Some factors in loudspeaker quality

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Some of the factors in loudspeaker design have been dealt with in the technical press many times but there are others which have received comparatively little attention, at any rate quantitatively. In this article it is proposed to deal with a few of the latter and to add some subjective data which is new.

In the presentation of this material it is intended to follow the frequency scale, that is to start at the bass and work upwards.

Bass response

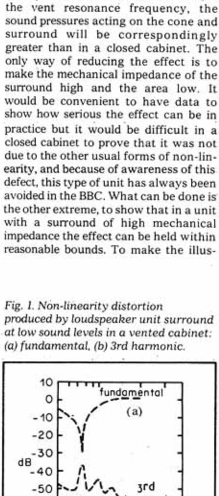
(a) **Effect of surround stiffness.** A loudspeaker is essentially a band pass device and it is well known that in a closed type of cabinet the lower cut-off frequency is set by the resonance frequency of the unit in the cabinet.

In an endeavour to obtain as extended a bass response as possible, various devices have been tried. One such is to increase the mass of the cone, but this carries with it the penalty of reduced output. Another device has been to make the combined stiffness of the spider and surround as low as possible thus allowing the cabinet volume to be the deciding factor in the effective resonance frequency[1]. The argument has been that the air stiffness is more linear than that of the spider/surround combination and that by making it the dominant factor, distortion at high sound levels is reduced [1,2,3]. This can be true at high sound levels but it will be shown here that this form of design can actually lead to increased distortion at low and medium sound levels.

Among the many functions the surround is called upon to fulfil is that of sealing the cone to the cabinet. When the cone moves backwards into the cabinet it creates a back pressure in it and this in turn attempts to drive the surround outwards, i.e. in the opposite direction to the movement of the cone. If the mechanical impedance of the surround has been made low this inverse surround excursion may be quite appreciable. Furthermore, the stiffness of most surrounds is not very linear for finite amplitudes, so that in practice under these conditions they may execute essentially a square wave and so generate a number of the higher harmonics together with the corresponding objectionable intermodulation products. It will immediately be seen that this effect is greatest at low and medium levels; at high levels the cone will drag the surround with it and the effect will be less in proportion. The worst surrounds will be those whose linearity ends abruptly, for example any type containing cloth as a reinforcing material, whereas if the surround is perfectly linear the only effect will be an appreciable loss in effective radiating area.

It will be realised that this effect also takes place in a vented cabinet where, at the vent resonance frequency, the sound pressures acting on the cone and surround will be correspondingly greater than in a closed cabinet. The only way of reducing the effect is to make the mechanical impedance of the surround high and the area low. It would be convenient to have data to show how serious the effect can be in practice but it would be difficult in a closed cabinet to prove that it was not due to the other usual forms of non-linearity, and because of awareness of this defect, this type of unit has always been avoided in the BBC. What can be done is the other extreme, to show that in a unit with a surround of high mechanical impedance the effect can be held within reasonable bounds. To make the illustration clearer a vented cabinet has been chosen so that the effect is local and easily distinguishable. Fig. 1 shows the acoustic output from an 8in unit having a free air resonance frequency of no less than 65Hz, with the microphone near the surround[4]. Curve (a) shows the fundamental together with the expected dip at the vent resonance frequency, whereas curve (b) shows the third harmonic distortion rising to a peak of 10dB at the same frequency i.e. where the excursion of the cone is least but the back pressure is greatest. Because these curves were taken at a fairly low level the distortion at other frequencies is low, and if the *total* output from the unit plus vent is assumed to be uniform at the 200 Hz level, the distortion due to the surround is not greater than about 2%. It should be stressed that this distortion is from a "good" surround – those of lower mechanical impedance would be much worse. Also for a vented cabinet design at high sound levels, the curve of the third harmonic will reverse and show an increase at the adjacent frequencies leading to a dip at the vent resonance frequency.

Fig. 1. Non-linearity distortion produced by loudspeaker unit surround at low sound levels in a vented cabinet: (a) fundamental, (b) 3rd harmonic.



(b) **Effect of total magnetic flux.** The effect of total flux on the sound output will now be considered. It is well known that if the design parameters are adjusted correctly a bass response curve like that shown schematically in Fig. 2 curve (a) is obtained. Now if it becomes necessary to increase the flux by a factor of two, curve (b) is produced. In the mass controlled and stiffness controlled areas, where the motional impedance is low, a rise in output of 6dB is produced for the same input voltage. On the other hand in the region of resonance where the motional impedance predominates, this impedance increases by four times with a corresponding decrease in driving current, and a quarter the current in a field of twice the flux gives a loss of 6dB. There is therefore a relative loss of 12dB and this has to be corrected by equalization.

