The Hows, Wheres, & Whys of Testing High Quality Loudspeakers

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THE OBJECTIVE TESTING of loudspeakers in a free-field environment has often been attacked on the grounds that they are not listened to under these conditions and that in any case there are subtle effects which are not amenable to measurement.

Whilst these arguments contain a certain amount of truth, there is no reason why we should go to the other extreme and ignore the extremely valuable information which can be gathered from such measurements. At the BBC, loudspeakers are, in the end, judged subjectively on their ability to reproduce program material accurately, not just as a pleasant sound, and are judged by comparing the reproduced quality with that in the studio itself. When, however, questions of the basic design or modifications are involved, it is found that these can usually be determined simply by objective measurements in a free-field room. This paper describes the hows, wheres, and whys of the tests made during the development of BBC monitoring loudspeakers. The order in which the items are given is not to be taken as an indication of their importance.

Frequency Response

The steady state axial frequency response characteristic test is carried out by measuring the axial sound output as a continuous function of frequency, at a specified distance from the loudspeaker, in free-field surroundings when a constant a.c. voltage is applied to the loudspeaker terminals. It is the measurement which is most often made and contains a great deal of information.

There have been suggestions that since a listening room clearly departs widely from free-field conditions that the loudspeaker output should be measured in a live room. It is assumed that because a listener usually sits sufficiently far from the loudspeaker to be largely in the reverberant sound field that this is the factor which should be measured. In fact, the ear does not take account of the reverberation as a first order quantity but only as a second order, otherwise a person speaking in one room would sound quite different when in another room having differing characteristics, and we know from experience that this is not the case. In practical conditions the ear fastens on the direct sound and although the reverberation cannot be neglected, relegates it to a secondary place. Measurements taken under specified free-field conditions therefore contain much more relevant and easily interpreted information than those taken under live conditions which apply to that room only.

Another suggestion [1] that has been made is that an intermediate condition should be used and measurements should be made with the loudspeaker radiating into an infinite plane, i.e. into 180 degrees instead of 360 degrees. Compared with free-field measurements this would give a bass lift to the response up to a frequency which would depend on the size of the cabinet. This bass lift would therefore be a variable quantity not easily allowed for; furthermore it is admitted in the same article [1] that we do not in practice hear such a bass lift and the free-field measurement seems to agree best with what is heard in a practical situation.

The test conditions for the steady state axial frequency response characteristic need therefore to be specified quite closely.

In the first place true free-field conditions are assumed for most cases, that is unless a loudspeaker is designed to be mounted in a corner or so that the sound is deliberately reflected from a wall or ceiling. True free-field conditions can only be obtained in the open air at least 30 feet from any obstacle or in a large enough free-field room. In the latter case, the author has shown elsewhere [2] that it is necessary for the tips of the wedges on opposite sides of a free-field room to be at least one wavelength apart at the lowest frequency of interest for free-field conditions to apply, even with perfect absorption at the wedges. The trouble is that excess absorption takes place, as in an acoustically lined duct, when the spacing is appreciably closer than this; with too small a room this will have the effect of giving an apparent bass cut. In the larger free-field room at the BBC's Research Department a special type of polyurethane wedge is used and the dimensions are such that free-field conditions exist to below 40 Hz within ± 1 dB out to 10 feet from the loudspeaker under test [3].

In addition to providing free-field conditions it is essential for the measurement of the axial frequency response to be made at an adequate distance from the loudspeaker, particularly for multiple unit designs. A minimum distance of five feet is adopted for this sort of work, for it can easily be calculated that at closer distances the relative contributions of l.f., m.f. or h.f. units is changed significantly and a wrong appreciation will be obtained of what the listener will hear, in practice, at a distance of 6½ feet or over.

The next question is that of the bandwidth to strive for. We can adopt the rather näive approach that as the ear can hear frequencies over a range of 16 Hz to 20 kHz, or to over 30 kHz for children, we should aim for this range with all its attendant difficulties. At the BBC, however, we have adopted the rather more mature engineering approach of trying to determine the narrowest bandwidth which can be used

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without the listener noticing any degradation in quality. In a series of experiments [4], known under the delightful name of Operation Clothear, the upper cut-off frequency of program material was altered and the number of persons who could detect the change on an ABA test was found. The program material was carefully selected to be the most sensitive for this sort of test and observers whose ears had been checked were used. Even under these very critical conditions, surprisingly few observers were able to detect a cut-off frequency of 12 kHz. As a result it has been decided that monitoring loudspeakers should have a response extending to at least this frequency and that if this can be achieved on the axis, greater weight should be attached to obtaining, (a) a good spatial distribution, (b) a smooth curve and, above all, (c) a high degree of repeatability, than to extending the frequency of cut-off.

At the bass end the decision is more difficult and as an engineering compromise between size, cost, and response, the latter is maintained to about 45 Hz and allowed to fall below this figure.

It should be made clear that whilst the axial frequency response characteristic is a necessary measurement, it is by no means sufficient to obtain a smooth or even flat response curve. Very little work has been done to determine either the smallest irregularity which is audible, how wide-range trends in response affect the reproduction, or even, given a perfectly smooth axial frequency response curve, whether it should be flat to give the most faithful reproduction. Although it is often assumed to be true, it is doubtful whether a flat axial response curve gives the most realistic performance, but in this connection it is necessary to state our own assumptions. At the BBC we assume that the microphone and the amplifiers should have a uniform response; for tests on new types of loudspeakers the microphones used are equalized to be uniform $\pm$ ½ dB over the frequency range of 40 Hz to 15 kHz, or beyond if it is possible to do so without degrading the signal-to-noise ratio too much. It then follows that for the most realistic performance the axial frequency response characteristic of the loudspeaker must be allowed to take any form dictated by the ear, and it is found in practice that a slight slope over the frequency range from 200 Hz to 5 kHz is desirable, the response at the latter frequency being about 3 dB lower than the former. It should not be surprising that a uniform curve is not ideal, for the sound field in the listening room is very different from that in the studio and if, psychologically, a trend in the axial frequency response characteristic gives a better illusion of realism, this is regarded as entirely justified. There is also the factor that the aural effect of small degrees of coloration can be reduced by "cooking the curve." This procedure must be used with care, however, as it is not rigorous and it can easily be overdone.

