high note and a very strong bass note simultaneously. The cone, the source of high notes, simultaneously moves backwards and forwards in frequency with the particular low note, that is. first towards, and then away from, the listener, so that the high note is heard first slightly too high, and then slightly too low, in pitch. This effect does not matter very much when the movements of the cone are very small, but may cause audible distortion when it occurs as a result of very large cone amplitudes.

For all pratical purpose such amplitudes occur only at the resonant frequency, at which the diaphragm vibrates most readily, yet distortion owing to Doppler effect can be kept below the threshold of audibility through effective suppression of the fundamental resonance. Owing to the difficulties associated with distortion tests it is customary to base the

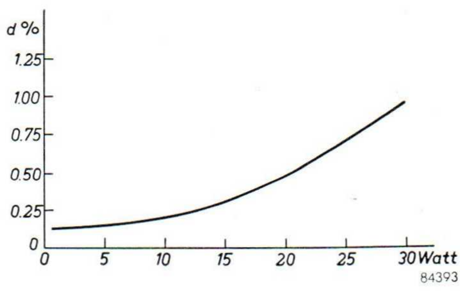


Fig. 84. Distortion characteristic, speaker-unit type AD 5002

maximum power rating of speakers primarily on life-tests instead of on distortion only. Rating a speaker at 6 watts means that it is safe to operate the speaker with a 6-watt amplifier. The amount of distortion at this wattage depends entirely on the design of the speaker; good speakers give only slight distortion up to just below the rated power-level, the break-down limit is about 50% above this level. To illustrate this point, fig. 84 shows the distortion curve of speaker unit type AD 5002, as plotted at 400 c/s. The rated power of this unit is 50 Watts, break-down occurs at about 70 watts.

CHAPTER IX-B

LOUDSPEAKERS: ACOUSTIC PROBLEMS AND SOLUTIONS

Section 5. Baffles and speaker cabinets

To operate efficiently, speakers such as we are now considering must be provided with a baffle, a cabinet, or a horn. Without these, such speakers would have no bass response. Also, they would soon develop defects, since owing to the short-circuit effect occurring when the shortest distance from the actual speaker to the edge of the baffle is less than ¼ wavelength, as explained in the preceding section, the cone moves so readily at low frequencies, and particularly at resonance, that any power reaching the speaker would cause the cone to deflect too much and so sustain damage.

Fig. 85 shows the critical frequency plotted against the smallest distance from speaker to edge of baffle. From the baffle areas as determined with the aid of this diagram it is easy to see that housewives would be liable to view any attempt to obtain "straight" response down to 60 c/s with great disfavour; a 10 x 10-foot baffle is a little too large for the average drawing-room.

Good response can also be obtained with a smaller baffle, however, provided that the dimensions are chosen with the resonant frequency of the particular speaker in view. Fig. 86 shows the response characteristics of a speaker with resonance at 60 c/s (full line) and another with resonance at 30 c/s (chain-line) both mounted in the middle of 5 x 5-foot baffles. It is seen that the speaker with the highest resonant frequency gives good response down to roughly 50 c/s; here the characteristic falls to 6 dB, which is just acceptable. On the other hand, the response characteristic of the speaker with resonance at 30 c/s falls to 6 dB at 60 c/s, so that for all practical purposes audible response in the bass is confined to those tones which correspond to the resonant frequency. There is also a risk of overloading at the resonant frequency, owing to the fact that the relatively small baffle does not provide enough acoustic load for the speaker; the

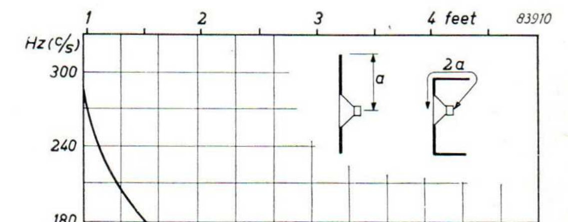


Fig. 85. Critical frequency plotted against baffle-size.

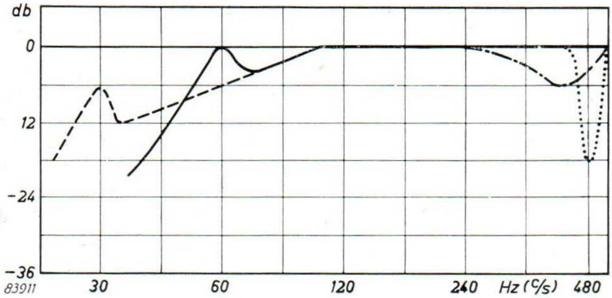


Fig. 86. Effect of baffle or speaker-cabinet on response.

principal outcome of this is distortion. In general, the best solution is to take the resonant frequency one octave below the critical frequency of the baffle or cabinet. The diagrams so far give no indication as to whether the baffle should be square, or round. In fact, the latter shape is quite unsuitable for baffles. As stated, there is a decrease in response when the wavelength is less than four times the distance from the centre of the speaker to the edge of the baffle. Something of the kind also happens at four times the frequency, that is, when the wavelength of the particular tone equals the above-mentioned distance. With a rectangular or square baffle the centre-to-edge distance is not the same in all directions, whereas with a round baffle it is so, and this results in a very sharp dip in the response curve. Again, with baffles which are not round neither the path-differences nor the associated wavelengths are the same in all directions; consequently the falling-off in response is not concentrated within a limited range of frequencies, as with a circular baffle, but is spread out over a wider range, so that, instead of a sharp dip, it produces a very much less troublesome "dimple" in the characteristic (see fig. 86: dotted-line = circular baffle, chain-dotted-line = square baffle).

The circular shape, then, should be avoided as far as possible, and it is further advisable to mount the loudspeaker on one of the diagonals of the baffle instead of exactly in the middle. Although on the one hand this brings the speaker closer to two of the sides, thus elevating the critical frequency, on the other hand it makes the baffle-dip very much shallower.

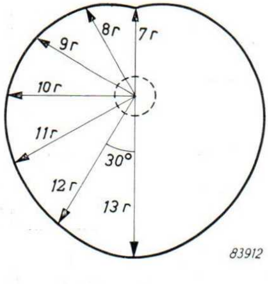


Fig. 87
Heart-shaped baffle
7 r = a in fig. 85

Fig. 87 is a shape with maximum dimensional variation, which smooths out baffle-dip so much that this is almost imperceptible.

Dimension 7r of this heart-shaped baffle equals dimension a as determined from diagram 85. Although the shape of the baffle is pleasing enough in itself, this does not necessarilly mean that it would fit in with any kind of room-furnishing. A shape which is efficient from the acoustical point of view and at the same time more likely to be acceptable to the housewife is shown in fig. 88. To give the best results, this baffle should be used with a 12-inch speaker.

It will be clear, however, that to ensure really good bass response it is necessary to employ baffles very much too large to be accommodated in ordinary houses. Almost the ideal solution is to mount the loudspeaker in a hole in the wall of the particular room. For all practical purposes this is equivalent to providing an infinitely large baffle, and ensures that even the lowest frequencies are reproduced quite faithfully; on the other hand, apart from the fact that the business of mounting the speaker is not easy, wall-mounting is ruled out in many cases because it makes the speaker sound equally loud in two adjoining rooms. It is also necessary to take into account the risk that speakers so mounted will be damaged by draughts or by the slamming of doors; wrapping the speaker tightly in a dust-cover or protecting it with an ample chassis of effective sound-absorbing material lessens this risk, but also elevates the resonant frequency. Accordingly, any method enabling results more or less equivalent to those obtained with large baffles to be procured also with baffles taking up very much less room is highly important.

One such method, employed in radio receivers, is to provide a baffle in the form of a box open at one side; these boxes are usually referred to as "speaker cabinets". In such cabinets dimension a (fig. 77) is the sum of dimensions p and q (fig. 89). As in the case of large baffles, the speaker should not be mounted centrally. Response depends to some extent on the amount of air in, and the dimensions of, the cabinet; if the cabinet is too deep a pronounced resonance peak occurs, usually near the critical frequency, which affects the quality of reproduction. The tone is then "boomy", with the result that speech and song sound unnatural and become almost unintelligible; also, the resonance dominates the reproduction so much that two different bass notes played on a 'cello sound the same, with a pitch equal to that of the resonant frequency. Accordingly, q should not be very much more than half of dimension p. With small cabinets, however, or when the ratios are computed very carefully and verified by experiment, the above rule need not be applied quite so strictly. The performance of a

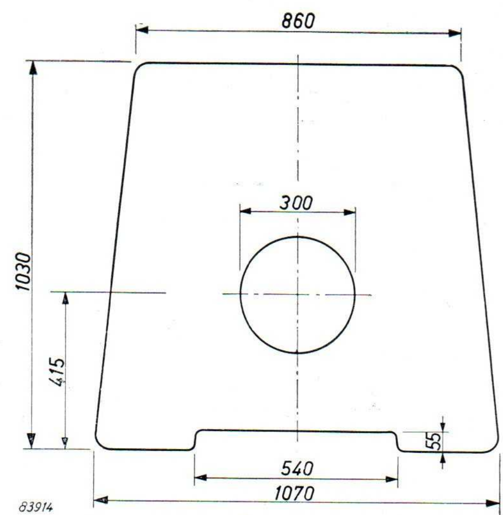


Fig. 88. Trapezium-shaped baffle

3 x 3 x ¾-foot cabinet is the same as that of a 5 x 5-foot baffle. Speaker cabinets should not be closed at the back, otherwise the air trapped inside them acts as a resilient cushion, with resonance at a certain frequency depending on the cabinet-dimensions. This will be seen from fig. 90, showing the response characteristic of a speaker cabinet with the back open (solid line) and with the back closed (chain-line). The critical frequency of this cabinet is 700 c/s, cabinet-dip occurs in the characteristic at rougly 2800 c/s, and the closing of the cabinet results in a very nasty resonance between 300 and 1000 c/s, which also affects the response below this range. When a backplate is necessary to keep the inside of such a cabinet out of reach of the childern, then, it should be perforated, or made of gauze. Also, open-back speaker cabinets should never be placed right up against the wall.

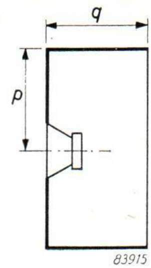


Fig. 89 Open-back speaker

Open speaker-cabinets do not always solve the problem of combining good bass-response with suitability as a piece of furniture; other cabinet designs, however, although they are more difficult to compute and build, offer good bass response without being too large.

In principle the acoustic box is a speaker-cabinet of certain dimensions, provided with a backplate. The difference is that inside it is fitted with panels of sound-absorbing material, to absorb the sound-waves radiated from the back of the cone. If properly designed, acoustic boxes behave in nearly the same way as an infinitely large baffle; in fact, they are sometimes referred to as such. The sound-absorbers are very porous cellulose plates, usually 1 inch thick, fixed to strips of wood half an inch from the outer walls; only the front wall is left uncovered.

An essential feature of acoustic boxes is that the walls are rigid enough to preclude all possibility of sympathetic vibration; therefore the walls must be made of one-inch panels. Also such boxes must not leak, that is, the walls must be close-fitting. This makes the actual construction rather difficult for amateurs. On the other hand the results obtained with acoustic boxes are very good — there is so much speaker-damping that for all practical purposes the peak at fundamental resonance is cut off, enabling the speaker to be operated with very much more power. Given a properly-designed cabinet, the critical frequency will be low enough to make the tone-decrement

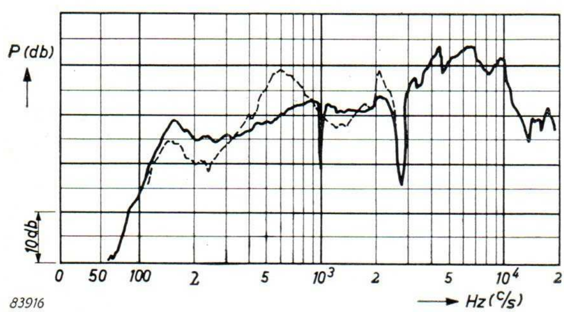


Fig. 90. Sound-pressure curves of speaker cabinet with and without back-plate

below speaker-resonance only 12, instead of 18 db per octave. At the same time there remains quite a lot of distortion below resonance. Fig. 91 shows what kind of results are to be expected (speaker-unit AD 5002). It is worth mentioning, as an indication of suitable dimensions, that acoustic boxes for use with speakers type 9710 or type 9710 M should have a volume of 3.75 cub. ft. for the types 9758, 9760 and 9762 the volume should be 5 cub ft. If more than one loudspeaker is used in the same acoustic box, the volume should be raised proportionally. The shape does not matter much, provided that it is not too tall or too shallow.

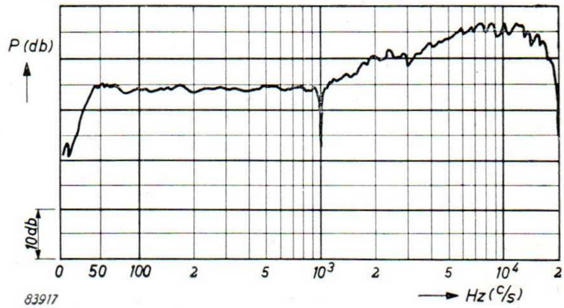


Fig. 91. Sound-pressure curve of speaker-unit AD 5002 (acoustic box with projectors)

Another type of speaker enclosure is the bass-reflex cabinet (fig. 92). This differs from the acoustic box in that instead of absorbing sound waves from the back of the cone, it operates on a principle whereby these waves are utilized. In bass-reflex cabinets the low notes originating inside the cabinet are brought out through a port (1) after a certain delay, as a result of which these waves amplify the direct sound waves from the front of the speaker (2), instead of attenuating them. The delay is procured through the space-effect of the cabinet, as will now be explained. The air in the cabinet is analogous to the capacitor in an electric resonant circuit, and the air-column in the port to a coil, or self-inductance, in a tuned circuit. The combination of the two is an acoustical tuned circuit, so that, given suitable cabinet-capacity and port-area, the resonant frequency of the cabinet will be that of the speaker; the desired delays will then occur at the same frequency, between resonance and the critical frequency. Bass-reflex cabinets give more sound intensity in the bass than acoustic boxes, but only at a price. Firstly, the air-column in the cabinet raises speaker resonance very much higher than it would be in an acoustic box, thus cutting off the lower bass notes; this is the more noticeable owing to the fact that the drop in response below the resonant frequency is then a full 18 db per octave. Secondly, cabinet dips and peaks very soon occur at frequencies just above resonance, and last, but not least, the resonance of the speaker is strengthened, rather than damped, by the resonance of the cabinet. All in all, then, the acoustic box is better, at any rate for quality-reproduction. Although there are many who hold this opinion, others do not share it. Speaker-resonance is more

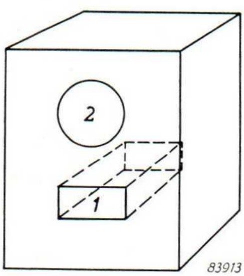


Fig. 92
Bass-reflex cabinet

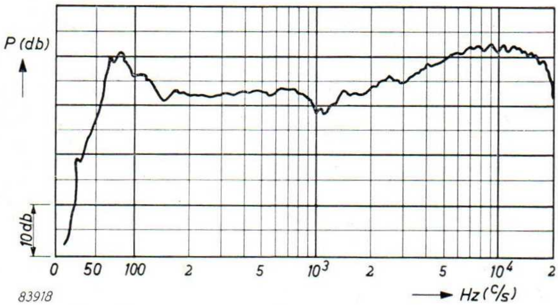


Fig. 93. Sound-pressure curve of bass-reflex cabinet

pronounced in bass-reflex cabinets than in acoustic boxes, where most of it is suppressed by damping, and exactly for this reason the former type of enclosure is preferred in some cases; fig. 93 is the response characteristic of a bass-reflex cabinet dimensioned in very much the same way as the acoustic box shown in fig. 96, and shows the difference quite clearly. Although the opinion here expressed is that resonance of any kind affects the fidelity of reproduction, it is necessary in all fairness to include a diagram (fig. 94) giving optimum dimensions for bass-reflex cabinets.