Now there is another way of arriving at the desired equalization. This is to attach an accelerometer to the voice coil former and connect the output of the accelerometer to a feedback network, and for a closed cabinet this can be made to give the same result. But there is no magic about feedback and it does not change the efficiency of the unit; in essence the same operation as before is being performed, that is, equalization is being applied.

Incidentally although motional feedback is now becoming popular it is salutory to remember that in fact the idea is quite old.[5] The earliest reference known to the author is to a patent taken out by P. Voigt in January 1924. It may surprise many people who thought that negative feedback came in with Black and Nyquist ten years later, to realise that the principles and advantages of feedback were appreciated so long ago, and that they were applied to so intractable a subject as loudspeakers. There have been at least two other patents on motional feedback, one by A. Sykes in 1926 and one by M. Trouton in 1928, before the Black and Nyquist papers.

Now to return to our subject. Another method available to help the bass response is to use a vented cabinet design. This is well-known and the provision of a high acoustic load at the rear of the cone reduces the motional impedance considerably and allows a greater driving current to flow, thus improving the matching. The mechanical circuit diagram is shown in Fig. 3; the series circuit represents the loudspeaker unit and the parallel circuit the vented cabinet. Now a very simple relationship holds provided only that the impedance of the parallel circuit is high enough at resonance to swamp the remainder. Taking the series circuit first, well above resonance the circuit is mass controlled and all of the open circuit force will be applied to the cone mass which will move with a corresponding velocity. On the other hand when the impedance of the parallel circuit is dominant all the open circuit force will appear across it and in particular across the vent mass which again will move with a corresponding velocity. If then the two masses are equal we get the same output at the vent resonance frequency as in the mass controlled region of the cone. If in order to use a smaller cabinet the mass of the vent is made to be twice that of the cone, then the output at the vent resonance will be down by 6dB, and for three times the mass by 10dB. Note that nothing has been said about the two resonance frequencies.

Novak pointed out that for the particular condition where the resonance frequencies of the two circuits are equal, the relative outputs depended on the ratios of the two capacitances. Of course this follows immediately as a special case; the masses are obviously the pertinent factors. The question of ripple in the frequency response must not be overlooked of course, but the relationship is very useful when first estimating values.

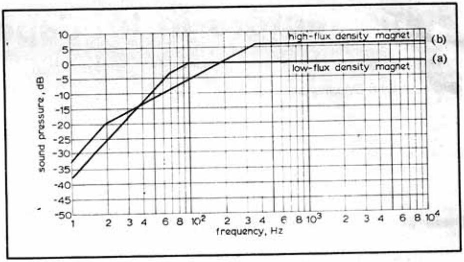


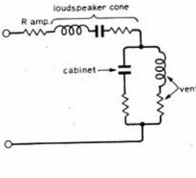
Fig. 2. Schematic curves of effect of flux on axial frequency response of a loudspeaker unit in a closed cabinet: (a) normal flux, (b) twice the flux.

Now the question arises again as to what to do if the output at the vent resonance frequency is below that in mid-band; once again equalisation is necessary.

Well what is wrong with this equalization? The answer is nothing, provided the implication is appreciated. This is that if it is desired to provide uniform sound pressure down to the cut off frequency, where 12dB of equalization was used, then 12dB more power input must be provided with all it implies, or else some distortion will be produced. If therefore it is intended to use a 50-watt amplifier for mid-band purposes then no less than 800W must be available at the bass; in addition the unit has to be capable of accepting this input without damage.

Fortunately if the input is restricted to programme the position is not quite as bad. When a high-quality monitoring loudspeaker was being designed[6] the relationship between peak programme level *overall* and the peak programme level in various octave bands in the middle and bass was examined, having the latter particularly in mind. Programmes known to have a heavy bass section were selected from classical music, pop and organ and it was found that the latter was the most demanding from this aspect. Fig. 4 shows the results of the tests. The one point at 70Hz was a solitary note from a pop group, which on the basis of statistics was ignored. It can be seen that at, say 50Hz the peak output is some way below the rest. Corresponding equalization can therefore be applied without demanding any extra power rating for the amplifier, but for any values of equalization *above* this figure the laws of nature demand a corresponding increase in available power. This aspect appears to have often been overlooked in the past.

Fig. 3. Mechanical circuit diagram of a loudspeaker unit in a vented cabinet.



Mid-band frequencies

Now let us go slightly higher up the frequency scale and consider the mid-band region.