It should be noted that the ear is not uniformly sensitive to broad-band changes throughout the frequency range. Thus a change in level in the 500 Hz to 2 kHz band of 1 dB is audible and one of 2 dB is quite marked. On the other hand, at the extremes of the range a change of 2 dB is barely audible at all.

Some figures from our experience are worth recording here. From the point of view of local irregularities we have an octave-band variable equalizer which in the "flat" condition shows a ripple on the frequency response curve of $\pm$ 1½ dB. That equalizer can be switched in or out and it can be stated definitely that this degree of ripple in a flat average response is absolutely inaudible. On the other hand we have had a case where a microphone had a smooth downward slope of 3 dB over the range of 100 Hz to 3 kHz. This was detected and equalized by ear by the program operators to within $\pm$ 1½ dB without reference to any kind of objective measurement! The obvious moral is that small local irregularities are permissible and that there is little point in aiming at too smooth a curve, but that broad trends are detectable to quite a fine degree.

Off-Axis Response Characteristics

The off-axis response is measured in a similar manner to the axial characteristic and is important for two reasons. Firstly, we do not always listen to a loudspeaker whilst seated on its axis, and secondly, it is largely the off-axis curves which determine the reverberent sound.

Taking the first point, it is important with monitoring loudspeakers, and to a lesser extent with the domestic types, that there should be a wide angle over which a listener can hear accurate reproduction, preferably indistinguishable from that on the axis. With multi-unit loudspeakers, apart from the coaxial types, this implies that care must be taken in mounting the units to get the best distribution in the desired plane. Thus for normal monitoring and domestic listening a two-unit loudspeaker would have the units mounted one above the other so that the system is symmetrical in the more important horizontal plane. In some cases in broadcasting, e.g. in a mobile control room, the opposite may be the case and it may be in the vertical plane that uniform characteristics are required. A further limitation with multi-unit loudspeakers is that there is a miminum distance at which they should be listened to if equal contributions from the units are expected.

The sort of trouble that is experienced off axis with a two-unit loudspeaker is illustrated in Fig. 1. The two units might, for example, be a 12 in. woofer and a two in. tweeter. If the overall response is made flat on the axis, that at 60 degrees might well follow the second curve, for at the upper end of its band the woofer could be quite directional whilst the tweeter, where it takes over, should be omnidirectional. This variation can be reduced by partially covering the woofer with plates leaving only a narrow slit to radiate the sound. The process must not be carried too far however, as the inertance of the slit resonates with the compliance of the air inside the cone giving a peak in the response followed by a sharply falling response. The degree of improvement effected by the slit never reaches the full theoretical amount; this is discussed in greater length in Ref. 5.

Greater uniformity in response with angle can of course be achieved, at a cost, by employing three units each covering a narrower frequency range. By judicious use of these methods the off-axis curves can be smooth and follow that on the axis within $\pm$3 dB for angles up to 60 degrees over most of the frequency range.

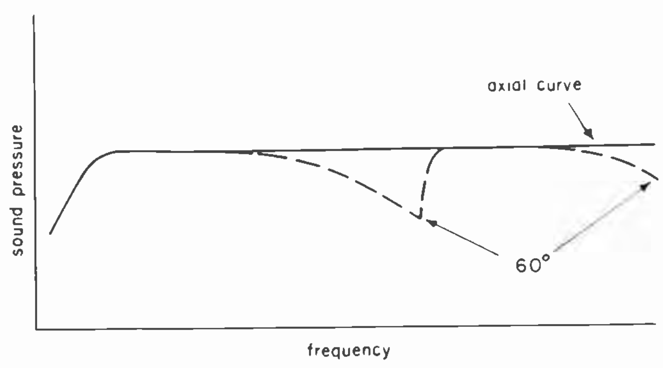

Fig. 1—Two-unit loudspeaker. Nominal frequency response characteristics on axis and at 60 degrees in the plane at right angles to that containing the two units.

The response in a plane containing the units is also irregular as at some angle in the crossover frequency range the two units are half a wavelength apart and a cancellation occurs at smaller angles than in the orthogonal plane, as shown in Fig. 2. Fortunately the crevice is narrow and the frequency varies at a discrete frequency; it does however mean that appreciable off-axis angles in this plane are to be avoided if possible. This means that contrary to many advertisements, multi-unit loudspeakers have definitely a "right way up" for serious listening.

The influence of the off-axis response on the reverberation is of course very large. If the loudspeaker is regarded as the center of a sphere which is divided into concentric bands occupying equal angles at the center, then the area covered by ± 5 degrees say will only occupy a small fraction of the area covered by 85 degrees ± 5 degrees, and the contribution to the total energy radiated into the room will be correspondingly small. The reverberation will thus be largely determined by the off-axis curves and it is at once apparent that any large discrepancy between direct and reverberant sound will be detected.

Polar Response

For this measurement the loudspeaker is mounted in the free-field room, and the measuring microphone rotated about it by a boom controlled by selsyn motors from outside the room and which also control the rotation of the polar recording paper. As with the axial frequency response characteristic, it is essential to provide true free-field conditions and the microphone must be at a distance from the loudspeaker great enough to give representative results, say five to six feet. Measurements are taken either at discrete frequencies or, more usually, employ bands of noise when general trends are required.