Such cabinets, like acoustic boxes, should be lined with sound-absorbing material. The "Liveliness" of the reproduction depends to some extent on the type, and the amount, of lining; they are best determined by experiment.

Connecting a pipe (dotted lines in fig. 92) to the port depresses the resonant frequency of the cabinet. As the same effect is obtained by reducing the area of the reflex port itself, it is even better to make this slightly on the large side and provide an adjustable slide to reduce the area of the port until it produces the best possible sound. The port should not be too close to the speaker, reflex cabinets are better large than small, and, as in acoustic boxes, strong wood should be used.

The ideal way of terminating loudspeakers would be to use a simple horn, but this is impracticable owing to the size of horn that would then be necessary for bass reproduction.

Horns to reproduce from 40 c/s upwards are eight feet in diameter and many yards long. Although this problem was solved at least in one instance by placing the speaker in the garage and thrusting the mouth of the horn into the living room, there would be no point in mentioning horns at all if this were the only solution.

Happily it is possible to pack the necessary dimensions into a very much smaller space. Horn dimensions suitable for the living room can be obtained by folding, but even good

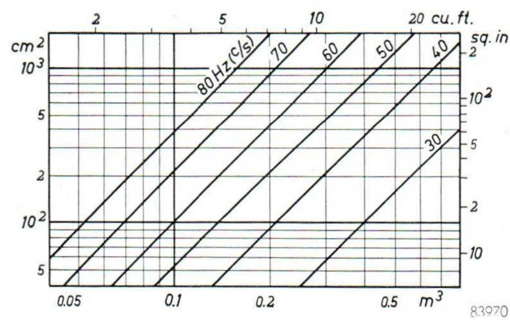


Fig. 94. Diagram for determining cabinet-volume and port-area of bass-reflex cabinets

folded horns are rather on the large side. Fig. 94 shows top and side views (a and b) of a folded horn designed to fit into the corner of a room. Here, the sound vibrations from the back of the cone follow a detour so dimensioned as to give the effect of a horn placed round the walls and projecting into the room, and those from the front of the cone are channelled up or down and then sideways into the same path as the back vibrations. In effect, the walls and floor of the room constitute an extension of the horn, this extension being an essential feature of the system. Practical folded-horn speaker cabinets are more complicated than the diagram suggests, and are therefore rather expensive. On the other hand, the use of the walls of a room as a horn does not cost anything, and this principle can also be applied to bass-reflex cabinets and acoustic boxes, which should then be shaped like a triangular prism to fit neatly into the corner of the room; this not only enhances the quality of reproduction, but also makes the cabinet fit in better with the rest of the furniture. So far we have considered only bass reproduction; the reproduction of high notes presents no special difficulty. Baffles or cabinets for separate treble speakers can be small, and in many cases may be dispensed with altogether.

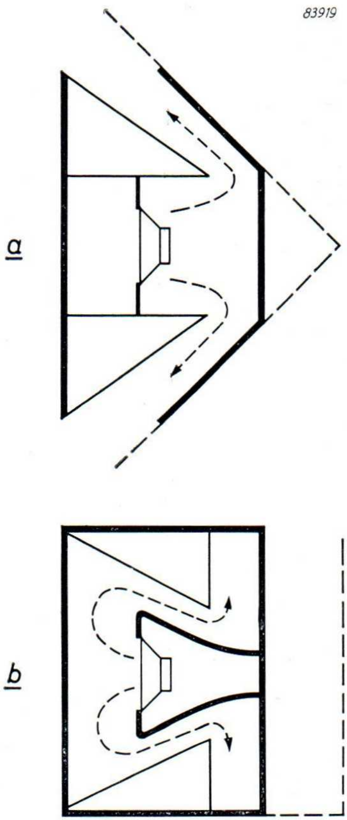


Fig. 95. Cross-section of folded horn: a) Top view b) Side view

Section 6. More than one speaker

The reasons for using gramophones with more than one speaker into the same room are:

a. To increase the range where the main speaker is short of treble response;
b. to reduce Doppler effect;
c. to ensure a certain amount of space-effect;
d. to handle more power.

Not very much need be said regarding **case a;** if the response of the main speaker does not extend much beyond, say, 6000 c/s, an extra speaker is required to reproduce the highest frequencies (tweeter). Although it is not necessary to use baffles at such frequencies, small horns are sometimes employed to increase the directivity of the sound waves, or groups of horns to spread the sound better. In some cases the tweeter is mounted in the throat of the main speaker-cone, and in others it is inserted behind this cone and the sound passes through a duct in the magnet system into a small horn in front of the bass speaker. These are

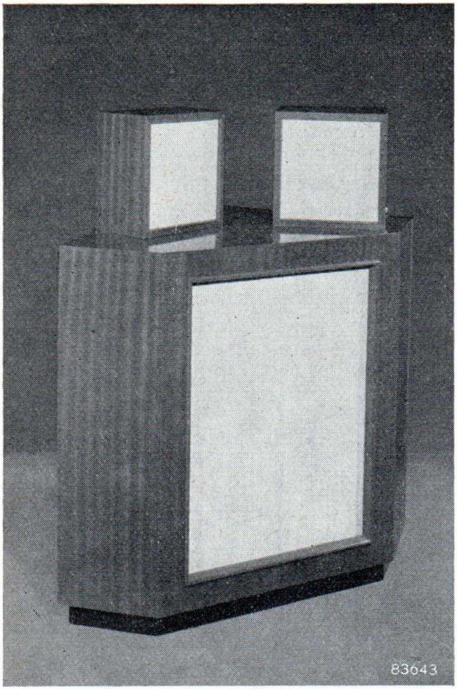


Fig. 96. Loudspeaker-unit type AD 5002

the only two systems in which the tweeter and the main speaker may be mounted in the same cabinet; in all other cases the one should be mounted in a separate enclosure in front of, beside, or above the other. To ensure that the music does not vacillate owing to changes in pitch, the tweeter and the bass speaker should not be placed too far apart. The high and low frequencies are separated and fed to the appropriate speakers by means of a dividing network.

Case b is very much the same, but here cross-over should be at a lower frequency; usually, one loudspeaker is employed to reproduce up to 1000 c/s and another to cover the other frequencies. The treble speaker should be provided with a baffle, though this may be quite small. Also it is necessary that the two speakers operate in phase, or in other words, both cones should move in the same direction, and both change direction with the current. This can be ensured by testing with a torch battery to see which speaker terminals must be touched with the positive pole to make the cone move forward. If the cones move in counter-phase the overall response characteristic will be very irregular where the individual frequency ranges of the cones overlap.

The space-effect referred to in **point c** is now in the centre of interest. Music played in a concert hall loses much of its quality when heard through a key-hole. Because loudspeakers are so much smaller than orchestras, the musical reproduction is very much the same as the original music as heard through the key-hole. This is made less noticeable by employing several speakers spaced in a certain way.

To ensure that the sound pattern will be stable, the response characteristics of all the speakers must be more or less the same. If one speaker reproduces, say, more treble than another, it may happen that a song crescendo towards the high C suddenly shifts from one corner of the room to another, which, if it does not upset the singer, is nevertheless very fatiguing and irritating for the listener. Again, it is necessary that all the speakers operate in phase. That all the speakers should have similar response characteristics applies only to the range above 300—500 c/s.

Further towards bass the ear has very little sense of direction; therefore, one large speaker cabinet for the bass is enough. The others for middle range and treble may safely be smaller. At the same time, dividing networks are necessary to keep the lower frequencies out of small cabinets, otherwise there is a risk of overloading at resonance. Still better results are obtained by mounting the treble speakers in sound-projectors, which beam the sound rather tightly against one of the walls in such a way that it bounces back into the room. The tonal value of electrically reproduced sound is enhanced by diffusion in very much the same way that diffuse light makes a room look cosier. The speakers used with such projectors should be equally efficient also at high frequencies; speakers with straight sound-pressure characteristics are not suitable (see page 111).

For example, fig. 96 shows a combination of speakers designed for this purpose. It comprises an acoustic box reproducing all the tones from 30 to 400 c/s, and two projectors, shown here on top of the box, to cover the range from 400 to 20,000 c/s (see also fig. 91). Properly arranged, the projectors emit a sound pattern several feet wide, in which the sounds of individual musical instruments seem to come from different points and do not shift when the pitch varies.

This also does away with the disadvantage that corner positions are not natural to soloists and orchestras. For psychological reasons it is best that the sound appears to come from one of the short walls of the room.

Again, **case d,** that is the case of several speakers for more power, does not call for very much comment. Usually, similar speakers are used for this purpose, but combinations with either of the other cases are also possible. Where several different speakers are used, however, it is necessary to see that there are no gaps between the characteristics and that the speakers are well-matched as regards efficiency — otherwise the results may be very disappointing. All the speakers must operate in phase, and this must be attended to when the speakers are being connected; it should also be borne in mind, of course, that the connecting of several speakers alters the loading resistance and therefore necessitates a different output transformer, or else another tapping on the existing transformer. The equivalent resistance of n speakers in series when each speaker has a voice-coil resistance of R Ohms is nR Ohms, and with the speakers in parallel R : n Ohms. With two branches in parallel, each comprising two speakers in series, the equivalent resistance stays at R Ohms.

Dividing networks are necessary when the high and low frequenties are fed to separate speakers. These networks are combinations of coils and capacitors, say, as shown in fig. 97 a. Fig. 97 b shows the characteristic of this network. Each branch attenuates current from a certain frequency on, by 6 dB per octave. One convenient feature of such networks is that the input resistance is almost equivalent to the resistance of one speaker, at any rate provided that both speakers have the same voice-coil impedance and that it is more or less independent of the frequency. Taking R as the voice-coil impedance plus the loss resistance of the coil, then, we have:

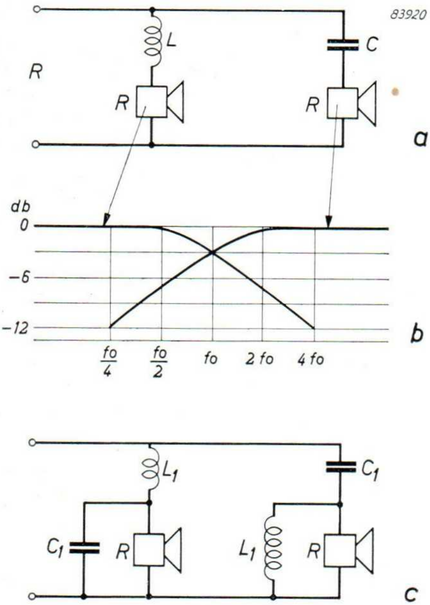


Fig. 97. Dividing network:
a) Circuit for 6 db of attenuation per octave. b) Frequency characteristic of circuit shown in a). c) Circuit for 12 db of attenuation per octave

$$L = \frac{R}{2\pi f_0} \text{ and } C = \frac{1}{2\pi f_0 R}$$

or, in practical terms,

$$L = \frac{159\ R}{f_0} \text{ millihenrys}$$

$$C = \frac{159\,000}{f_0 R} \text{ microfarads}$$

where f_0 is the crossover frequency. Networks whose characteristics fall off more steeply are employed in certain cases; the dividing network shown in fig. 97c gives 12 dB of attenuation per octave. The input impedance of this filter also equals the voice-coil impedance of one speaker. Such networks are used when the crossover frequency is less than 2 octaves from that of the tone control (the crossover of the tone control referred to in fig. 61 and fig. 62 is 1000 c/s — see also fig. 37 and fig. 38). Otherwise it may happen that with, say, the treble control at full boost, too much treble reaches the bass speaker.

The values of the different components are computed by means of the following formulae:

$$L_1 = \frac{225R}{f_0} \text{ millihenrys and } C_1 = \frac{112\,000}{f_0 R} \text{ microfarads}$$

The fall-off should not exceed 12 dB per octave.

Bipolar electrolytic capacitors may be employed, or, at a pinch, conventional electrolytic capacitors with an operating voltage at least 30 times the A.C. voltage to be expected at the loudspeaker.

Ferroxdure-cored coils are used, as far as possible, in commercial networks because they combine smallness with low loss-resistance. On the other hand, air-core coils are best for home-made networks (fig. 98, computed for wire 1.6 mm thick, coils with less than 350 turns a = 1 inch, b = 1 inch, c = 3 inch; larger coils a = 1 inch, b = 1½ inch, c = 4 inch). Badly designed coils with iron cores are a source of much distortion. The networks alter the phase of the current, which has a 90° lead in the treble branch, and a 90° lag in the bass. This must be taken into account when the speakers are being connected; the connections should be made in such a way that without the dividing network the speakers would operate in counter-phase.

Section 7. Listening-rooms

So many books on acoustics have been written that it would take a lifetime to read them all. To avoid adding one more to the list, we shall discuss this subject as briefly as possible.

Despite the abundance of literature on acoustics, it is too often overlooked that even the best of instruments can never sound really well in a room with poor acoustic properties; this applies not only to gramophones, but also to "live" music. As everyone knows there are concert halls with good, and others with bad, acoustic properties; so also with ordinary rooms. One of the more important properties is reverberation-time. The reverberation-time may be defined as the time in which the sound intensity decays 60 dB after the original sound comes to an and. The desired reverberation-time depends on the size of the particular room; for sitting-rooms it should be roughly ½ second. Since it is a rather laborious task to compute the reverberation-time, the ins and outs of it are better explained through analogy, as follows.

The reason that completely white-tiled rooms are not exactly cosy is that the light in them is much too strong; at the other extreme, rooms whose walls, floor and ceiling are painted black are similarly unpleasant. Neither the one, nor the other kind of room would do justice to a work of art exhibited in it.

Cosiness, and a fitting background for beauty, are obtained through suitable colour-schemes.

So also with sound; between bare walls it booms and blares, until the listener gets a headache from the sheer force of it, whereas in too-heavily draped rooms sound becomes thin and flat, and therefore uncompelling. Although a room with two white and two black walls would give an average reflection factor just about right for the amount of light in it, another in which the reflection factors of the individual walls are roughly equal would be very much better. Similarly, rooms in which all the sound-absorbing and sound-reflecting surfaces are distributed more or less evenly are better, acoustically, than others with two bare walls and the rest draped with heavy curtains. Where there is reason to complain that the sound lacks body, then, it is advisable to try leaving the curtains open one evening, and if the music blares too much there is one more argument in favour of buying velvet curtains or a thicker carpet. Although accurate calculation (apart from the price of the curtains) is usually out of the question, those enthusiasts who want to experiment with materials from the store-cupboard may be helped by the following table.

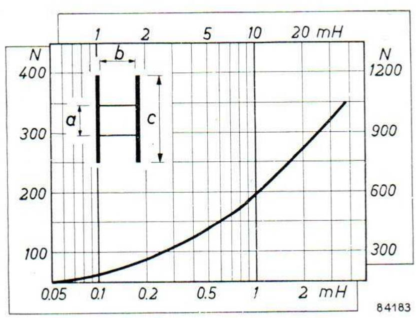


Fig. 98. Diagram for computing coil-dimensions for dividing networks.