It is well known that units become more directional as frequency increases and that to avoid excessive directional problems at least a two-unit system is normally used. If the axial frequency response curve is equalized to be flat then it is well known that the off axis curves, say, at 60° in the horizontal plane will look like the curve in Fig. 5. The off axis response is by no means uniform and on the basis of subjective tests this is undesirable, and the question arises as to what can be done about it. One simple answer is to use a three unit system, but of course, this is expensive. A cheaper solution was suggested by Chapman and Trier[7] in 1947, that is, of placing a slot over the offending unit. The idea was that sound should radiate from the slot and if the slot axis were vertical then a much better spread of sound would be obtained in the horizontal plane. It looks so simple but in practice there are a number of difficulties.

Firstly the mass of air in the slot is in series with that of the cone and will reduce the efficiency accordingly. Secondly this air mass will resonate with the stiffness of the air behind the slot and in front of the cone, and a local increase in sound output will be produced. Thirdly above this frequency the acoustic circuit will act as a single-section low-pass filter and the output will be severely reduced. The magnitude and frequency of these various effects will depend, among others, on the width of the slot, and successful design depends on achieving an optimum result for any one unit. One rather unexpected result is that in addition to an improvement in the horizontal directivity there is also a small improvement in the vertical plane.

The directivity in the horizontal plane would appear to be a simple function of the slot width but in the process of carrying out various designs the author has found that this is not so. Finally in the design[6] of the BBC LS5/5 loudspeaker it was decided to investigate the problem a little more closely. A 12in bass unit was being used with two possible alternative designs, one with the bass unit crossing over at 400Hz, the other with a crossover at 1500Hz. For the latter it appeared that a slit of 100 mm would give adequate directivity. Now it is a little difficult to estimate just what the radiation pattern from the slit will be. For example is the slit to be regarded as a line source, or as a piston in an infinite plane, or alternatively as a piston in the end of a cylinder, all possibilities for which the radiation pattern is known and for which the radiation, at say 60° relative to that on the axis, can be calculated from formulae of varying degrees of complexity. If all these assumptions are valid it would be expected that the answers would be similar, at least for small ratios of slit width to wavelength, and indeed this is so as shown in Fig. 6. It can be seen that for small values of d/λ the curves (a), (b) and (c) agree quite well, and for the value of d/λ of 0.3 chosen, the 60° response should be within about one to two dB of that on axis. In practice this was by no means obtained; curve(d) shows the measured results and the discrepancy is gross. The question arose as to whether the slit was uniformly "illuminated", and going to an extreme, if all the sound were concentrated at the two edges the radiation pattern would obviously be different, and calculation gives curve (e) which is in better agreement with curve (d). However a quick test with a probe microphone showed that this energy distribution was not followed, in fact the sound pressure at the centre was slightly higher than that at the edges. In desperation the problem was then worked backwards and the apparent width of the source calculated; it turned out to be exactly the width of the cabinet for values of d/λ up to 0.7; the points are plotted as (f). It is now clear what is happening; the slit is indeed working as expected but because of this, sound energy flows along the front of the cabinet until it meets the discontinuity at the edges and is then re-radiated. The obvious moral is, to make the front of the cabinet as narrow as possible. As pointed out elsewhere[8] by the author this solution has other advantages from the aspect of structural resonances in the cabinet walls.

Above the frequency quoted, the slit tends to radiate on its own as shown by curve (d) approaching the calculated curves, but only for a short while, it then becomes more directional again. Neither is this the end. In the loudspeaker design mentioned the same slit width is used over the bass and middle frequency units in, of course, the same width cabinet. It might therefore reasonably be expected that the directivities of the two sources would be the same, but they are not. The radiation from the 8in middle frequency unit has a wider beam than that from the 12in bass unit. Time has not permitted the problem to be investigated further but it is clear that in practice the performance of slits is not as simple as would appear at first sight.

High frequencies

Let us continue this question of directivity but now include the high frequences. It has often been suggested in the literature that the variation of the spherical response with frequency is the most important feature of a loudspeaker. Methods of measuring this include the use of a reverberation chamber, measuring the polar response at various angles and frequencies in a free field room and calculating the result, and finally a method developed at the BBC by Gee[9] which uses an integrating meter to give a direct answer at any frequency or band of frequencies. The first method is limited in that it requires a room much larger than one to ISO standards to ensure adequate diffusion at the bass. The second method is rigorous if sufficient measurements are taken and if the free field room is adequately large[10], by no means always the case. It is however extremely laborious and time consuming, and is rarely used. The third method also relies on an adequate size free field room but is quite rapid. It has moreover the advantage over the first method that it is possible to weight sound coming from differing directions, e.g. sound from the front hemisphere relative to that from the rear.