The polar response is of course another way of regarding the off-axis curves discussed above. It is not used extensively however because it is not the polar response as such which is listened to but the frequency response characteristic at a specific angle. The polar response measurements are therefore largely used to supplement the response at angles when a specific feature is to be examined at one particular frequency or band of frequencies during the design of the loudspeakers.

It is also useful in estimating the service area which will be well covered by one loudspeaker or in calculating the directivity or total power radiated by the loudspeaker.

Directivity and Power Response

The directivity of a loudspeaker is a measure of the degree to which a loudspeaker fails to be omnidirectional and is defined as the total acoustic power radiated at a frequency, or band of frequencies, compared with the power which would be radiated by an omnidirectional source having the same axial output. When measured in bands over the whole frequency range, it gives an indication of the way the reverberant sound will differ from the direct sound heard on the axis for a nominally flat axial frequency response characteristic and the two are therefore best dealt with together.

Since the parameter we want determines the reverberation level, this at once gives a clue as to one method of measurement. The loudspeaker is stimulated with bands of noise and the reverberant field measured as a function of frequency. By knowing the absorption characteristics of the room, the total radiated power can then be calculated from the formula:

$$SPL = PWL + 10 \log_{10} \left(\frac{Q}{4\pi r^2} + \frac{4}{R} \right) + 0.5 \text{ dB}$$

Where SPL is the sound pressure level re 2×10^{-5} N ÷ m² PWL is the power level, Q is the directivity factor, r is the distance in feet from the loudspeaker to the microphone and R is $S\alpha \div 1 - \alpha$ where α is the average sound absorption coefficient for the room and S is the area of the bounding surfaces of the room in square feet. In practice a reverberation room is used as this gives a more uniform field and has known absorption characteristics. However, similar limitations as to size apply to this room as to the free-field room and unless the room is large enough, true integration will not take place at the bass and in addition there is always some danger of the vent resonance in a vented cabinet being affected. It is however the most widely used method and properly instrumented, taking measurements at a number of points in the reverberant field, can give fairly accurate results.

The directivity can also be obtained in a free-field room by recording the polar radiation pattern at a large number of angles around the loudspeaker and calculating thence the directivity. As these measurements must be carried out at a number of frequencies, the labor involved is quite large and this method is rarely used.

The method employed at the BBC is similar but more convenient and quicker, the details being described in Ref. 6. In practice it consists of integrating the total power output "ρ" of a microphone as it is rotated around the loudspeaker in

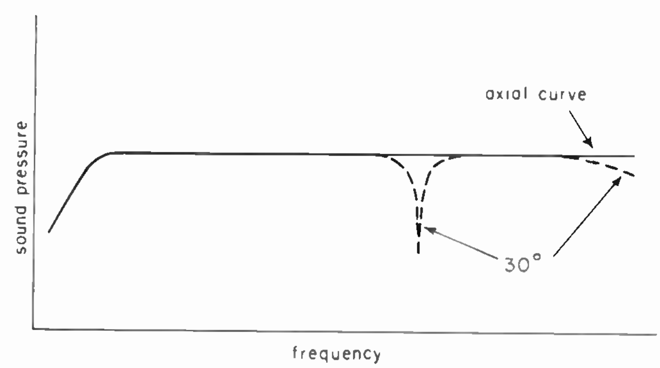

Fig. 2—Two-unit loudspeaker. Nominal frequency response on axis and at 30 degrees in the plane containing the two units.

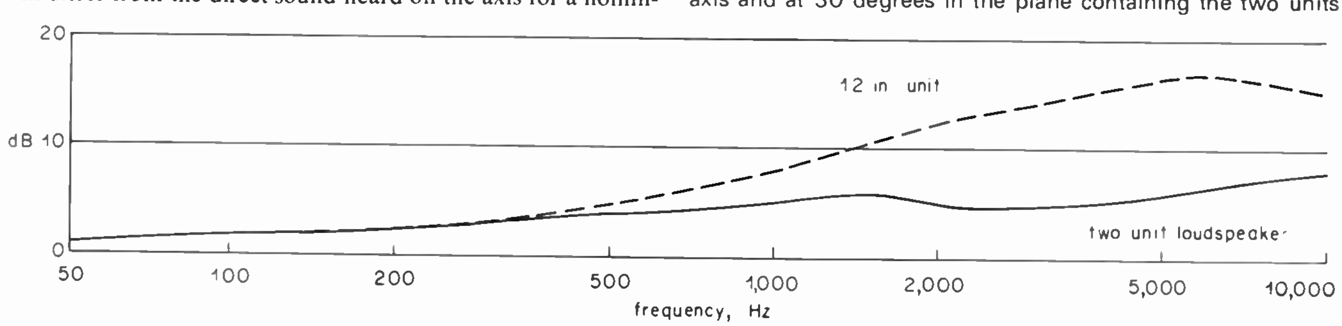

Fig. 3—Directivity of a single 12 in. unit and of a two-unit monitoring loudspeaker having a 7 in. slot in front of the 12 in. bass unit.

the free-field room in sectors, rather like the segments of an orange, for which the integral to be determined is

$$\int_0^{\pi} \pi\rho^2 \ \operatorname{Sin}\theta \times d\theta$$

The microphone is fed to a sine law potentiometer and to an integrator so that the directivity can be measured for any frequency. Since the free-field room is usable down to 40 Hz, the directivity can be measured over the whole spectrum without difficulty.

An illustration of the sort of result obtained is given in Fig. 3, both for a simple 12 in. radiator and for a two-unit monitoring loudspeaker.