Sound-absorption		
Slight	Average	Much
Concrete Brickwork Glass Lace curtains Oilcloth etc. Plastered and draped walls Wood Rubber	Velvet curtains (thin) „Celotex" Paintings Thin woollen blankets Plaids Under-felt Rockwool	Carpet, more than two tenths of an inch thick Cocos matting Velvet curtains (heavy) Cushions Eiderdowns Upholstered furniture Rockwool (less dense) Acoustic „Celotex"

As an example of the effect of such materials, covering the walls of a 20 x 13 x 8-foot room with „Celotex" makes the reverberation period more or less correct, but if the „Celotex" be afterwards painted with water paint, this changes the acoustics completely, extending the reverberation period and making the music sound harsher and more strident. Supposing that „Celotex" plates are fixed to the walls in order to improve the acoustics, and a small amount of space is then left between plates and walls, it may happen that this space acts as an undamped sound-box and therefore affects the acoustics of the room. This can be remedied by drilling a number of holes of roughly ½-inch diameter in the plates.
If the plates are also intended to isolate the room as far as possible from sound coming from outside it, the sides facing the walls should be painted or varnished to enable them to reflect such sound more effectively; of course, drilling holes in the plates is then out of the question. Acoustic „Celotex" and similiar special materials are too absorptive to be used as overall room linings; also, such materials are so fragile that they should be used only where they are out of reach. Painting or varnishing them affects the sound-absorbing properties.

Experience has shown that the bass, only a small proportion of which is absorbed by wall-coverings, usually depends very much upon the type of flooring. Springy wooden floors, particularly when there is a cellar below, give much scope to the bass notes. Sometimes too much, so that the bass drowns the rest of the music. When this happens the speaker cabinet should be placed on rubber feet or on a foam-rubber cushion; in flats, this also benefits the downstairs neighbours.
At the other extreme, stone or cement floors are acoustically bad; in halls with such rigid floors the low notes are usually faint and lifeless; it makes no difference whether they are actually played by an orchestra or merely reproduced. A good deal can be learned by taking the gramophone first into the bathroom and then into a bedroom of roughly the same size; it then sounds entirely different. Such tests

provide information which is useful in assessing the acoustics of sitting-rooms or music-rooms.
The differences in acoustics are brought out still more clearly by tape recordings of the same sound made in different rooms and played back one after the other; such experiments also provide information which can be used to improve the acoustics of listening-rooms.
It goes without saying that background noise should be kept down as far as possible — music never sounds well against a background of rumbling traffic, rattling cups, conversation, and so on; when the general noise-level is high, the highest and lowest notes are the first to be masked by such disturbances. Windows, cups, glasses and trinkets of various kinds vibrating in response to the music are also inconvenient, but here the cause of the noise is easy to find and eliminate.
More difficult to solve, but at the same time very much more instructive, was the case of a gramophone which suddenly, and for no apparent reason, started to emit distortion in the bass. Only after the pickup, the amplifier and the speaker had all been examined, the culprit was found to be a small cabinet, a recent acquisition having nothing to do with the gramophone, which was vibrating in sympathy with certain notes, thereby not only amplifying the notes themselves, but also producing very badly distorted overtones.
As regards the positioning of the speaker we have already seen that the best place for it is in a corner of the room; furthermore, it is advisable to see that the speakers are as far from the listeners as possible. Although this necessitates rather a lot of walking about, at any rate when the speaker is in the same cabinet as the record player or changer, and the amplifier, the effort involved is, after all, only a small price to pay for the very much more natural timbre imparted to the sound by using the reflections and consequent reverberation of the room to full advantage.
With several speakers, one of them in an adjoining room, the best results are obtained usually when the reverberation period of this room is slightly longer, so that less of the sound is absorbed by the walls, than in the actual listening-room.

CHAPTER X

HIGH-FIDELITY — EVALUATION AND TESTING

Section 1. High-Fidelity.

It would be reasonable to suppose from the title of this book that a definition of High-Fidelity is given pride of place in the first chapter. The fact is, however, that no simple definition of the term exists; it is evident from the number of articles already written on the subject that the present-day definition must be very much more complex than the original description of High-Fidelity which came out long before the war, i.e. "Reproduction of all the frequencies up to a minimum of 7500 c/s". Although now obsolete, this was a strictly objective, technical definition. The writer has a subjective definition of his own: "Reproduction giving me as far as possible the same amount of pleasure as would the original music played in a concert hall". As far and away the most important feature of musical reproduction is the music itself, and the technical installation is merely a means to that end, the question of subjective appreciation will be considered first.

It is known that very famous judges of music sometimes own gramophones offering only a very meagre and distorted reproduction of the sound as produced in the studio, and nevertheless derive much pleasure from listening to such reproductions. Their occasional exclamations of disapproval do not refer to the abominable tone of the gramophone, but merely indicate that they do not agree with the particular musical interpretation. Such people listen to music stored in the mind — for them, the sound emitted by the speaker is not very much more than a baton to guide the memory. The writer, who is very far from being a famous musician, requires a lot more assistance from his gramophone — it would suit him best if, apart from the absence of throat-clearing and fidgeting, he could imagine himself to be present at the actual concert instead of listening quietly in his own home.

This does not imply, however, that the requirements of the listener as regards quality of reproduction are inversely proportional to his feeling for music. Far from it. All that is meant is that **some** listeners are the more easily satisfied owing to their vast musical experience, whereas others less experienced compensate for this by striving towards better and better quality in the reproduction. At the opposite end of the scale to musical genius we have the very much larger category of listeners whose musical experience is confined almost entirely to „canned music"; they, also, are very easily satisfied. Another similarity between the two extremes is that neither knows any better, the very musical listeners because they either overlook the technical possibilities or have no interest in them, and the "canned music" listeners because they have never heard music played as it should be.

The writer is only too aware of having presented a rather exaggerated picture of the situation. The essence of what we are now considering is "tone", and this, like music itself, is very difficult to explain; that is why the above strikes a note of assurance which does not by any means ring true in every case. People previously quite content with Low Fidelity do not always react favourably to High Fidelity reproduction on hearing this for the first time. This is only to be expected of those listeners who for various reasons do not attend concerts, or at any rate attend them only rarely — they are unable to recognise true tone. To give an odd example, after a lifetime of hearing nothing but cellos it would take some time to become accustomed to the sound of a violin.

With the musically experienced, things are rather different. They find that reproductions which are only a shadow of the original music do not disturb the process of "listening from memory", whereas an ideal reproduction, like the music in the concert hall itself, gives less scope for such "inner listening".

Here, the music is complete and therefore claims our full attention. The conductor, accustomed to using an orchestra as an instrument, is then confronted with a situation in which the orchestra is not swayed by him and therefore plays just ahead of his thoughts instead of following them. Another difference is that one can concentrate very much more deeply in the seclusion of one's room than in a crowded concert hall.

On the other hand, supposing that a virtuoso listening to a really good reproduction suddenly hears music that he does not already know, or an instrument of exceptional quality — particulary a voice — he will usually sit up and take notice and then becomes extraordinarily critical of the reproduction.

Evidence of the kind of error into which we may fall through ignoring the technical possibilities is to be found in a review of the performance by Grumiaux of Paganini's 4th violin concerto. The unfortunate music critic wrote that this soloist had not yet completely mastered his instrument; in fact, the apparent wrong notes which prompted this remark were due to distortion in the amplifier, not to any lack of skill on the part of the violinist himself. This demonstrates the importance of attending actual "live" performances before writing critical reviews of gramophone music.

Now, the subjective definition already suggested by the writer is not without hidden difficulties. At concerts Grumiaux's Stradivarius sounds different to him than to the audience, because the soloist has the instrument close to his ear, whereas the nearest row of the audience is several yards away from him. So also the microphone, placed close to the violin, picks up and conveys to us through gramophone records the sound as heard by the violinist, and because the listener is not accustomed to such intimacy in the concert hall, it sounds rather strange when brought into the home. This is still more noticeable with orchestral recordings. To violinists in an orchestra the sound of their own playing is barely audible owing to the proximity of other, louder, instruments. The audience, of course, hears the usual "sound at a distance". On the other hand the microphone, provided that it is properly positioned, picks up and brings to the home-listener the sound of violins as the players themselves would hear it but for the surrounding din (if you will excuse the word). Whereas at "live" concerts we hear only one particular violin or piano played, gramophone records may bring us one after another, say, the sounds of a Bechstein, a Pleyel and a Steinway, or a Stradivarius and an Amati; this can be very confusing, the more so for people who do not know which of the instruments is which. Accordingly, High Fidelity can also be defined in another way i.e.: reproduction such as to do full justice to the characteristic timbre of individual musical instruments. With all these different aspects of the matter to be considered, it is very difficult to judge gramophone reproductions by ear; nevertheless, this is ultimately the sole criterion.

Not only the subjective, but also the technical definition of High Fidelity has its difficulties; it, also, is by no means cut and dried. One of the more obvious requirements is that the reproduction be free from distortion of any kind.

Therefore:
1. The response should be uniform throughout the audio-frequency range;
2. the dynamics of the live music should be reproduced without compression;
3. non-linear distortion, and particulary intermodulation and beat-note distortion, should remain below the threshold of audibillity, even during the loudest of fortissimo passages;
4. hum, noise, and so on, should not be noticeable even during pianissimo passages;
5. impulse sounds should be reproduced without distortion, that is, without decay-effects.

It is possible to arrive at a technical definition of "High Fidelity" by establishing certain quantitative requirements for each of the above points.

For example:
1. Response straight to within 2 dB from 16 to 20,000 c/s;
2. dynamic range 60 dB;
3. intermodulation distortion less than 2%, and beat-note distortion less than 0.75%, at maximum sound-intensity;
4. noise level 60 dB below fortissimo passages;
5. equipment free from resonance between 0 and 60,000 c/s; phase-rotations less than 10° from 10 to 20,000 c/s.

This definition is also subjective, however, because we still do not know enough about the relative "nuisance-effect" of the different distortions individually, still less when they occur simultaneously, to say with scientific accuracy what is, and what is not, permissible. There is as much to be said for tightening, as for relaxing the above requirements; every choice within reasonable limits is to some extent arbitrary. This would not matter very much if it were possible to say quite definitely that the equipment giving the best test results also offers the best reproduction; unfortunately, there is no such assurance. Clearly there are still a number of factors which escape present-day tests, even when other properties than those mentioned in the above are also measured.

Such factors are partly psychological and partly physiological. Let us take an example of one of them. During one of our tests a piano was played in the laboratory studio and the music was reproduced in an adjoining room. Although the microphone, the amplifier and the speaker were amongst the best that modern science has to offer, the sound of the reproduction proved rather disappointing. It then occured to the writer to close one ear and bring the other close to the microphone; the live piano-music then sounded very much the same as when reproduced by the speaker. This shows that a single electric ear, the microphone, is not an entirely satisfactory substitute for our two ears. The problem can be solved, although not ideally, by employing two microphones with separate amplifiers and two speakers. Such stereophonic reproduction is better than when only one channel is used, nevertheless, apart from the expense, it is not much use to record lovers, because for all practical purposes none of the records now obtainable are suited to it. Further studio-tests have shown that a very much more faithful reproduction of the original sound is obtained when the single speaker is replaced by a suitable arrangement of several speakers covering different response ranges.

From a strictly scientific point of view the effect obtained by using several speakers with the same amplifier in the same room can be considered as a kind of distortion, but in this case it is distortion which enhances the reproduction. Another example of better quality obtained through distortion is that a more natural-sounding reproduction can be obtained by making the response characteristic not quite straight in the middle register. Other examples are known, and it is probable that there is very much more to be learned about such matters. The moral is that measure as you will, listener tests must be carried out before any definite conclusions can be drawn. Although it is not essential to know very much about music in order to judge by ear, this does require many years of listening-experience, a good technical background and a good ear for tone. Moreover, all this will be wasted if the listener lacks a reasonable capacity for self-criticism to prevent him from being lead astray by, say, an unconscious preference for overemphasis of treble or bass, or bias when judging equipment which he has designed himself. The above will probably disappoint those who have less experience of such matters because it provides no clear answer to the question, what is High Fidelity? At the same time, it explains why certain manufacturers are able to describe inferior equipment as "Hi-Fi" without coming into conflict with the law.

Failing a definition of High Fidelity, however, we have a criterion of it, namely the amount of satisfaction derived from long and attentive listening. Although we cannot claim that listening to gramophone records can be quite as pleasurable as attending live concerts, it is safe to say that the reaction to hearing a recorded symphony, violin solo or song recital is now very much less likely to be "How nice that must have sounded in the concert hall". If your equipment is such that, psychological factors apart, you do not feel very much regret at not having attended the concert itself, then there is every chance that you have yourself discovered the true definition of High Fidelity.

Section 2. Judgement.

Ultimately, all gramophone equipment is judged by ear. Without material for comparison conveniently to hand, this is no simple matter, because our memory for music, although good as regards composition and pitch, is not usually very

accurate as regards tone. Recent tests in the Amsterdam "Concertgebouw" have shown how far the technique of sound reproduction has now advanced.
For example, one of these tests concerned music for two pianos in which one of the parts was recorded and played back during the playing of the second part. None of the musicans, technicians and music-lovers present on that occasion was able to say, without looking, which of the two parts was the recording. Most of us, however, must appeal to our memory as an aid to judgement, firstly because privately-owned musical instruments are usually inferior to those played for gramophone recordings, secondly owing to the difference in acoustics as between sitting-rooms and concert halls, and last but not least, because very few people apart from recording technicians actually have the opportunity of using symphony orchestras as a basis for comparison.
The following list of records may be useful. It constitutes only a small selection from the almost infinite variety of records available, and is not intended as a guide to collectors; Moussorgsky's "Pictures at an Exhibition", for example, should be bought only as an interesting piece of music, not on account of the quality of the recording; nevertheless, owners of this record will also find it useful as a means of judging the quality of their reproducing equipment (A star by the number indicates that the particular record carries more than the one piece of music).

a. Bass response.

Big drums and kettledrums have very low fundamentals. With reproducing equipment whose bass-response is too limited, the bass-drum is not heard at all, or sounds like a rumbling kettledrum, and kettledrums themselves sound as if they are not properly tuned. Bass drums and other large percussion instruments can be heard distinctly in the following recordings:

Moussorgsky-Ravel "Pictures at an Exhibition" (Philips A 00607 R), in the movement "The Gates of Kiev";

Tschaikovsky "Ouverture Solemnelle 1812" (A 00603 R*) Finale;

Beethoven, Ninth Symphony (A 00145/46 L*) between vocal parts of last movement;

Dvorak, Symphony No. V (A 00154 L) at the beginning;

Duke Ellington Uptown (B 07008 L) in "Skin Deep" and the beginning of "The Mooche";

Music by the Marines II (P 10050 R), amongst other things Manoeuvre March.

Some organ recordings also contain very good bass notes, e.g.

Mendelsohn-Bartholdy, Sonata No. 2 (N 00118 L*).

The lowest recorded tone ever heard by the writer is that of the gong at the end of Begdja's "The Gamalan Boy" (N 00165 L); since the fundamental is below 20 c/s, the gramophone cannot be blamed if it does not reproduce this tone.
Dukas, L'apprenti Sorcier (A 00175 L*) contains several very low cello-tones, say, in the part where the broomstick begins to fill the bath; these tones should be reproduced quite distinctly and with noticeable differences in pitch.

b. Middle register.

If the installation reproduces the middle register insufficiently the music sounds flat

and vocal parts give the impression of coming from behind the orchestra. The words of the song then tend to sound garbled, the more so if the lower frequencies are over-emphasized by cabinet or speaker resonance.
When listening to the following recordings, you should be able to follow the words without difficulty (provided that you know the language).
Leoncavallo — Pagliacci (A 01102/03 L)
R. Strauss Salome (A 00163/64 L)
Patachou chante Brassens (P 76010 R)
American Vocal Parade I (B 0764 2 R)
Mascagni "Cavalleria Rusticana" (A 01612/13 R).
Again, this is to mention only a few of the many recordings suitable for our purpose, particularly of popular music.
c. Treble tones.
The violin and many percussion instruments are excellent as a means of judging the quality of treble-reproduction.
Paganini, Violin concerto No. IV (A 00741 R).
The violin should sound lucid, not sharp. Distortion of the higher violin tones is an indication of high-frequency resonance (bad transient-characteristic, see page 87).
Mozart, Sonata for piano and violin, K.V. 301 and K.V. 304 (A 001122).
The same applies. At the same time, the tone of the Paolo violin is not as rich in treble as the "Titian" Stradivarius in the above-mentioned recording, and if the difference is not heard quite clearly this is probably owing to insufficient treble response.
Beethoven, Sonata No. IX (A 01609 R) for piano and violin (Kreutzer Sonata) also includes a clear-toned violin; to ensure good tone balance, the treble reproduction should be cut down slightly as compared with the two last-mentioned recordings.
Kreisler, Rondino on a theme by Beethoven (N 02101 L).
The violin sounds quieter than in the Kreutzer Sonata.
Triangles should be distinct and recognisable in:
Liszt: Piano concerto No. 1 (A 00144 L) 2nd movement
Chabrier: Espana (N 00161 L*)
Dukas: L'apprentie Sorcies A 00175 L*).
Cymbals can be heard in the following recordings:
Liszt: Preludes (A 00144 L*)
Chabrier: España (N 006161 L*).
Brass, loud but not strident:
Music by the Marines II (P 10050 R)
Santorsola: Concertino for guitar and orchestra (N 00626 R*)
Dukas: L'apprenti Sorcier (A 00175 L*)
Moussorgsky-Ravel "Pictures at an Exhibition" (A 00607 R)
Tschaikovsky "Ouverture Solemnelle 1812" (A 00603 R*)
Again, in Benevoli: Mass for 53 Voices (A 00622/23 R), the "Zischlaut" (sharp

German s) should sound distinctly from time to time, whereas any such effect would be quite out of place in Debussy's Sirèsnes (A 00160 L*), where it is an indication of undesired resonance in the pickup or the amplifier.
In dance music the difference between Brushes and Maracas must be clearly audible.

Maximum Power

Fortissimo passages in the above-mentioned recordings of Ouverture Solemnelle 1812, Pictures at an Exhibition, Dvorak's fifth Symphony, and I Pagliacci, as also in Overtures Coriolanus and Egmont by Beethoven (A 00145 L*), "Peter and the Wolf" by Prokofiev (N 02605 R) and many other symphonic works, should be reproduced without distortion. Since it is always difficult to judge the sound-intensity of the reproduction to ensure that this is no too loud or to soft, it is advisable to adjust the volume to a pianissimo passage in such a way that the sound is true pianissimo, and nevertheless loud enough to be heard clearly. (For example, the Catacomb movement in the "Tableau d'une Exposition"). For works containing no marked pianissimo passages, the amplifier can be adjusted to the proper level with the aid of a softly recorded violin or piano solo (e.g. Mozart K.V. 301).

Various musical instruments.

The following records, as well as those already referred to, include purely recorded tones of certain musical instruments.
Piano: Moussorgsky: "Pictures at an Exhibition" (N 00652 R), bass very sonorous.
Rachmaninoff: Concerto for Piano and Orchestra No. 2 (A 00162 L), ditto.
Mozart, Sonatas K.V. 301 and K.V. 304 (A 00112 R).
The tone of the Mozart piano used by Alice Heksch is very much thinner than in the two last-mentioned recordings; the difference should be heard quite clearly in the reproduction. Resonance at low frequencies makes the Mozart piano sound too "heavy", whereas too much cut in the bass brings the grands played by Uninsky and de Groot too close to the tone of ordinary, and Mozart, pianos.
Ravel. "La mère l'oye" (N 00637 R*)
Lavry "Hora" (N 00641 R*) many short treble notes (staccato)
Gould "Interplay" (N 00130 L*)
Liszt, piano concerto No. 1 (A 00114 L*) (A 00200 L*)
Several instruments in "Peter and the Wolf" by Prokofiev, as also in Piano concerto No. 3 (A 00650 R) by the same composer.
Percussion in, amongst other things, Gould's Spirituals (N 00130 L*), in the Dance of the Seven Veils, from Strauss's Salome, in the previously mentioned Ellington Uptown-record, and in Les Elgar's Sophisticated Swing (B 07656 R).
Guitar: Recital, performed by Luise Walker (N 00626 R), including the concertino already referred to, here recorded with the microphone close to the guitar so that the effect of hall reverberation is only very slight.
Clavicembalo — The Bach series, performed by Isolde Ahlgrimm. The tone of this instrument must be quite pure.
'Cello: Bruch "Kol Nidrei" (N 00107 R*)
Dvorak "Concerto for 'cello and orchestra" (A 00687 R).

Clarinet: Mozart "Concerto for clarinet and orchestra" K.V. 622 (A 00698 R).
The New Benny Goodman Sextet (B 07024 L).
Trumpet: Verdi "Gloria All'egitto" (Aida) (S 06018 R*).
Harry James "Soft lights, Sweet trumpet" (B 07603 R).

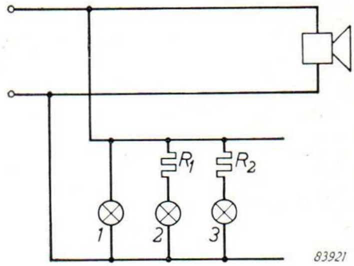


Fig. 99. Simple output-meter
1, 2, 3 = 2.5 V 0.1 A lamps
R_1 = 3,75 Ω, 1 watt
R_2 = 9 Ω, 1 watt

Section 3. Tests on gramophones

Many tests can be carried out on reproducing equipment, but with only one exception they require so much technical knowledge as to be suitable only for specialists. Moreover, good measuring equipment is far too expensive for the amateur. The exception is the plotting of frequency characteristics; it requires only a small amount of technical know-how and a minimum of equipment. The method is to place a frequency-test record on the turntable and measure the voltage across the speaker by means of a valve-voltmeter. Long-playing frequency-test records ($33^1/_3$ r.p.m.) cost about the same as classical recordings, but the less expensive 45 and 78 r.p.m. test records are just as useful to the amateur.

Valve-voltmeters are rather expensive, and most of the other A.C. meters, apart from the fact that they are not cheap, are quite unsuitable for the purpose.

Fig. 99 is the circuit of a test-meter suitable for amateur use, which should be connected in parallel with the speaker. This meter contains three lamps in parallel, two of them provided with ballast resistors. When 0.6 V is applied to such a circuit, lamp one glows faintly, producing roughly the same amount of light as a red-hot nail. At 0.85 V lamp 2 glows faintly and at 1,2 V lamp 3; the other two lamps then burn fairly brightly.

As the ratio of 0.6 to 0.84 V and of 0.84 to 1.2 V is 1 : 1.4 equivalent to a difference of 3 db, the instrument can be used to plot frequency characteristics. The first step is to lower the pickup on to the 1000 c/s band of the test-record. We then connect the meter to the speaker and turn the volume up until lamp 2 begins to glow faintly (lamp 1 is then burning brightly). Next, we move the pickup back to the start of the record. If lamp 2 goes out and lamp 1 emits only the faintest glow at, say, 400 c/s, this indicates a fall of 3 db in the characteristic at 400 c/s. Again, supposing that lamp 3 begins to glow at 2000 c/s, we have a rise of 3 db in the characteristic at this frequency.

Of course, the differences may well exceed 3 db. Keeping to the same examples, we then begin by plotting the part of the characteristic between 400 and 2000 c/s. Having done so, we return the pickup to 400 c/s and adjust the volume until we see a faint glow in lamp 2. We then plot the part of the characteristic covering the range below 400 c/s, drawing it as an extension of the part already plotted. Supposing that lamp 1 dims until it is almost out at 200 c/s, then the characteristic is 3 db lower at this

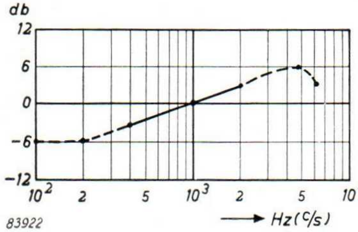


Fig. 100. Frequency characteristic plotted with the aid of the simple output-meter

frequency than at 400 c/s, or 6 db lower than at 1000 c/s. If the brightness of the lamp does not vary from 100 to 200 c/s, the part of the characteristic between these frequencies is flat. At 2000 c/s we turn the volume down until lamp 3 goes out and only a faint glow is seen in lamp 2. We may find, say, that lamp 3 begins to glow again at 4500 c/s, indicating that the characteristic has then risen 3 db as compared with 2000 c/s, or 6 db as compared with 1000 c/s. If lamp 2 dims again at 6000 c/s, the level is the same at this frequency as at 2000 c/s, and so on.

Although all this may seem rather complicated, experience has shown that after a certain amount of practice it is quite possible to plot response characteristics quickly and with reasonable accuracy, and to make the necessary volume-adjustments for differences greater than 3 db without interrupting the test.

To avoid lamp-failure owing to sudden overloads caused by switch surges, it is advisable to switch on the amplifier and lower the pickup on to the moving record before connecting the meter, and to disconnect this before raising the pickup. Also, we should see that the volume is turned right down before connecting the lamps. As it is not always easy to obtain the necessary resistors for this circuit, the alternative circuit shown in fig. 101 may be preferred, although this is not quite as accurate. Here, lamps 2 and 3 light up whenever the speaker-signal rises 3 db above the voltage producing a glow in lamp 1. As with the other meter, we begin the test by lowering the pickup on to the 1000 c/s band of the test-record and bringing the volume up so that first lamp 1, and then lamps 2 and 3, show a faint glow. Next, we centre the volume control as accurately as possible between the two positions. Lamp 1 remains alight, but lamps 2 and 3 should then go out. As long as the lamps remain in this condition, we know that the output voltage is constant to within 3 db above and below the level to which is has been adjusted; also, we can tell from the amount of light radiated by lamp 1, whether the speaker voltage is more, or less than the voltage at 1000 c/s.

The test results depend not only on the pickup, the amplifier and the position of the tone-control, but also on the particular test-record.

The recording characteristic of such records is usually shown on the label. Where it is not, the needle velocity (see page 30) can be computed from the "lightband width", that is, from the width of the bright band of light seen on the record when this is illuminated from an acute angle (see fig. 102).

The width of the light-band is measured by means of a ruler or sliding callipers, the observer looking squarely at the portion to be measured, that is, at W_1

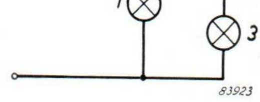


Fig. 101
Simple alternative to fig. 99

of the band nearest to the light source and at W_2 of the band on the other side. The average band-width is then :

$$W = 2(W_1 \times W_2) : (W_1 + W_2).$$

Having established this average, we compute the needle-velocity by means of the formula: $V = (W \times N) : 19$, inch/sec (W in inches; N is the r.p.m. of the particular record).

Separate calculations are necessary for each frequency-band of the record. This method gives very accurate results, provided that the width of the light-band is measured precisely, — record manufacturers use measuring microscopes for this purpose. Another condition is that the light-rays falling on the record must be parallel; it is satisfied as far as possible by employing the smallest light source suitable for the purpose, placed not less than 3 feet from the record. Light-coloured, reflecting walls act as secondary light-sources and therefore affect the accuracy of the measurement. Fig. 103 illustrates the principle of this method. Here, the vertically-reflecting parts of the groove are indicated by short transverse lines; all the other parts reflect more or less obliquely. The larger the amplitude, the wider the zone reflecting light vertically into the eye of the observer, ((a) large, (b) small amplitude), and for an unmodulated groove (c) vertical reflection is confined to one point. The arrows under the diagram show the direction of the incident light. On unmodulated parts of the test-disc, then, the light-band is normally very narrow, but if the groove is not smooth, and therefore causes a certain amount of record noise, such parts will also show a fairly wide band. In theory, the maximum permissible light-band widths are 1 mm or 0.04 inch for 78 r.p.m., 1.4 mm (0.06 inch)) for 45 r.p.m., and 2 mm (0.08 inch) for $33^1/_3$ r.p.m., records.

In practice, however, the apparent width is often very much more, owing

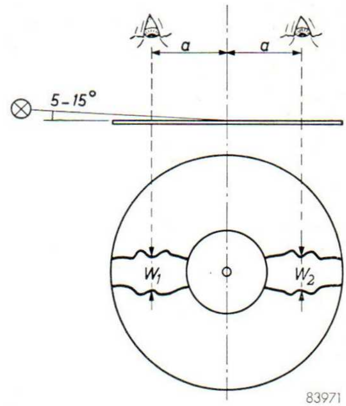


Fig. 102. Measuring the light-band width of frequency-test records

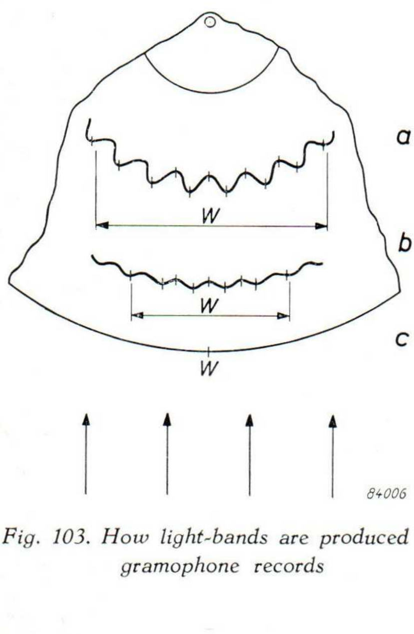


Fig. 103. How light-bands are produced on gramophone records

to the fact that amateurs, at any rate, cannot always ensure that the incident light-rays are exactly parallel.

A simple test to ensure that the tone-arm moves lightly enough in its bearings and is properly balanced, instead of over-balanced, is to place a 45 r.p.m. record with a large centre-hole on the turntable so as to be as far off-centre as possible, using the spindle for $33^1/_3$ r.p.m. records.

When once the pickup is lowered onto the record, the needle should not jump out of the groove, even when the record is played at 78 r.p.m.

CHAPTER XI
MAGNETIC TAPE RECORDINGS

Section 1. Basic principles

Sound was first recorded and reproduced magnetically towards the end of the nineteenth century, but the method did not become widely known until after 1940. The magnetic recorder was invented in 1898 by Valdemar Poulsen, but for the next fifty years it remained very much inferior to the gramophone as regards quality of reproduction; also, the equipment, and particularly the recording material, was very much more expensive than that used to make and play gramophone records. At the same time, the scientific and technical experiments of that period were not wasted, and because of them it is now possible to make very good magnetic sound recordings at reasonable cost.

That the development of magnetic recording took so long suggests that the principles are more obscure than those of the mechanical process used in the making of gramophone records. So they are; in fact one of the most important stages in the process of recording, namely the high-frequency pre-magnetization of the tape, has still to be explained in a manner acceptable to everyone. On the other hand, the basic principles are not very difficult.