It will be noted that the curve of the directivity of the latter, although much more uniform than that for a single 12 in. unit is still not flat. In the nature of things a 3 dB slope is to be expected as the bass unit is fundamentally omnidirectional whilst the tweeter can at best only radiate into a hemisphere. It is clear therefore that even if found to be desirable, a loudspeaker having a flat total power response cannot be achieved using a conventional cabinet. It is of interest to note here that a monitoring loudspeaker with a close approach to an omnidirectional middle and high frequency unit was designed by one broadcasting authority [7] but later designs from the same authority have retreated considerably from this concept. Our tests on this loudspeaker with speech and solo instruments certainly indicated that the directivity was too small for this type of program material and the later changes by the designers indicate an acceptance of this verdict. For example, with speech, too great a degree of diffusion will give the impression of a voice spread over a large area. On the other hand, at the BBC with more conventional types of monitoring loudspeaker, any increase in angle of radiation so far has been welcomed. There is therefore some sort of optimum which, however, has never been satisfactorily determined, and measurements such as the total power response for differing types of loudspeakers will help to settle this feature in the future.

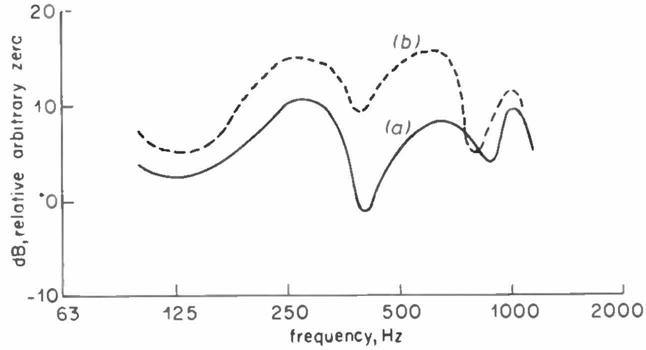

Fig. 4—Curve a, measured frequency response characteristic of loudspeaker in a corner using warble tone. Curve b, frequency response characteristic of loudspeaker in a corner calculated from three images in walls and ceilings. (Curves arbitrarily displaced.)

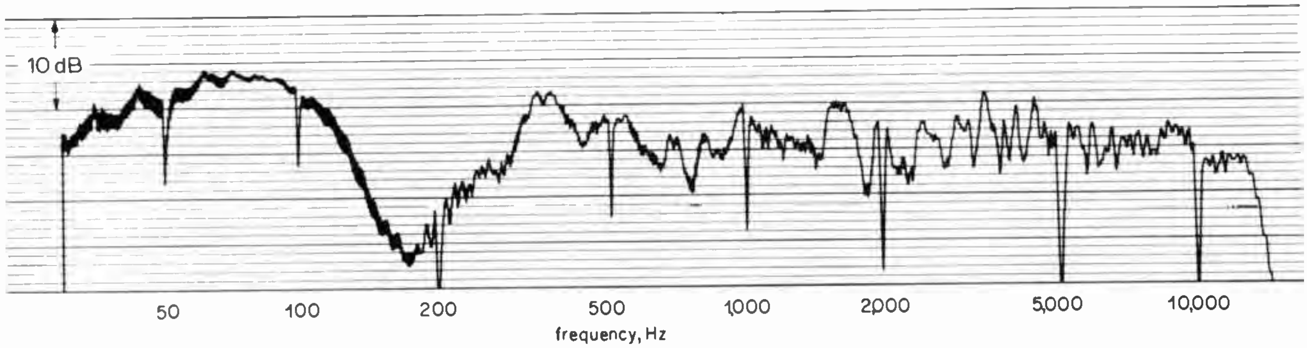

Fig. 5—Measured frequency response characteristic of a loudspeaker when touching the three surfaces in a corner.

Corner Mounting

It is sometimes most convenient to mount a loudspeaker of the conventional cabinet type in a corner. This may be to try to narrow the angle of the area to be covered or simply to hang the loudspeaker out of the way of the general impedimenta in the room. At the BBC this has been carried out particularly in television control rooms where the monitoring loudspeaker has been hung over the television monitors which are placed in a corner.

However, as the quality of speakers has improved there has been increasing dissatisfaction with the quality of reproduction from a corner placement and complaints of coloration have been made which do not apply when the same loudspeaker stands free of the corner. Measurements of the output of a speaker, fed with warble tone to remove standing waves in the room, have been carried out *in situ* with the results shown in Fig. 4, curve a. On the assumption that the irregular response was due to interference between the various images formed in the adjacent walls and ceiling, curve b shows the expected response. It will be seen that the two curves are very similar and it is not surprising that coloration was noticed at a frequency just below 300 Hz. The effects of a corner position can be mitigated by asymmetrical mounting and also by the use of absorbing materials in the corner, but these are palliatives and the use of corners for normal speakers is to be avoided whenever possible. The effect on the frequency response characteristic of placing the loudspeaker right in the corner is shown in Fig. 5 as an awful warning! For further details of these tests see Ref. 8.

Distortion and Overload

This is a subject on which most authors are silent, though not without reason. Total harmonic distortion figures of small fractions of one percent are gaily quoted by amplifier and equipment manufacturers and are expected by the customers, but figures as to the distortion generated by the loudspeaker and therefore actually heard, are few and far between. The problem divides itself into two parts, the difficulty of making meaningful measurements and the interpretation of the results.