When a piece of steel is inserted in a coil carrying direct current, it becomes permanently magnetized. The magnetic force then depends not only on the properties of the steel and the number and size of the turns in the coil, but also on the current in the coil.

If a piece of steel thus magnetized is moved past or through another coil, it induces a voltage in this coil. It will be remembered that magnetodynamic pickups, as described on page 27, operate on the same principle. The voltage depends, amongst other things, on the power of the magnet.

Fig. 104 shows how the two properties, magnetisation and induction, can be used to record and reproduce sound. Here we have a coil (1) connected across an amplifier (2) to a microphone (M). The coil contains a soft-iron core with soft-iron pole-pieces at both ends.

The microphone intercepts sound vibrations and converts them into A.C. voltages, which are amplified to produce alternating current in coil 1. As a result, magnetic flux alternating in frequency with the sound vibrations, and proportional in strength to the sound-pressure on the microphone at any given moment, occurs in the air-gap between the two pole-pieces. Now, steel wire is pulled past the air-gap at a fairly fast rate, and the part of the wire opposite the gap at any given moment is magnetized by the field produced by coil 1.

The direction and amount of magnetization of the wire depend upon the direction and strength of the current in the coil. Given a purely sinusoidal current in the coil (see fig. 8), the first length of wire to pass the air-gap will not be magnetised, the

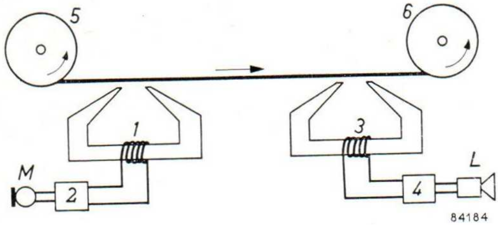


Fig. 104. The principle of magnetic recording

next will be slightly magnetised, the next more so, and so on, the magnetization gradually increasing to a certain maximum and then tapering-off again to zero. As more wire is pulled past the gap, the magnetization again increases, but this time towards a maximum in the opposite direction to that of the first magnetization. Having reached this maximum the current and the magnetization both decrease, and both reverse direction again at zero. Thus the variation of the current is recorded on the steel wire as zones of different magnetization.

Having passed coil 1 the magnetized wire proceeds to coil 3, which is likewise provided with a core and pole-pieces of soft iron. In them it produces a flux very much weaker than that orginally produced by coil 1, but varying in exactly the same way. In effect, then, the field in the core of coil 3 is a replica on a very much smaller scale of that originally produced by coil 1 — albeit slightly delayed owing to the wire having to pass from 1 to 3. The alternating flux generates in coil 3 A.C. voltages proportional to the field-strength at any given moment, therefore, continuing the argument as before, it can be shown that these voltages, although very much weaker, are in every other respect identical with the A.C. voltages originally applied to coil 1. These weaker voltages are amplified (4) and applied to the speaker (L), which then emits an exact reproduction of the sound vibrations intercepted by the microphone.

The wire retains the different zones of magnetization after it is wound on the take-up reel (6), so that when it is re-wound on the supply reel (5) and then pulled forward again past the pole-pieces of coil 3, the speaker reproduces the same sound, however long it is since this was recorded.

Such recorders are quite practicable, but have many disadvantages, e.g. poor sound-reproduction owing to severe distortion, high noise-level and a notable lack of treble response; also, the sound is not entirely free from wow and flutter.

Let us consider the factors affecting treble response. One period of a tone at, say, 10000 c/s is 0.1 millisecond. The current in coil 1 reverses after 0.05 milliseconds, that is, halfway through this period. Accordingly, a certain part of the wire is magnetized in the one direction during the first half-cycle, and the next part is magnetized in the other direction during the second half-cycle. The first part of the wire must clear the air-gap before the start of the second half-cycle, otherwise it will be demagnetized. It will be evident, then, that either the wire must be pulled past the air-gap very quickly, or the gap itself must be very small.

Supposing that the speed of the wire is $7\frac{1}{2}$ inches per second, the amount of wire pulled past the air-gap in 0.05 millisecond will be 0.00005 x $7\frac{1}{2}''$ = 0.000375 inch, that is, 0.0095 mm or 9.5 microns; this imposes a maximum limit of 10 microns on the air-gaps of the recording and reproducing coils.

Even with so small an air-gap, however, the reproduction of 10,000 c/s notes leaves much to be desired, for reasons which will now be given. Firstly, it is difficult to guide steel wire close to the pole pieces; secondly, the fast-moving wire soon cuts grooves in the poles, which is bad in itself and also involves a risk that dust from the grooves will block the air-gap. Thirdly, steel wire does not "take" high-frequency signals very well, even when its speed is increased to 3 ft/sec or more.

The use of wire also involves certain mechanical difficulties. The wire is magnetized mainly at the side facing the pole-shoes. If it happens to twist slightly during a playback, the zone of magnetization presented to the air-gap of the reproducing coil will be very much weaker; therefore the particular tone will be attenuated. As the position of the wire is rather indeterminate, the sound-intensity also tends to vary more or less at random. If the wire is played back at a speed different from that used in the recording, there is also a difference in pitch between the sound as reproduced and as recorded. In the recorder of fig. 104, reel 6 revolves at a constant speed, therefore the wire-speed increases as this reel fills up. Since this happens during recording as well as during playback, it does not matter very much in itself; but if the wire is shortened after the particular recording, it afterwards passes the reproducing coils at less than the original recording-speed and therefore alters the pitch.

Still more serious is the fact that the wire never winds perfectly smoothly on to the reel, and therefore moves rather jerkily; such speed-fluctuations cause wow and flutter.

It is also worth mentioning, as a final drawback, that if the spring-steel wire develops a kink and therefore breaks after several playings, it is apt to fly yards across the room.

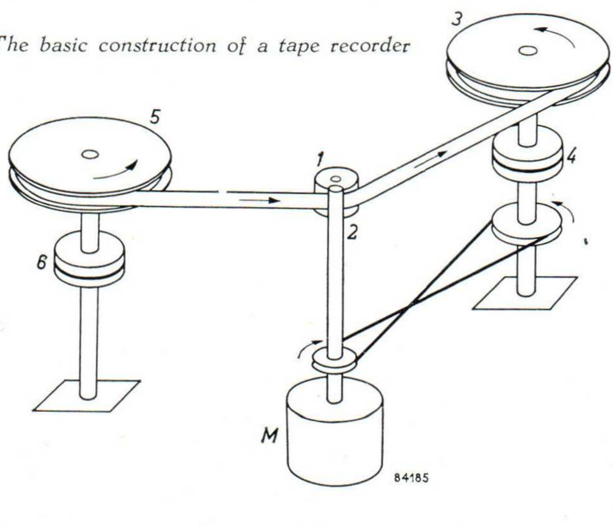
Section 2. Tapes and sound-recording

Many of the above-mentioned difficulties are avoided when steel tape is used instead of steel wire. Such tape does not twist and therefore ensures very much more uniform tone-reproduction; also, it runs more evenly past the pole-pieces, thus enhancing treble response, and being flat, is relatively easy to drive at constant speed, as will be seen from fig. 105. Note that the tape is pressed against the driving roller or capstan (2) by a pad roller (1); this ensures that it moves at constant speed from right to left, at any rate as long as the speed of the motor (M) remains constant. Motor M also drives the take-up reel (3), by means of pulleys and a cord. Reel 3 must rotate just fast enough to keep the tape taut, therefore the speed of this reel must be reduced gradually as it picks up more tape, otherwise the pull on the tape will become excessive.

This is taken care of by a slip coupling (4) between the driving wheel and the reel; thus the pull on the tape is maintained constant. To avoid any slackening-off in the

left-hand part of the tape, the supply-reel (5) is controlled by a friction clutch(6). In principle, all modern tape recorders are similarly constructed.
In many respects, then steel tape is better than steel wire; on the other hand it has two drawbacks, namely that it is too expensive for general use and that it does not give enough improvement in treble response.
All the same, the tape itself, is not the only cause of poor treble response; the recording and reproducing heads are also to blame. Alternating current in the recording coil produces alternating flux in the soft-iron core. Since this core is conductive, the flux induces eddy currents in it, and owing to the resistance of the core these currents cause losses which reduce the flux in the air-gap. This effect is worst at high frequencies. To lessen it, the core and the pole pieces are not made out of solid iron, but are built by stacking thin, soft-iron plates with layers of insulating material between them to prevent eddy currents. Another point to be considered is the quality of the iron. In ordinary iron the magnetic losses are still too high, therefore special nickel-iron alloys must be used to ensure good treble response.
Of course, eddy currents are not confined to the core, but may also occur in the steel tape. Still more, however, the recording-intensity of high-frequency signals depends upon the effect which will now be described. Experience has shown that when a large piece of steel is inserted in a coil carrying direct current, it takes several seconds to become fully magnetized. This is because steel resists the magnetizing force, and the larger the piece, the longer it is able to resist. With very small pieces of steel, the delay is very short, and the time to reach maximum magnetization is only of the order of one two-thousandth of a second (½ millisecond).

Fig. 105. The basic construction of a tape recorder



Even so, 0.5 millisecond is too long when the steel is used for sound recordings; as one half-cycle of a 5000 c/s tone lasts 0.1 millisecond, it only allows enough time to partly magnetize the particular length of steel tape, and tones at higher frequencies make no impression at all.

To be ideal, the magnetic sound-carrier should be inexpensive and should not cause eddy-current or other losses; also, the delay effects should be negligible. Moreover, such tape should be thin and flexible, and nevertheless mechanically strong; it should not stretch, and when once magnetized it should retain this magnetism undiminished for a long time.

The mechanical requirements are satisfied by certain plastics, and the magnetic and electrical requirements by certain iron compounds, of which Fe_2O_3 (ferri-ferrite) and Fe_3O_4 (ferro-ferrite) may be mentioned as being best suited to the purpose. These ferrites are very much like rust in that they are both reddish-brown powders, but, unlike ordinary rust, they can be magnetized. Accordingly, magnetic-recording tapes are made by coating one side of a strip of plastic with a mixture of very finely divided ferro-ferrite and a binding agent.

This coating, only 15 microns (0.0006-inch) thick, is polished to enable the tape to slide smoothly along the recording and reproducing heads without wearing them noticeably and without leaving too much space between the air-gap and the tape-surface. That the contact between playback head and tape must be very close will be evident from the fact that with an air-gap in this head of 7 microns, a space of 12 microns (about 0.0005 inch) from head to tape is enough to cut-down the output voltage of the head by more than 30 dB.

The ferrite granules must be extremely small, otherwise (as in the case of gramophone records) they cause noisy reproduction. With good tapes the grain diameter is less than 1 micron; the overall tape-thickness is usually 55 microns, although nowadays very much thinner tapes are also obtainable. The width is ¼ inch. Tapes in which the ferrite powder is worked as far as possible evenly into the plastic instead of being carried as a coating are also sometimes used. Although plastic tapes are very much better than steel ones, they do not entirely eliminate the drop in response at high frequencies. Because the magnetized zones are very short (only 10 microns in the example given in Section 1) they tend to demagnetize themselves. This makes the signal strength of the highest tones even weaker than when first recorded.

Section 3. Distortion and high-frequency pre-magnetization. (Magnetic bias)

So far we have discussed only linear (frequency) distortion, leaving, non-linear (amplitude) distortion out of consideration. Up to a point, linear distortion in tape recordings is analogous to playback loss in gramophone records (see page 42) and to the other loss caused by elastic deformation of the records during playback. Non-linear distortion in tape recorders occurs mainly during recording, and in this respect is the opposite of disc tracing-distortion, which occurs only during playback.

We refer, of course, only to the fundamental causes of distortion, not to distortion in the amplifier or owing to the use of bad-quality tape or inferior recording and reproducing heads. To understand what causes recording-distortion in tape recorders, we have to know how magnetic materials are magnetized.

Fig. 106 shows the variation of the magnetization of a length of tape with the current in the recording coil. It is seen that magnetization takes place gradually up to about 0.5 mA, rises sharply from 0.5 to 2 m.A. and does not increase very much towards higher current-values. If the current is switched off on reaching 3 mA, the flux density drops; not, however, to zero, since we are considering what is known as a permanent magnet, but instead to 26 gauss. Supposing that the current is cut off at 1.5 mA, the remanence, or remaining magnetism, will be 16 gauss. Fig. 106b illustrates the variation of remanence with current-strength. It is evident from the fact that this diagram is curved, that the recording of vibrations on magnetic tape cannot take place without distortion. In fig. 106b we have (bottom) a sinusoidal current, flowing in the recording coil, and (right) the variation in tape-magnetization with this current. The distortion is made up of odd harmonics, the third harmonic predominating. If the magnetization on the tape is a distorted recording of the original signal, it will, of course, generate a similarly distorted voltage in the reproducing head. Something of the kind happens in all magnetic materials, and although it affects some less than others, the resultant distortion is invariably too noticeable for really good-quality reproduction. The only time that the distortion is small enough to be ignored is when the magnetization is very weak, and unfortunately this does not help us much, because the output voltage of the reproducing head is then so small that for all practical purposes the desired signal is drowned by background noise.

A method of reducing such distortion was discovered in America as long ago as 1922, but was then more or less forgotten; although re-discovered in Japan in 1938, this method of reducing distortion, by means of high-frequency pre-magnetisation, was not finally adopted until 1941, this time in Germany.

Fig. 107 gives a rough idea of the way in which H.F. pre-magnetization reduces distortion. It shows the magnetization curve obtained when the current in the coil,

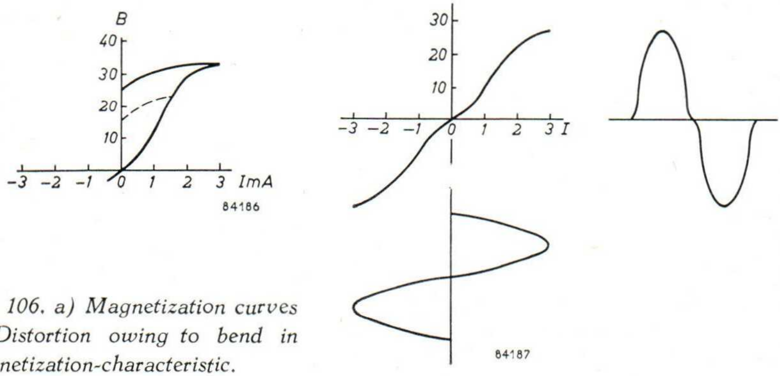


Fig. 106. a) Magnetization curves b) Distortion owing to bend in magnetization-characteristic.

after having been 3 mA, is brought back to zero, then reversed, and, having returned to 3 mA, is brought back to zero again, then reversed again, and so on (loop PQ). Loop RS shows what happens when the current is varied between + 1 mA and — 1 mA instead of + 3 mA and — 3 mA, and the smaller loop, inside RS, represents the variation in magnetization caused by varying the current between + 0.6 mA and — 0.6 mA. Supposing that the coil carries alternating current, the large loop, PQ, is the variation of the magnetization with current having an amplitude of 3 mA, RS is the variation with 1 mA A.C., and the smallest loop shows what happens at 0.6 mA. As the alternating current is reduced gradually from 3 mA to 0 mA, the magnetization loop shrinks to a point on the vertical axis, and the tape becomes demagnetized.