A loudspeaker has a number of sources of distortion, viz. voice coil amplitude, spider, surround, and of course, the cone. The latter can be regarded as a transmission line, open circuited at one end and only roughly terminated at the other, having differing velocities of propagation in the radial and circumferential directions. In the latter case the fundamental frequency for a straight sided 12 in. cone will be between 50 and 100 Hz with frequent overtones above this. Radial modes do not usually set in before 400 Hz but the surround can cause trouble in this frequency region too. Even the spider will resonate and have standing waves causing an irregular frequency distortion curve, and the only item which has a smooth curve in this respect is the voice coil-magnetic field system.

It is therefore not surprising that the frequency distortion curves are extremely irregular, much more so than those of the fundamental. In order to obtain meaningful results, therefore, it is even more essential than it is for the fundamental to employ a method of measurement giving the various orders of distortion as a continuous function of frequency.

There are three such methods of measurement available. One due to Olsen and Pennie [9] employs a series of high pass filters which are switched in automatically as the test frequency is increased so removing the fundamental and allowing the sum of all harmonics and noise to be measured. Although better than nothing, it will be shown later that this measurement of total harmonic distortion is not very meaningful.

The second method is due to Bruel and Kjaer who use their 1/3rd octave band-pass filters, again switched in automatically, to measure the second and third harmonics as a function of frequency. This is better but of course we would very much like information on the higher harmonics, which is not possible with this set-up owing to the comparatively wide bandwidth of the filters. What we really need is an instrument which measures harmonic distortion up to about the eighth order as a continuous function of frequency. Since, by definition, this order harmonic cannot be measured at a higher fundamental frequency than three octaves below the upper cut-off frequency of the loudspeaker, these curves should be supplemented by intermodulation tests, again as a continuous function of frequency, which of course can extend right up to the cut-off frequency itself. Since no such instrument was available one was designed by the author for use at the BBC [10]. This is not the place to enter into details of its design, which is described in the reference given, but by means of heterodyne methods, this versatile instrument enables both harmonic and intermodulation distortion curves to be taken as described above. It is a pity that although the patent is available for exploitation, no instrument firm has produced it for use by other organizations. A typical set of curves is given in Figs. 6 and 7 for a high quality monitoring loudspeaker taken at a sound level of $1N \div m^2$ at five feet in a free-field room. Note not only how low the average distortion is but also that the higher order distortion curves are very irregular and that the frequency at which one harmonic is a maximum may even by a minimum for another. For example, if we look at the difference between the sixth and the eighth harmonics at 55 and 59 Hz, at the lower frequency the eighth is at least 22 dB above the sixth, whilst at the higher frequency it is 19 dB below, a relative change of at least 40 dB in 4 Hz! Between 250 and 260 Hz, there is a corresponding difference of over 25 dB. In fact the figures are even greater than these but the curves have been cut off at -90 dB as they cannot be guaranteed below this level.

The intermodulation curves are comparatively smooth in this case as they largely relate to the tweeter which in this design moves almost as a rigid piston up to the highest frequencies, and therefore does not break up into resonance modes.

The interpretation of these curves needs some care. In the first place, although we can see that the general trend of the curves for such a high quality loudspeaker is smooth, on the other hand, because of the irregular detail as described above, it is not possible to get the average separation of the curves by means of measurement at a few spot frequencies. The next most important point is that a simple rms sum of the levels is quite inadequate. No one would seriously dispute that one percent of seventh harmonic is far more objectionable than the same level of second harmonic. As long ago as 1937 it was demonstrated by the R.M.A. [11] that to get a reasonable subjective assessment, the level of the harmonics should be weighted at least according to their order. Since then two papers [12, 13] have clearly indicated that the weighting should be according to the square of the order, that is, instead of using the rms sum of the harmonics, each harmonic level is multiplied by $n^2 \div 4$, where n is the order of the harmonic, before taking the rms sum. In this way the level of the second harmonic remains unchanged. It is the need for this type of weighting which shows the inadequacy of the simple rms figure measured by the first of the tests described earlier.

It should be noted here that some nonlinearities can be highly nonlinear, that is to say that they may even increase rapidly with input level and then decrease again as a percentage of the fundamental. The surround is particularly susceptible to this, both near the half wave resonance point and in the bass. In the first case owing to resonance, the amplitude may increase rapidly with increasing input until the highly nonlinear region is reached. Distortion is then at a maximum and cannot increase. However, as the input voltage is increased, the output from the cone will still increase and the total percentage distortion will therefore be reduced. A similar case occurs at the bass end, particularly at the vent resonance frequency of a vented cabinet. Here, when the cone moves, say, inwards there is a very high back pressure in the cabinet pushing the surround outwards and, if it is very compliant, the surround may actually move in the opposite direction to the cone until the elastic limit is reached. Thus it will execute almost a square wave in antiphase with the cone, but again as the input power is increased the total distortion will reach a