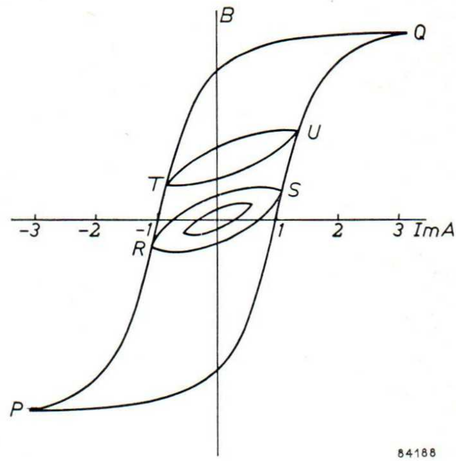


Fig. 107. High-frequency pre-magnetization

Supposing that not only 1 mA A.C., but also 0.4 mA D.C. is passed through the coil, the magnetization loop shifts, to become TU (fig. 107). Because the sides of the large magnetization loop, PQ, are almost straight, the displacement of the smaller loop is for all practical purposes proportional to the direct current. Now, if the direct current and the alternating current be both reduced, the one as much as the other, loop TU shrinks; when the total current, A.C. and D.C., is zero, all that remains of TU is a point, which, because there is no longer any current, must be located on the vertical axis, namely at the intersection of this axis with a straight line joining T and U. The displacement of the magnetization loop being proportional to the direct current, the residual, or remanent, magnetism is likewise proportional to the direct current which has passed through the coil.

As we have seen, in tape recorders a magnetic tape travels past the recording head. If at a given moment 1 mA A.C. and 0.4 mA D.C. be passed through this head, the part of the tape exactly opposite the air-gap at that moment will be magnetized according to loop TU (fig. 107). In the next moment, the tape moves on to bring the particular part halfway beyond the air-gap, where the field from the recording coil is weaker, and the associated magnetization loop smaller; in another moment the whole of the part referred to moves so far from the air-gap that the magnetization loop shrinks to a point, leaving this part of the tape with only its remanent magnetism. As explained, the amount of remanent magnetism is proportional to the amount of direct current in the coil.

Now, provided that the frequency of the pre-magnetization alternating current is very high, say, 100,000 c/s, a low-frequency alternating current may be substituted for the direct current. We have seen that good treble-response necessitates an air-gap small enough to ensure that the signal current in the coil does not vary appreciably in the time that a given part of the recording tape is in front of this gap. What happens after that time does not matter very much, as by then the particular length of tape has left the air-gap. Accordingly, the remanent magnetism of a given part of the tape depends entirely upon the instantaneous value of the A.C. signal at the moment when this part of the tape is in front of the air-gap, as the H.F. pre-magnetization current ensures that the remanence is proportional to the instantaneous signal.

Admittedly all this is by no means simple, nevertheless a full explanation of the effects referred to would be very much more difficult to understand; even without going any deeper into the matter, however, it will be clear that the frequency of the pre-magnetization current is not very critical. What is important is that this frequency should be well above the highest frequency recorded; at the same time, it should not be too high, otherwise it will be difficult to pass the necessary current through the recording coil; also, there is a risk of interference with radio reception. The strength of the pre-magnetization current is also important. Although it would be reasonable to suppose that the optimum current can be computed from the diagram in fig. 107, such calculations are only very approximate owing to a difference in magnetization as between the upper and lower parts of the magnetic coating on the tape — all that it is safe to say is that the pre-magnetization current should be of more or less the same order of size as the highest permissible signal current. Subject to this condition, the distortion can be held below the threshold of audibility.

It often happens that the particular recording is required for only a limited time, therefore it is ncessary to have some means of erasing recordings from the tape so that this can be used again. As explained in connection with H.F. pre-magnetization, a piece of magnetic material in a gradually fading alternating field loses its magnetic properties the moment the field fades out. So also with material already magnetized, at any rate when the original alternating field is stronger than the remanent magnetism in the particular material and, of course, provided that no other current passes through the coil during the time that demagnetization is taking place. Tape recordings can be demagnetized by taking them past the air-gap of what is known as an erasing head, which carries a suitable high-frequency current. Such heads are usually positioned in front of the recording head to remove all traces of any previous recordings, thus ensuring that the new recording is not marred by interference "out of the past". Again, the exact frequency of the erasing current does not matter much, provided that it is high enough.

In most cases this current and the H.F. current for the recording head are supplied by the same oscillator; here, the exact current strength is not very important; the number of ampere turns is usually larger than in the recording head. Another way of erasing recordings is to run the tape past a permanent magnet. Although less expensive than with H.F. current, this method is less popular because usually it does not erase the recording completely, and therefore leaves a certain amount of interference on the tape.

Section 4. Tape recorders - Different models and characteristics

Tape recorders, most of which are designed to reproduce as well as record, fall into three classes:

1. Professional — the best, for use in studios.
2. Semi-professional — not up to first-class standards, but nevertheless very good.
3. Non-professional — to satisfy more moderate requirements, say, in the home.

The separation of these three classes is extremely vague, and the division between classes 1 and 3 is becoming smaller and smaller; this is only to be expected seeing that the best tape recorders are so close to perfection that is is difficult to find scope for improvement, whereas the cheapest class can be improved very much more readily. Consequently, it is difficult to draw a clear dividing line between classes 1 and 2 on the one hand and classes 2 and 3 on the other.

In professional tape recorders separate erasing, recording and reproducing heads are employed (fig. 108a), whereas in non-professional equipment one head is used as recorder and as reproducer (fig. 108b). The advantage of the former arrangement is that it enables the sound to be monitored continuously during recording; tape recorders in which one head is employed first to record and then to reproduce do not offer this facility, but have the advantage of being less expensive. To the amateur, moreover, monitoring is often more of a liability than an asset, since it is apt to cause sympathetic vibration during recordings made with the aid of a microphone.

The main difference between the three classes is the speed at which the tape runs past the heads. From the example given in section 1 of this chapter, the relationship between the size of the air-gap and the frequency at which a drop of 6 dB would occur in the response is:

$$\text{Frequency} = \frac{\text{Tape speed}}{2 \times \text{gap length}}$$

(Tape speed in inches/sec. size of gap in inches)

Values computed by means of this formula are given in the following table, from which the frequency at which the response characteristic drops 6 dB can be read.

Size of gap	Tape speed			
μ	9,5 3¾	19 7½	38 15	76 cm/s 30 inches/sec
7,5	6300 c/s	12600 c/s	25200 c/s	> 30000 c/s
15	3200 c/s	6300 c/s	12600 c/s	25200 c/s
30	1600 c/s	3200 c/s	6400 c/s	12600 c/s

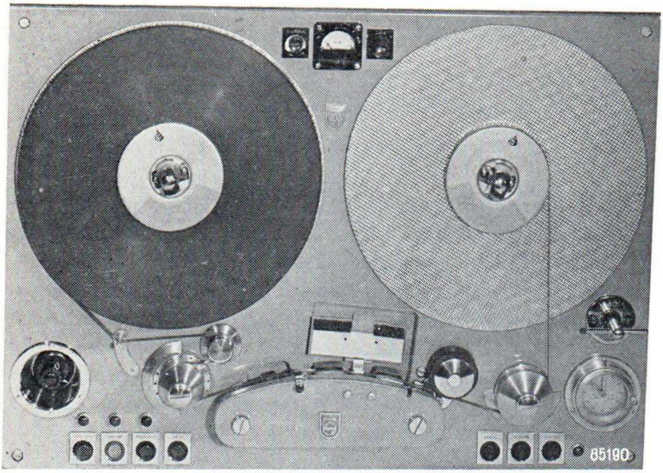
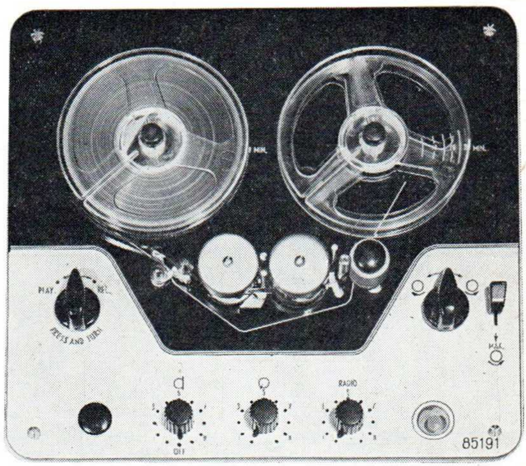


Fig. 108
Professional and non-professional tape recorders



Within limits, then, the smaller the air-gap of the recording and reproducing heads and the higher the tape speed, the better the treble response. Treble boost in the recording and playback amplifiers keeps the overall characteristic almost straight at higher frequencies, but it is not practicable to go more than half an octave above the frequencies given in the table. This will be evident seeing that, again from the example in section 1, doubling the frequency taken in this example would mean that the part of the tape magnetized during the first period does not have time to clear the air-gap before the start of the second half period, and is therefore demagnetized. It follows that at twice the frequency for a playback loss of 6 dB the tape is no longer magnetized.

Apart from the practical difficulties involved, there is no point in making the air-gap too small, since a certain amount of cut in treble response owing to self-demagnetization of the tape will remain.

Studio recorders are therefore provided with relatively large air-gaps and operated at the highest tape-speeds; non-professional tape-recording equipment, on the other hand, operates with a lower tape speed ($7\frac{1}{2}$ or $3\frac{3}{4}$ inches per second, dropping to $1\frac{7}{8}$ inch per second for speech recordings), because this is less expensive, and the smallest of air-gaps.
The choice of tape speed affects the price in two ways.
The tape-length to give 32 minutes of uninterrupted playing is 200 yards at $3\frac{3}{4}$ inches/sec, or 400 yards at $7\frac{1}{2}$ inches/sec. The tape is expensive enough to encourage economy in the use of it. Another item, affecting the price of the recorder itself, is the tape-drive. Tape 200 yards long comes on reals 5 inches in diameter, and tape twice this length on 7-inch reels; the fastest recorders require 8-inch, or still larger reels. It will be evident that the larger, and heavier, the reels, and the faster the tape speed, the greater the load on the drive. Fig. 105 shows a drive-system suitable for tape-speeds of $3\frac{3}{4}$ and $7\frac{1}{2}$ inches/sec — with higher speeds and larger reels, one motor is not enough to drive both the take-up reel and the tape, therefore a separate motor is required for the reel.
Here we encounter another problem, which does not arise with gramophone records. After being played, the tape is on the take-up reel and must be re-wound on the supply reel before it can be played again. It would not be very convenient to have to do this at the same speed as the playing and thus wait, say, 32 minutes before the tape is ready to be played again. Therefore, facilities must be provided not only for operating the reels in reverse, but also for increasing their speed of rotation in reverse. On the one hand, this imposes a heavier load on the motor, so that it is more difficult to make one motor carry all the load, and on the other hand it necessitates another extra motor, making three in all, in cases where one motor is already insufficient. With three motors, moreover, rather complex circuits are required, therefore recorders so equipped are usually very much more expensive than those with only one motor. Thanks to the growth of technical knowledge in recent years, good recorders which formerly required three motors can now be made with only one.
At the same time, tape acceleration should be possible not only for rewinding, but also in the normal playing direction. Whereas with gramophones the pickup can be lowered by hand anywhere on the record, with tape recorders the only way of selecting given parts of a recording is to run the tape past the reproducing head until the particular passage is reached. To avoid long delays, then, "fast forward" is also necessary. In general, the fast speeds are about ten times the normal speed for recording and reproducing. To avoid extra head and tape wear and to spare the motor as much as possible, the driving roller and the heads are held clear of the tape during accelerated reeling.
Because the previous recording is erased from the tape automatically when a new recording is made, recorders must be so designed that it is impossible to switch from playback to recording accidentally and thus destroy what may be an irreplaceable and cherished recording. In some cases such accidental erasure is avoided by employing two push-buttons to start recording, instead of only one; in others the

recording switch is controlled by a rotating knob so designed that it will not turn to the recording position until pressed down, instead of push-buttons (see fig. 108).
Recordings made without enough current in the recording head are unpleasantly noisy in reproduction; too much current in this head causes distortion. It will be evident, then, that some means of checking the signal strength during recording must be made available. Special meters are provided in professional equipment, but are too expensive to be used in the more popular models. Instead, small cathode-ray tubes of the type used as tuning-indicators in radio sets, or neon lamps, are employed in most cases, The amplifier is then so adjusted that the tube or lamp glows faintly during the loudest passages; the tube begins to glow just before the maximum permissible recording level is reached. Experience has shown that this arrangement is quite satisfactory.
The voltage generated in the playback head is of very much the same order as the output voltage of a magnetic pickup, and therefore too low to be applied straight to the pickup socket of a radio set. As the microphone-output, is likewise on the low side, even with a crystal microphone, extra pre-amplification is necessary. Tape recorders therefore contain a sensitive amplifier, used in recording as well as reproduction, as also an oscillator to supply the H.F. alternating current. Fig. 109 is the block diagram of a typical circuit.
With the switch in the left-hand position, we have: microphone (M) to input, and recording/reproducing head (R) to output, of amplifier (1); also oscillator (2) to recording head (R) and erasing head (E). Moving the switch to the right cuts out the microphone and the oscillator and connects head R to the input, and the speaker to the output of the amplifier.
As we have seen, a certain amount of attenuation takes place during the recording of high-frequency signals. To make good, say, 6 dB attenuation at 8000 c/s, a filter giving 3 dB more amplification here than at lower frequencies is provided. As such filters are employed in reproduction as well as in recording, the associated loss of sensitivity is fully compensated. Hence it is possible, say, to run the tape at 3.75 inches/sec past a 7½-micron air-gap and nevertheless obtain straight response up to 8000 c/s. In tape recording, as in disc recording, a certain amount of cut in the bass is necessary as a means of avoiding distortion. A filter is provided to boost the bass back to the proper level in the reproduction.

Section 5. Extending the playing-time

The playing-time of tape 55 microns thick on 5-inch reels is 16 minutes at a speed of 7½ inches/sec, or 32 minutes at 3.75 inches/sec.
As these times are not always long enough, various methods of extending them without employing larger reels have been tried. Firstly by using thinner tape; 5-inch reels hold 200 yards of ordinary-size recording tape, or 300 yards of tape only 37 microns thick, an increase of 50% in the playing time. As often happens in technical development, however, this improvement is accompanied by one or two problems. Because the thickness of the magnetic coating must not be reduced, this must be 15 microns on thin, as well as on ordinary, tape, and therefore leaves only about

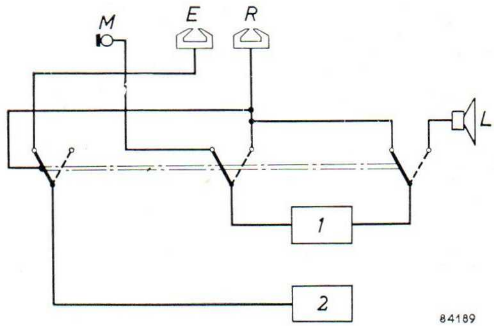


Fig. 109. Block diagram of a tape recorder: M microphone; L loudspeaker; E erasing head; R recording/reproducing head; 1 amplifier; 2 H.F. oscillator

22 microns for the thickness of the plastic base. Nevertheless, to avoid snapping or undue elongation the thin tape must satisfy the same mechanical requirements as the ordinary tape. Polyester has the necessary properties, so that by using it instead of cellulose acetate we solve the mechanical problem, and have only an electrical problem to deal with.