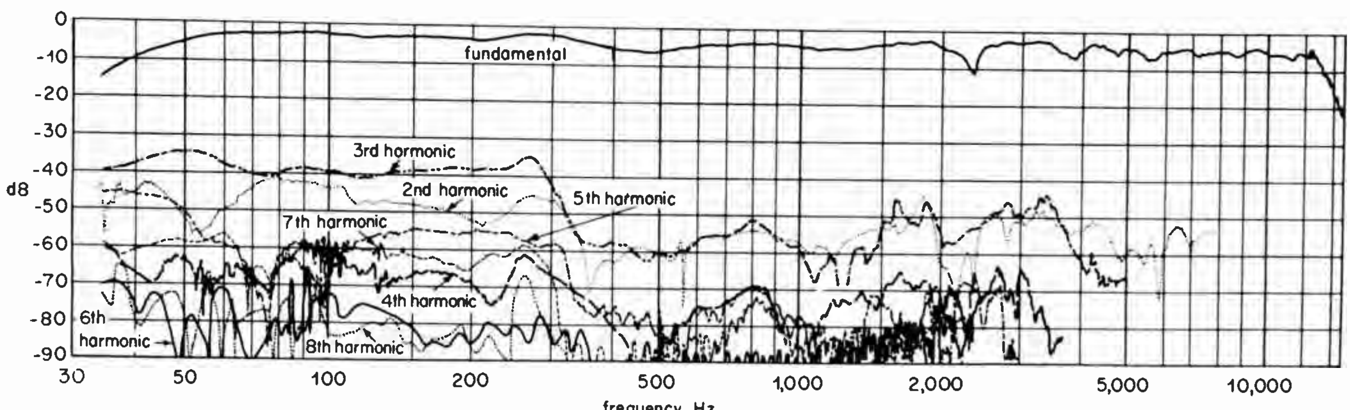

Fig. 6—Harmonic distortion curves of three-unit monitoring loudspeaker in free-field room. Sound level $1N \div m^2$ at five feet from loudspeaker.

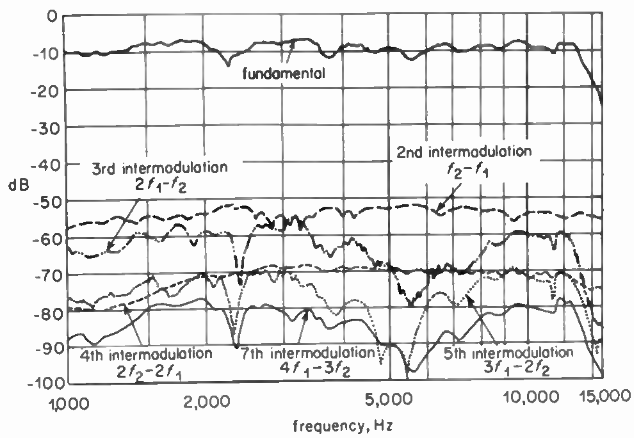

Fig. 7—Intermodulation distortion curves of three-unit monitoring loudspeaker in free-field room. Sound level $1 N \div m^2$ at five feet from loudspeaker.

peak and then be reduced. Note that in each case high orders of harmonics may well be produced in the process and that the maximum distortion may actually be at low or medium sound levels.

The overload level is related to the distortion level in a complex way. The two variables are the peak to rms ratio of the program and the spectrum concerned, as in practice amplifier/loudspeaker systems overload on peaks well before they burn out. For example solo piano will overload loudspeaker systems at much lower loudness than will an orchestra, and organ pedal notes will show up any excessive bass equalization. Thus to arrive at a stable figure, we use bands of pink noise. It may seem surprising that the overload point of noise can be heard in view of the nature of the spectrum but in practice it can be determined by ear within ± 1 dB.

Transient Response

The transient response of a loudspeaker can be measured by placing it in a free-field room and determining the response to a sudden impulse such as a square wave or by the response to bursts of tone. In theory the former test contains all the desired information but in practice it is difficult to analyze, particularly because, as will be shown later, it is necessary to measure transients well below the steady state level.

In practice therefore only the chopped tone method is useful. In this test the input to the loudspeaker is gradually changed in frequency whilst the amplitude is chopped at the input of the power amplifier (so maintaining the correct damping at the terminals of the loudspeaker) at a rate of about five times per second, so that the burst of tone lasts for about 100 mS and the off period for similar length of time. The repetition rate is a cross between a high value allowing a rapid frequency glide and a slow enough rate to allow steady state conditions to be established. For very high Qs even slower repetition rates are necessary. During the off period the output of the loudspeaker is examined for resonance which will show up as a "tail" on an oscilloscope. The degree to which the level of the commencement of the tail is below the steady state is measured; this is known as the dilution of the resonance. The Q and the frequency of resonance are also noted. At one time it was customary, at the BBC, to take delayed response curves, that is curves of the output from the loudspeaker at intervals of 5, 10, 20, 30, etc. mS after the tone had been cut off. This gives a very good picture of the transient response but is a rather lengthy procedure and the present practice is merely to glide throughout the frequency range noting the parameters given above.

The measurement of transients is another aspect of loudspeaker testing which reveals our ignorance on the subjective side. The importance of the transient response generally seems to be badly underestimated for it is no exaggeration to say that with modern high quality units the coloration caused by a poor transient response is the main factor in determining the sound quality of the loudspeaker. A good example is shown in Fig. 8. This shows the axial response of two loudspeakers of similar size, and as a matter of interest, designed by the same engineer when he was at two different firms. The top curve shows the axial frequency response curve of his first design and the lower curve his second, both curves taken by the present author. The progress made in smoothing out the axial response is commendable but the awful fact is that the first loudspeaker sounds very much better than the second. This latter has severe coloration centered around 500 Hz just where it will be seen that the axial response curve is specially smooth, whereas the irregularity in the upper curve near this frequency is relatively innocuous. This amount can be capped by the behavior of a middle frequency unit designed by us for a three-way monitoring system [14] and which also had a nicely smooth axial frequency response characteristic. On completion of the loudspeaker, listening tests showed a marked coloration in the 1500 Hz region even though the middle frequency unit had passed our usual tests. Still more careful measurement with chopped tone, however, showed up three resonances close together in frequency which had a dilution of no less than 40 dB, but a Q of about 500! Two things are noteworthy here. Firstly the effect of such resonances on the steady state is only 0.1 dB peak if they are in phase with the steady state

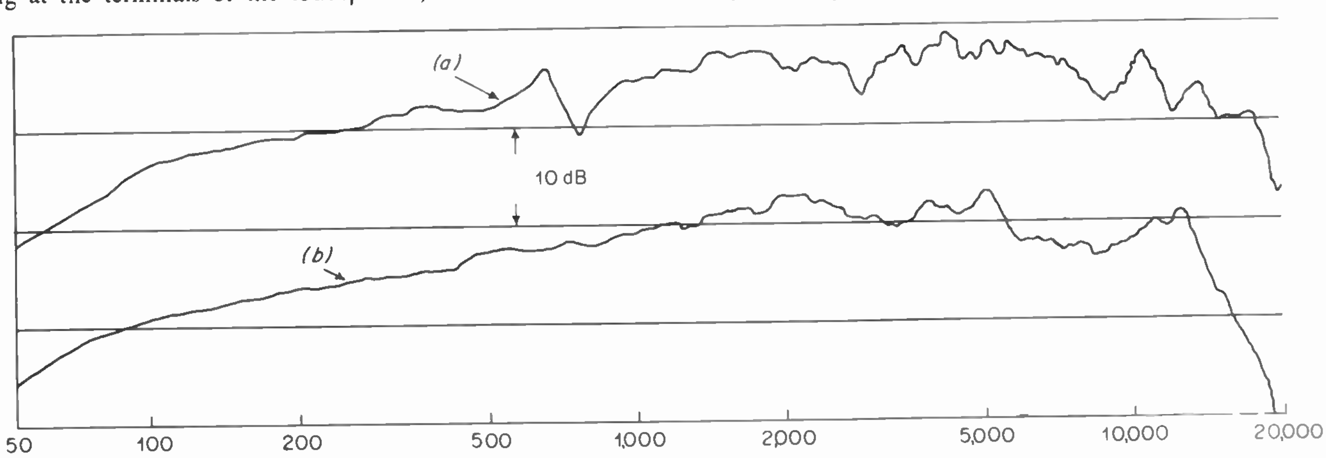

Fig. 8—Axial frequency response curves of two small two-unit loudspeakers; (a) first design, (b) second design. (Curves arbitrarily displaced.)

response and very much less still if th. y are in quadrature. Secondly it is rather surprising that a material which in flat sheet form has a Q of much less than 1 should, when formed into a hyperbolic cone, have such a high Q. It is clear that a smooth steady state curve, whilst obviously desirable, is not in itself a guarantee of absence of coloration. It is here that the German DIN standard No. 45 500 falls down. In an attempt to define conditions necessary for a good loudspeaker from the transient aspect, a maximum slope is laid down of 12 dB per octave over any portion of the axial frequency response curve. Whilst, incidentally, this rules out a vented cabinet with its bass slope of 18 dB per octave, it is quite impossible to cope with the case cited above, of a maximum disturbance of 0.1 dB, in these terms. Whilst theoretically the information is indeed in the steady state curve, in fact it is too oblique a measurement of this parameter to be useful and the chopped tone technique is the only possible approach.

Several attempts have been made to record automatically the energy in the tail of the transient but in practice the required parameter is by no means clear. Experiments to determine the subjective correlation between frequency, Q, phase, and dilution are at present being conducted here but an indication of the difficulties in the way of instrumentation lies in the fact that subjectively a suppressed zero seems to be involved and preliminary results suggest that two resonance close together in frequency may add, not on an rms basis, i.e. 3 dB as do pure tones, or even arithmetically, i.e. 6 dB, but possibly to the tune of 10 dB. At the moment therefore we cannot predict the effect of a resonance; all we can do is to listen to program material or to pink noise through the loudspeaker, measure with chopped tone those resonances in the vicinity of a coloration, and increase either the dilution or damping or both until the resonance is inaudible. In the meantime it is clear we must examine at least 50 dB below the steady state and look for Qs up to 500 or more. This calls for refined conditions of measurement, particularly in terms of standing waves in the free-field room.

Phase Frequency Response Characteristics

With few exceptions [15] the phase frequency response characteristics of loudspeakers are usually regarded as unimportant, and this accords with our experience. Measurements of phase response have been made here with the test loudspeaker in the free-field room by employing a wide range capacitor test microphone, a delay line and a phase meter of the zero crossing type connected to a level recorder. Except in the region of crossover such measurements have not been found to provide useful data and even in this restricted case equally useful information can be obtained by observing the individual contribution of the two units concerned and the way they add together.

One organization does go as far as to displace the various loudspeaker units one behind the other in order to be able to reproduce a square wave well, but it should be noted that this will only apply on the axis and leads to a complicated expensive cabinet system. We have found that by approximately adjusting the crossover network, the outputs can be made to add simply even when the units are in the same plane so that it is impossible to detect from the axial frequency response curve at what frequency the crossover is. Furthermore this will hold over the whole horizontal plane containing the axis. We have not found any further attention to phase to be necessary.

Doppler Effect

This falls into a similar category to the effects of phase in that while it must exist, we have never been able in practice to attribute any ill effects to this cause. This may partly be due to the fact that all serious listening at the BBC is done on loudspeakers with at least two units, and this of course will greatly reduce the Doppler effect. Even however with such wide range single-unit loudspeakers as the author has examined, it can be said that other faults have at least been far more important, but it is of course possible that with further progress the Doppler effect will become noticeable on program material as a small residual. No tests are therefore made for this effect.

Subjective Testing

This is the touchstone and none of our previous work is adequate if this test fails. It may be asked how this is possible in view of all the measurements we have taken, and some indications have been given in the sections concerned but it will not hurt to repeat them here. To start with, for a monitoring loudspeaker the quality of reproduction must be that of the original in all its stark reality, with no pandering to a "pleasant sound." In this it is assumed that we start with a microphone having a perfectly flat frequency response curve. But in spite of this we are still not sure what the optimum frequency response characteristic of the loudspeaker is, how much coloration we can stand, or what the best directivity is. Since the final result is subjective, we can only determine these conditions by subjective experiments and then lay down the objective results. Finally for a monitoring loudspeaker the results must hold for any type of program material. Thus a loudspeaker which obtains a very high degree of diffusion pleasant for reproducing an orchestra will not do for speech if a commentator appears to have a mouth six feet wide! The desired listening conditions must also be laid down. For a broadcasting organization it is assumed that the majority of listeners will be in their own homes, probably, for serious listening, in a living room. To this end a very large number of measurements have been made in listeners' homes and an average reverberation time of 0.4 seconds arrived at [15]. Listening rooms are therefore made to have this value of reverberation whenever possible.