We have seen that when a piece of steel is placed close to a magnet it also acquires permanent magnetism. Something of the kind also happens when an unmagnetized piece of recording tape is placed side-to-side with a magnetized strip; the "virgin" tape is then magnetized slightly by the other. The strength of such "copied" magnetization depends upon the distance between the two layers of magnetic material, and this distance is for all practical purposes simply the overall tape-thickness. The length of time that the two strips of tape are in contact also governs the strength of the copied (or echo) magnetization, which, in itself, is proportional to the magnetic power of the originally magnetized strip. Last but not least, this echo effect also depends on the frequency of the signal recorded on the original magnetic strip. With 55-micron tape run at 30 inches/sec the echo-effect is strongest at 2000 c/s; the critical frequency decreases in proportion with the tape speed, so that at 3.75 inches/sec it is 250 c/s. Magnetization echo at the critical frequency is roughly 50 dB weaker than the magnetization which caused it. At higher and lower frequencies the difference is about 55 dB. The pre-echoes and post-echoes sometimes heard on gramophone records are usually due to this effect. Being 50 dB weaker than the signal producing it, such echo as a rule is not very noticeable; its strength is usually below the level of the recorded pianissimo passages.

That the effect is not the same at all frequencies explains why echo is more noticeable in some music-recordings than in others. If the critical frequency is, say, 2000 c/s and the particular fortissimo passage contains no tones at or near this frequency, then for all practical purposes there will be no magnetic echo.

Now, with thin tape the space between the layers of magnetic material is smaller and the echo-effect therefore stronger. Using 37-micron instead of 55-micron tape increases this effect by roughly 5 decibels, so that the echo- signal is then only 45 dB below the fortissimo level. This means that the echo is roughly equivalent in strength to a pianissimo passage, and therefore interferes very noticeably with the music; hence gramophone-record manufacturers do not use thin tape. The critical frequency, at which the echo effect is strongest, is roughly half an octave higher with thin tape than with tape of normal thickness. In the case of non-professional recorders, operating at tape speeds of 3.75 and 7½ inches/sec, the frequencies at which magnetic echo is most likely to occur are 400 c/s and 800 c/s. Since relatively faint sound at these frequencies is not very audible, echo in such recorders will not be noticed very much even when thin tape is employed.

Another method of extending the playing-time, applicable to ordinary, as well as to thin tape, is the recording of two sound-tracks on one strip. This is done by recording first along the top, and then along the bottom edge of the magnetic coating, as shown in fig. 110. Whereas we have stated so far that the height of the air-gap should equal the width of the tape, in twin-track recording it should be slightly less than half the tape-width. On the other hand, it might almost be said, inevitably, this method has a drawback. The flux to generate voltages in the playback head during the reproduction is weaker when the recording covers only half the tape than when the magnetization covers the full tape-width. Hence the voltages generated in the playback head are also relatively weaker, and the ratio of these voltages to the background noise is less favourable. Owing to this fact twin-track recordings are no use for professional recordings, nevertheless the noise level is low enough not to trouble the amateur.

With twin-track recordings the top track of the tape is recorded (or played) first; when the whole tape has passed from the one reel to the other, the two reels are exchanged to bring the bottom track to the top and facing the recording or reproducing head. It is also possible to provide the recorder with two sets of heads, so positioned that the top track passes the one, and the bottom track the other, pair of air-gaps. Then the reels need not be exchanged after the recording of the first track; all that is necessary instead is to reverse the tape drive, and even this can be done automatically. At the same time, it will be evident that the use of two sets of heads with the associated extra switches makes quite a difference to the price of the equipment.

One very useful feature of ordinary tape recordings is that parts of them can be cut out as required, and the remaining strips spliced together; also, extra strips can be spliced in.

On the other hand, twin-track recordings cannot be cut and spliced, which is another reasons why they are not used in commercial recording studios; amateur sound technicians working with twin-track equipment should remember that they can use only one track of the tape if they wish to do any splicing.

Section 6. Tape or gramophone record.

Tape recorders are regarded in some quarters as the rival of the gramophone, and it is sometimes forecast that they will supercede the latter in the foreseeable future. Now, prophesy is always rather risky, particularly when it is to appear in print, and to be on the safe side we shall stick to a more sober comparison of the respective merits of tape and disc. Also, we shall refer only to commercial recordings of music on tape and on gramophone records, leaving home-made recordings out of account.

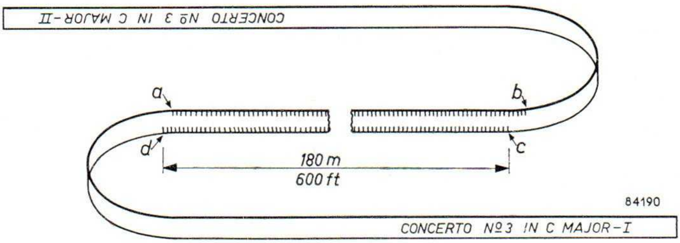


Fig. 110. Twin-track recording: a) start of first movement, b) end of first movement, c) start of second movement, d) end of second movement. Hatching indicates the two sound tracks

The three factors to be considered are: quality, simplicity of operation, and price. Except for people like the late Jules Verne, whom the writer cannot emulate, it is impossible to write about things not yet invented (and perhaps never to be invented), therefore we shall keep strictly to the situation as it is at the time of going to press. In other words, the only tape speeds which will be considered here are 3.75 and 7½ inches/sec, since at other speeds the recordings are either too poor in quality or too expensive. We have seen from the table and so on in section 4 of this chapter that the highest frequency recordable at a tape-speed of 7½ inches/sec with a 7.5 micron air-gap is roughly 18000 c/s, which is more or less the same as with microgroove records. This does not imply that frequencies up to 18000 c/s are recorded on all microgroove records; all it means is that this frequency is about the limit for tape and for discs. Tape recordings have a slightly wider dynamic range than gramophone records, that is, they permit of more contrast between pianissimo and fortissimo; at the same time this in only a minor point, since many records nevertheless offer more contrast than is appreciated in sitting-rooms. The root causes of non-linear distortion are inherent in tape and in disc. In both cases it is to be expected that the effects will be reduced to a minimum, and that the difference in non-linear distortion between tape and discs will remain small. Admittedly tapes do not wear as much as discs, but with proper care record wear is in any case so slight that private owners are not very much troubled by it, although this does not necessarily apply to equipment operated in cafes, schools of dancing, and so on. Damage, of which there is a certain amount of risk with gramophone records, is out of the question with tape recordings; in this respect, then, tape is very much better, at any rate apart from the risk of accidental erasure. Whereas it is quite easy to find certain passages of recordings on discs, this is very much more difficult with tape recordings, and then involves a lot of chopping and changing. Also, fitting the reels, threading the tape and rewinding where necessary make tape recorders less easy to operate than gramophones. This drawback weighs most heavily with album recordings comprising several short compositions, but also affects longer works.

A special problem arises with twin-track music-recordings. Symphonies, for example, are in four movements, usually of different length. With gramophone records the first two movements are usually recorded on the one, and the other two movements on the other side. Where certain of the movements are rather short (taken together, the third and fourth movements of Beethoven's Fifth symphony play for roughly 14 minutes against 18 for the first two movements) this does not matter very much, seeing that the needle enters the run-out groove in any case as soon as the last note is played. With tape recordings, however, there are two possible ways of dealing with two movements of unequal length. Firstly, to save tape, the transition from the first to the second music track can be taken exactly in the middle of the musical composition. The total playing-time of Beethoven's 5th symphony in 32 minutes; 16 minutes after the start, the orchestra is well into the Andante, so that here economy is incompatible with musical taste, which cannot tolerate any interruption of the Andante. Supposing that the musical requirements are satisfied, the length of tape required depends on the overall playing-time of the 1st and 2nd movements (about 18 minutes); this leaves about 55 yards of the second music track (corresponding to 4 minutes playing-time) unplayed at the end of the symphony. Such non-playing periods are the more inconvenient when they occur on the first track; then, either the interval between the first and second parts of the recording is too long, or the music does not start until several minutes after the reproducer is switched on, at any rate unless an overture is used to fill the gap.

The price problem cannot be discussed either simply or in definite terms owing to the fluctuation in the prices of gramophone records and music tapes. The cost price of classical long-playing records can be split up into three parts: Cost of recording (musicians, recording technicians, rent of studio, and so on). Cost of multiplication (moulds, moulding compounds, pressing-costs) and Cost of selling (programme agency, advertising, packing and dispatch, sales staff).

At a very rough estimate, the three items are about equal. The recording and selling costs of music tapes are the same as those of discs, therefore possible price-differences can only arise from a difference in the cost of multiplication.

As we have seen in Chapter III, when once the matrices are made and the presses adjusted, gramophone records are pressed in one operation, so that it takes only about one minute to multiply forty-five minutes of music; with magnetic tape, on the other hand, such a high rate of multiplication is not obtained quite as readily. The most widely used method is to play back the particular recording, amplify the voltages thus generated in the reproducing head and apply them to a recording head, which reproduces the music on another tape. It takes as much time to copy forty-five minutes of music in this way as to record it, and that is, of course, far too long. The way to shorten the copying time, and thus reduce the costs, is as follows. Firstly the tape is driven at, say 4 times the nominal speed, and in the case of twin-track recordings both tracks are recorded at the same time. Secondly, the copy-recording is made backwards, so that after copying the beginning of the recording is on the outside of the reel and rewinding (apart from the original recording) is not necessary. And last but not least, several copies are made simultaneously, instead of only one at a time. Although thanks to these methods the cost in labour and machines of copying magnetic tape recordings has been cut considerably, it is still very much more than the corresponding cost of record-pressing, partly owing to the reel-changing and tape-threading involved. On the other hand, the fact that tape-recordings are copied without the aid of matrices is a slight advantage from the price point of view. The material used to make gramophone records is by no means cheap; still less in the case of magnetic tape. The present-day price of a 200-yard reel of unrecorded tape (64 minutes of twin-track playing-time at a tape-speed of 3.75 inches/sec, and 32 minutes at $7\frac{1}{2}$ inches/sec) is certainly not less than one third of that of a 12-inch long-playing disc. It is therefore safe to say that even without the cost of copying, the price of the tape itself makes music tapes more expensive than comparable gramophone records. This price comparison also holds good for thin, long-playing tape. By and large, then, we see that from the point of view of quality tape offers one or two slight advantages, and also that it is less vulnerable; on the other hand gramophones are simpler to operate. Tape recordings at 3.75 inches/sec are more expensive than 12-inch long-playing discs, and those at $7\frac{1}{2}$ inches/sec are very much more expensive than 10-inch long-players, which have roughly the same playing time. Hence it seems likely that music tapes will supplement, rather than supersede, gramophone records.

The manufacture and storage of matrices, including "fathers" and "mothers", and the adjusting of record presses are so expensive that gramophone records do not pay unless pressed in relatively large batches. Therefore musical works for which there is not enough demand are not issued on gramophone records, even when there is an opportunity to record them in a concert hall or broadcasting studio. On the other hand, the preparations for copying tape recordings are relatively simple, therefore music tapes are the ideal medium for the less popular compositions.

Works such as Wagner's Nibelungen Sage, still not available as a set of gramophone records, could be sold as a music tape. Modern works by young composers, rarely played, are still more rarely obtainable as recordings; it is only by exception that we find, say, "Musique concrète" in a record catalogue. Here, then, is ample scope for the music tape.

Fig. 111. Tape recorder in use in a recording studio.

CHAPTER XII

TECHNOLOGY AS THE SERVANT OF MUSIC

Section 1. The recording session

Technology and music converge only at the beginning and end of the long chain from microphone to ear. In preserving and reproducing music, technology is entirely subordinate to it; the listener has the last word as to whether technology has been equal to the task, and whether proper use has been made of it.

The closest association of music with technology occurs in the recording studio, where we have on the one hand a group of musicians to interpret the particular composition as pleasingly as possible, and on the other hand a team of non-instrumentalists to see to it that the performance is recorded to the best advantage in every respect. The writer has purposely refrained from referring to the latter team as "technicians", because the studio is responsible not only for technical efficiency but also to a certain extent for the quality of the music itself. In effect, the studio director is the intermediary between technology and music and must therefore know enough of both; his work begins long before the actual recording.

Once it has been decided to engage certain musicians to perform a given musical work, the studio director begins to study the composition. He reads the score, probably plays it through on the piano, and, if necessary, learns something about the history of the music. The information thus obtained enables him to decide what requirements should be imposed on the hall in which the recording is to be made. It will be evident that a string quartet should not be recorded in a church, nor the Messiah in a small cinema, and in practice the relationship between musical composition and recording space also imposes other, very much more stringent, requirements; halls ideal for opera may well be less suitable for, say, a Haydn symphony. Therefore, to ensure the best possible results, recordings are often made not in specially designed studios, but in a hall which happens to have the acoustic properties best suited to the particular piece of music, in many cases a hall built in the period when the music was composed.

When once the hall is chosen, the microphone arrangement is worked out on paper. Very often, the best place in the hall from which to hear the performance is not the best position for the microphone. The ear is able to perceive from what direction sound reaches it, and this gift enables us to follow a violin solo quite easily, although the overall sound of the accompanying orchestra may be louder. Lacking this sense of direction, the microphone does not distinguish between soloist and orchestra, and if it is put down more or less at random the violin tones in the recording may well be masked completely by the orchestra. Hence it is often necessary to use more than one microphone.

This applies not only to the recording of orchestras accompanying a soloist, but

Fig 112
Experimenting with the microphone arrangement to record a performance by the Concertgebouw orchestra, conducted by Eduard van Beinum

also to purely orchestral recordings, where correct balance between the different sections of the orchestra cannot always be obtained with only one microphone.

In some cases, one of the extra microphones is placed some way away from the orchestra in order to take full advantage of reverberation and thus give the particular recording more "presence" than it might otherwise have. Another point to be considered is the nature of the musical composition; in Mozart piano concertos the solo instrument and the orchestra are to some extent opposed, whereas in Brahms concertos piano and orchestra play more or less as one; it will be evident that different microphone arrangements are required to give full expression to such different concepts.

The above does not imply that orchestral recordings are always made with several microphones, nor that one microphone is always enough to record music by small groups of musicians. There are no hard and fast rules governing microphone arrangements; studio directors are guided for the most part by intuition, experience and a thorough knowledge of the technical and musical requirements. Often they walk about the hall during rehearsals, blocking one ear with a finger to hear how the music will sound to the microphone, and use this information to improve the arrangement already planned.

As a result of such experiments the director may decide to use only one microphone to record a symphony orchestra, possibly with one or two baffles and a certain amount of re-arrangement of the orchestra itself, or to use two microphones for a small ensemble. Again, where a singer with a very powerful voice is to be accompanied by a guitar, there is a risk that the guitar will not come out clearly enough; although better balance is then obtained by placing the microphone close to the guitar and relatively far from the singer, it may be found when the recording is played back that the singer sounds to be standing further from the listener than the guitarist. This undesirable effect arises from the fact that owing to the singer being farthest from the microphone the voice reverberation is stronger than that of the guitar, and to avoid it two microphones must be employed. Amateurs should also listen with one ear in order to determine the best position for the microphone,

Section 2. Tape-splicing and sound-effects

Gramophone recordings are carried out even more carefully than radio broadcasts. Whereas listeners may not notice minor slips in the performance during the first playing, they are almost certain to do so after hearing the recording several times, and nothing is more irritating than listening to a gramophone record, knowing that a wrong note occurs at a given point in it. Therefore all recordings are preceded by long rehearsals; individual parts of the music are rehearsed separately until (at any rate in the opinion of those responsible) it is reasonably certain that the interpretation is in keeping with the spirit of the particular composition, and can be played without error. As experience has shown that the longer works cannot be played faultlessly from start to finish, such works are recorded in sections, which are then joined together (without much difficulty, thanks to the tape-recorder), or the whole, or a larger part, of the work is played through, and sections in which errors have been made are played again, cut out of the tape and replaced by the faultless strips.

It is usually up to the studio director to decide when the musicians have mastered the composition enough to be ready for the actual recording. He also decides what passages are to be re-played, bearing in mind that if they are too long fresh errors are likely to be made in the re-play, whereas if they are too short the musicians will have no time to catch the spirit of the music.

The studio director is also responsible for the editing of the different strips of recorded tape; he must therefore be able to read the score fluently so as to ensure that no notes are omitted or duplicated on the record.

It would be reasonable to suppose, then, that the conductor is subordinate to the studio director. Not so; the two are responsible for entirely different aspects of record-making. The conductor interprets the particular composition, for which interpretation he alone is responsible, and extracts the best possible performance of it from the orchestra; the studio director is in the same position, literally and metaphorically, as the listener, therefore the music as heard by him is quite different from the music as heard by the conductor. Hearing the orchestra from a distance, the director, provided that he has sufficient knowledge, notices certain errors sooner than the conductor, who, being close to the orchestra, is not in a position to hear, say, an ill-timed entry quite as readily.

Having completed the recording, the conductor and the studio director decide between them whether it is up to the prescribed standards. Of course, the musicians are not always to blame for imperfections in the recording; a wrong adjustment by the recording technician, noise from low-flying aircraft or heavy traffic outside the studio, or the slamming of a door may also spoil recordings. The writer has known a recording to be spoiled owing to a stage hand coming in with a large notice "Silence, recording in progress".

If the recording were cut direct in the lacquer disc, such interruptions would mean starting all over again: with tape recordings, however, the spoiled part is simply cut out of the tape and replaced by a separately recorded strip. In many cases such "plastic surgery" is confined to a single note or measure. Despite long experience, as also the fact that errors can be corrected, recording still takes a very long time; a whole day to record a symphony or ninety minutes to record a foxtrot are by no means exceptional. Faster recording is possible only if the studio director or the musicians are not very particular; then, of course, the recordings are cheaper and at the same time inferior.

Now that High Fidelity reproduction has caused some listeners to be more critical about the distinctness of triangles, cymbals, piccolos and so on, there is a tendency so to arrange the microphones that, say, the triangle sounds very clearly and almost as loud as the entire orchestra. In many cases there is no musical justification for this, therefore it is worth pointing out that the "fidelity" requirement refers entirely to music. Studio directors should never chase after effects; on the other hand, listeners should not reject a recording merely because the cymbals do not "crash out of it", when this was not intended by the composer. Again, overaccentuated fortissimos, so recorded in order to sell the record, have nothing whatever to do with true High Fidelity. The proper recording-level, particularly for symphonic music, is sometimes very difficult to find. With musc involving more contrast between fortissimo and pianissimo than can be recorded on the disc, the volume must be adjusted with reference to the score — therefore the sound technician must also be able to read music.

Whereas the contribution of technology to the recording of classical music, although very important, is nevertheless usually entirely passive, it is sometimes very much less so with more popular numbers. In classical recording echo-chambers are some-

Fig. 113
Five microphones in use at a recording of a light orchestra. The recording engineer is visible behind the window.

times used to produce given effects, but in most cases that is all; for popular music, on the other hand, a whole range of technical sound effects is available.
Echo-chambers are large compartments, with hard bare walls, containing a loudspeaker at one end and a microphone at the other. The loudspeaker is connected through an amplifier to the studio microphone, and the echo-microphone to the recording equipment; the sound reaching the echo-microphone comes partly direct from the speaker and partly by reflection from the walls of the chamber. Also, a screen may be placed between microphone and speaker, and the reverberation in the chambers so equipped is often louder than the direct sound. The output voltage of the echo-microphone is added to that of the studio microphone and applied to the tape recorder. Volume, tone and reverberation time can be varied at will by moving the speaker and the echo microphone and varying the output voltage of the latter. For example, such a chamber is used during the recording of a part of Strauss' opera Salome, and thanks to it Jochaanan's voice sounds as if coming from a dungeon. Special effects can be obtained by having a singer and an orchestra in separate studios and superimposing an echo-signal on the song. This gives the impression that the soloist is standing some distance from the orchestra. An unusually intimate atmosphere is created by a vocalist standing very close to the microphone and singing very softly; this can also be enhanced by means of an echo-chamber.
Where no echo-chamber is available an electronic echo can be used instead. The signal from the studio microphone is then recorded on magnetic tape, re-played immediately, re-recorded on the same tape, and so on. Whereas the echo-chamber produces a rather vague reverberation, electronic echo gives us a succession of relatively sharply-defined echoes. This more pithy sound is better suited to certain types of popular music than the more natural echo obtained with echo-chambers, and enables the effect of several musical instruments playing together to be obtained with only one. Another possibility is to record the sound of a certain musical instrument, with or without echo, and copy the recording together with another solo on a different magnetic tape. This is the method employed by vocalists to give the effect of singing duets with themselves, as also by some instrumentalists.
When recordings are played at a speed different from that at which they were recorded, there is a proportional change in pitch. When the response characteristic is also changed, that is, when the treble or bass is boosted or cut, very special effects are obtained. For example, an accordeon recording played at half speed and with bass-boost sounds something like an organ, but with effects not obtainable with a real organ. A male voice played fast, possibly with a certain amount of bass-cut, has a "Donald Duck" effect. Playing recordings backwards also creates remarkable effects. By and large, the most unusual results can be obtained by combining one or two of the above-mentioned methods and others not mentioned here.
In how far all this is justified from the musical point of view is another matter, but such "trick" recordings, as applied to popular music at any rate, sometimes produce very pleasing results.

APPENDIX

On the following pages is set out certain data connected with the reproduction of sound. These data refer to loudness ratios (decibel chart, fig. 114) and frequency range.

In fig. 115 the frequency range of a number of musical instruments has been shown, the figures along this chart indicate the frequencies (in c/s) of the fundamental of the corresponding musical notes.

Finally, the last three illustrations give the response curves of a number of Philips' pick-ups, as these were measured with various load resistances.

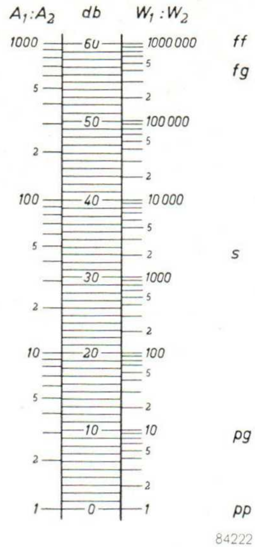


Fig. 114. Decibel chart

$A_1 : A_2$	voltage, current or sound pressure ratio.
$W_1 : W_2$,	power or sound intensity ratio.
db	the number of decibels corresponding with the above ratios.

The letters on the right indicate the sound intensity ratios for: an orchestra in a concert hall pp = pianissimo, ff = fortissimo; gramophone reproduction pg and fg = pianissimo and fortissimo respectively; s = approximately the ordinary speech level.

Fig. 115. Frequency and pitch

The tonal range of various musical instruments and voices. The fundamentals are indicated by a fully drawn line, the overtones of a number of instruments have been indicated by dots and chain lines.

1 (from left to right)
Double-bass
'Cello
Viola
Violin

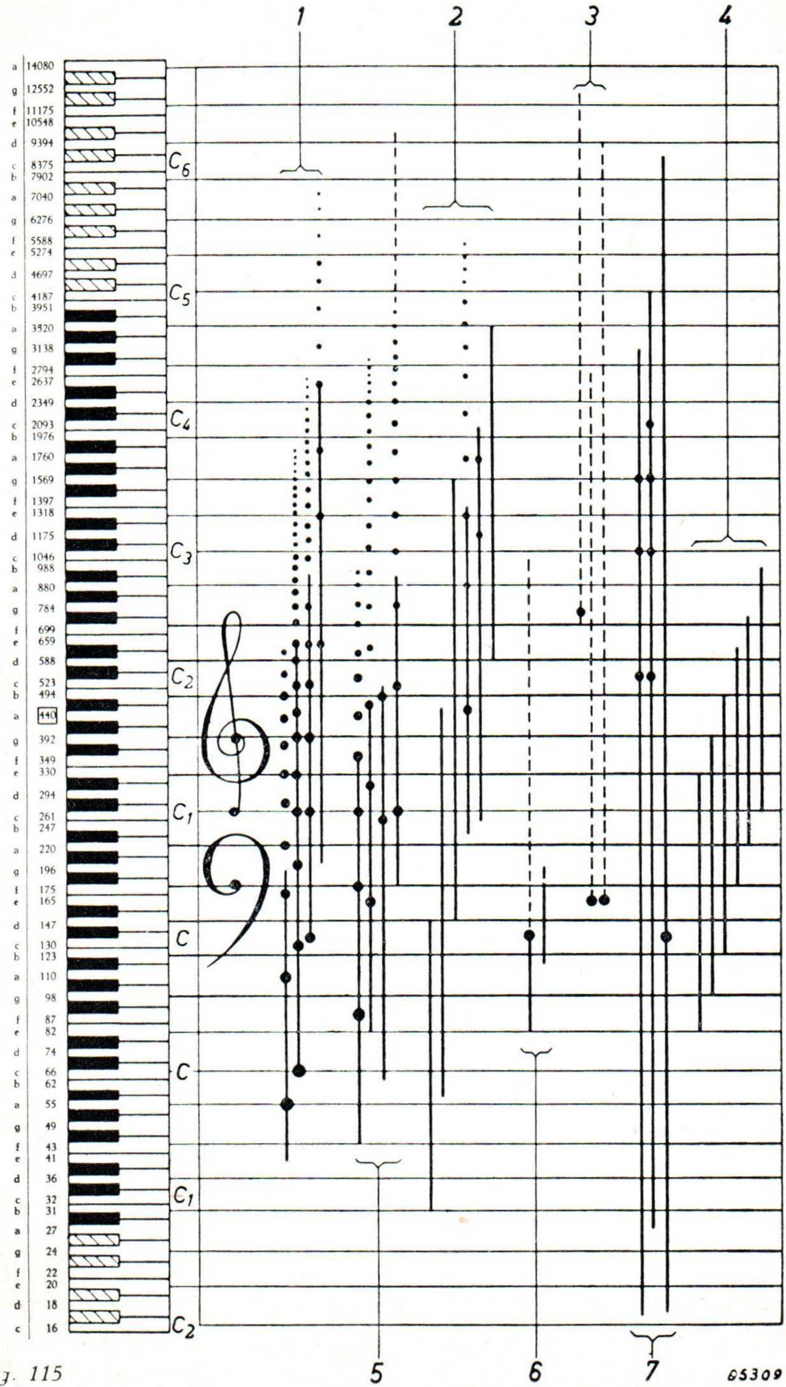
2 Contra-bassoon
Bassoon
Clarinet
Oboe
Flute
Piccolo

3 Triangle
Cymbals
Struck Cymbal

4 Bass
Baritone
Tenor
Alto
Mezzo-soprano
Soprano

5 Tuba
Trombone
Horn
Trumpet

6 Large kettle drum
Small kettle drum

7 Harp
Grand Piano
Organ


1
2
3
4
5
6
7
a 14080
g 12552
f 11175
e 10548
d 9394
c 8375
b 7902
a 7040
g 6276
f 5588
e 5274
d 4697
c 4187
b 3951
a 3520
g 3138
f 2794
e 2637
d 2349
c 2093
b 1976
a 1760
g 1569
f 1397
e 1318
d 1175
c 1046
b 988
a 880
g 784
f 699
e 659
d 588
c 523
b 494
a 440
g 392
f 349
e 330
d 294
c 261
b 247
a 220
g 196
f 175
e 165
d 147
c 130
b 123
a 110
g 98
f 87
e 82
d 74
c 66
b 62
a 55
g 49
f 43
e 41
d 36
c 32
b 31
a 27
g 24
f 22
e 20
d 18
c 16
C6
C5
C4
C3
C2
C1
C
c
C1
C2
65309

Fig. 115

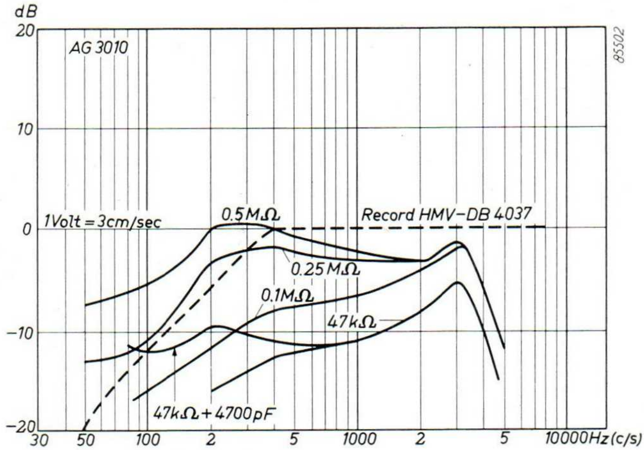


Fig. 116. Response characteristics of the crystal pick-up AG 3010 measured on the HMV test record, DB 4037 (78 r.p.m.)

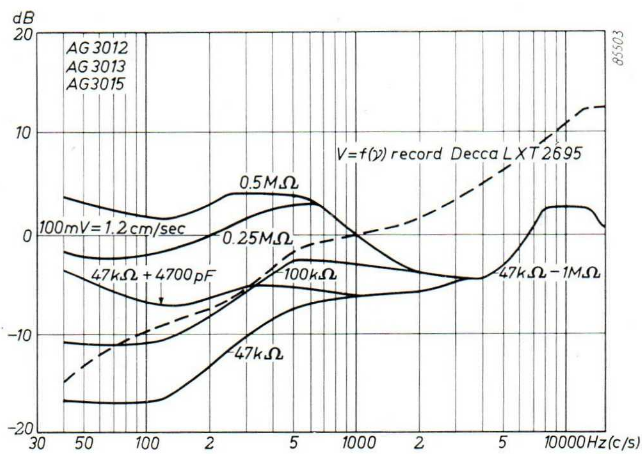


Fig. 117. Response characteristics of the crystal pick-ups AG 3012, AG 3013 and AG 3015, measured on the Decca test record LXT 2695 (33-1/3 r.p.m.)

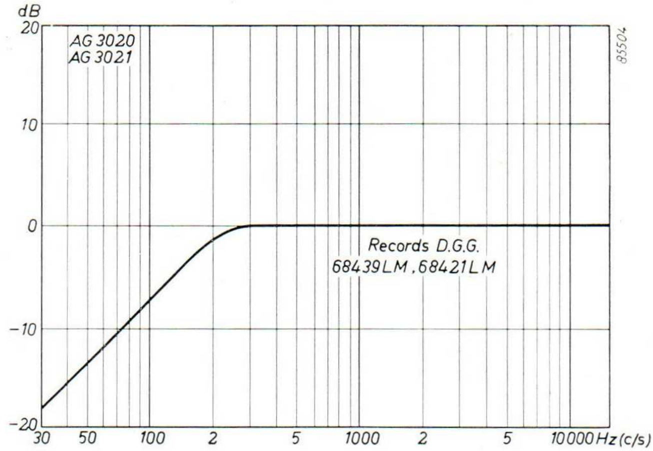


Fig. 118. Response characteristic of the magneto-dynamic pick-ups AG 3020 and AG 3021, measured on the D.G.G. test records 68439 LM and 68421 LM (78 r.p.m., fine groove, load resistance 470.000 Ohms)

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