One of the best forms of test material is also, strangely enough, the easiest to obtain, that of well known male voices speaking from dead surroundings. It is a fact that we are particularly sensitive to nuances in the human voice, a vast number of differing voices can be distinguished, and a well known voice is excellent test material. It has often been observed here that a loudspeaker which is balanced to reproduce the male voice is also excellent, over this frequency range, on music and other types of program material while the reverse does not necessarily hold at all. A further advantage of the speech test is that the person whose voice is being reproduced can stand behind the loudspeaker concerned and alternate live with reproduced speech.

For music tests it is necessary to have a studio at one's disposal together with an adjacent listening room, and to listen to a wide range of instruments, solo as well as in a full orchestra. Furthermore it is essential to use a single microphone pickup rather than multimike technique, or else it is not possible to listen directly to what the microphone is picking up. Recordings are a poor second best to a real performance as it is not possible to know the microphone characteristic, the reverberation, or even how the orchestra was playing that day. The latter point is quite important, as on one occasion, for example, the author thought the sound of the violins rather harsh over an experimental loudspeaker and was very relieved on entering the studio to find that harshly was exactly how they were playing at that moment.

Finally for outside broadcasts the listening conditions are often far from ideal and the loudspeaker has to be able to cope with these too. Generally the fault with such conditions is that there is not enough acoustic treatment present, a trend which is also becoming apparent in some modern homes, where the old type of deeply cushioned furniture is replaced with more sparcely upholstered types. In such circumstances the sound tends to be harsh and so will emphasize any such tendency in the loudspeaker. It is a truism to say that any excess is objected to more than a corresponding degree of deficiency. The experimental loudspeaker is therefore sent on a field trial under differing listening conditions and with differing studios.

It should be mentioned under this heading that one very convenient form of subjective test when an alteration is to be carried out is to make an instantaneous changeover between the two conditions. Now this is not always possible directly, as for example when the amount of damping compound on the cone is to be changed. It is not usually satisfactory to have one loudspeaker in one condition and one in another, as generally the difference between loudspeakers even of the same type is audible. The method we have employed is to record test material such as pink noise on the axis at a specified distance in the free-field room under each condition, one on either track of a two-track tape recorder. On replay on a monitor, switching between the two tracks can be instantaneous and the effect of even small changes can be made quite obvious and a record of them held.

Tolerances

When the design of a loudspeaker is fixed the only three parameters likely to change in production are the frequency response characteristic, the crossover network, and the coloration. Other factors such as the directivity, overload, etc., are usually constant. For a monitoring loudspeaker one essential goal is that all units should sound alike, so that if a producer records program material in one studio using one loudspeaker and edits the tape elsewhere, the balance should be identical. As all makers of loudspeakers know only too well, such a condition is extremely hard to achieve and until recently would have been thought impossible. The tolerances which the user will fix will therefore be tight as possible but at the same time must be realistic or no loudspeakers will pass the test. The response of paper pulp cones has been notoriously difficult to control in the past, particularly the thinner cones, for after all they are merely an exercise in statistics with the pulp fibers as the variable! The position has been radically improved with the use of vacuum-formed thermoplastic materials. With the right materials these can be exceptionally free from coloration and give repeatable frequency response characteristics.

This leaves the crossover network components as the remaining variable. In order to obtain the sort of frequency response required of monitoring loudspeakers, crossover networks for the last quarter of a century in BBC designs also act as equalizers for the units themselves. The tolerance on components for these two purposes is fixed at $\pm$ 2 percent to maintain monitoring standards, and for this reason paper dielectric capacitors and gapped mu metal-cored inductors must be used to maintain the required stability over a long period.

For the studio monitoring loudspeaker type LS 5/5 the tolerance over the whole frequency band for the general trend of the frequency response characteristic is $\pm$ 1 dB with respect to the standard laid down. A further small allowance is made for minor local deviations from this standard. It is found to be much more satisfactory to divide the tolerance in this manner than if, for example, the two tolerances were added to give, say $\pm$ 2 dB overall instead. This would allow larger deviations in the general trend, which is not desirable.

As may be expected from these tolerances, the degree of uniformity of performance is very good. It is rarely possible to be able to detect differences between loudspeakers even on a direct changeover and certainly not by walking between differing studios. It also means that any two loudspeakers can be used as a stereo pair and provide an excellent sharp image.

It is of some satisfaction that we can state that the first production batch of these loudspeakers all passed this stringent test without any failures thus indicating the degree of precision now possible in the loudspeaker field. For comparison it should be noted that the tolerances on the frequency response characteristic of the best grade of capacitor microphones is 5 dB at the middle and high frequencies and 7 dB at the bass. With careful design, monitoring loudspeakers can at last be regarded as precision instruments.

Conclusions

It has been shown that many objective measurements are necessary during the design and testing of loudspeakers and it is true to say that the time has passed when a high quality loudspeaker could be constructed without their aid. On the other hand there is still a good deal of ignorance as to the exact design goal defined in objective terms, and further research should be carried out to elucidate these items, especially in the field of coloration. In the absence of such information we still have to fall back on a subjective test as the final assessment. Æ

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