This raises the whole question of what we are trying to measure and why. In the BBC the spherical response of a number of loudspeakers has been measured and efforts made to correlate it with sound quality in a live room, but with very little result.

When for example we listen in a room of normal reverberation time to a rather directional loudspeaker on its axis, it is common experience that the sound quality does not change drastically when in the near or reverberant sound field. On the other hand if we were really listening simply to the sound pressure at these two points then the direct response and the spherical response would indeed be the determining factors. Furthermore a similar factor must be involved in the fact that with such a loudspeaker in a live room the directional properties are clearly audible even when listening well into the reverberant field.

These experiences indicate clearly that the spherical response is not the predominating factor in determining sound quality under live listening conditions and to check this a formal experiment was carried out at BBC Research Department. A monitoring loudspeaker was taken having three units and representing as omnidirectional a device as was possessed at the time, and for comparison an 8in wide range unit representing as directional a device as was likely to be met. Listening on axis in a free field room and using

Fig. 4. Peak spectrum (octave bands) of middle and bass for various types of programme.

Fig. 5. Schematic frequency response of two unit loudspeaker; on axis and at 60° in horizontal plane.

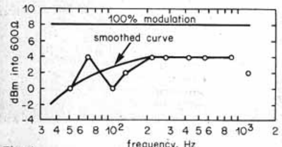


Fig. 4

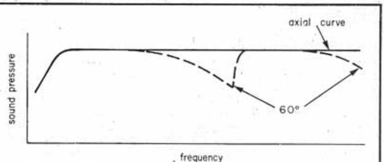


Fig. 5

speech and a team of experienced observers, the two were equalized by ear to sound as closely similar as possible. They were then transferred to a listening room well away from the walls; the room had a reverberation time of about 0.4s, and the loudspeakers were again compared, listening on the axis. The results in the two conditions were almost identical within the experimental error, although a small change towards the known spherical performance could be discerned but not guaranteed. The conclusion therefore was that it is essentially the direct sound which determines the sound quality and not the spherical response. The measurement of frequency response at various angles in a free-field room is therefore a much better indication of performance than the spherical response even when listening in the reverberant field, and this has been confirmed by careful listening tests many times since.

The question still arises however as to what is the optimum delivery and here a look at history is useful.

At one end of the scale, a loudspeaker developed by Harz and Kosters of NWDR[11] in 1957 used a bass unit facing upwards, and a middle and high requency cluster of no less than 32 units mounted on the surface of a sphere. This resulted in a very close approximation to an omnidirectional loudspeaker and gave a pleasant spacious image on orchestra. However, when an announcer spoke it sounded as if his mouth were six feet wide.

This design has been followed by another German design much more modest in outlook in which units are only mounted in the sides and front, none in the rear; in this design even the side facing units can be switched off leaving only the front ones, so it looks as though our experience was that of others too.

In the BBC we have gradually progressed from the opposite direction. The first loudspeakers were single cone wide-range devices which were very directional in the treble, and subsequent multi-unit designs have all tended to increase the angle of radiation at high frequencies and this has been approved by users. Of course over the years stereo has been introduced and this has involved other factors. For the last high quality loudspeaker designed in Research Dept. there were some vague suggestions that the angle of radiation might be too wide for stereo. Fig. 7 shows the axial and off-axis curves for the loudspeaker concerned. (For this discussion the bass cut should be ignored, and is due to the fact that the free interior volume of the cabinet is only 1/6 cu. ft.) It may well be therefore that any loudspeakers more omnidirectional than this will fail to provide first-quality stereo. In this discussion it has been assumed, of course, that a sharp stereo image is regarded as essential; these comments are not applicable where the stereo image is made rather diffuse over the whole seating area.

The next point to be discussed is the question of optimum axial frequency response. This question is not concerned with how wide a frequency range should be covered, but what shape the response curve should be. First of all the underlying assumption must be clearly stated. This is that both the microphone and all associated amplifiers have a uniform frequency response. The usual conclusion is that the loudspeaker should also have a uniform axial frequency response but this is precisely what is being challenged. Not even in stereo reproduction are the sound wave-fronts produced in a listening room similar to those heard in the studio or concert hall and it therefore seems clear that if by "bending" the axial response curve of the loudspeaker a more realistic psychological impression is obtained, then this is entirely justified. Thus, for example, if a uniform output is maintained at all frequencies an orchestra sounds extremely close.

This condition is quite unnatural and a much better sense of perspective is obtained if a slight dip in the 1 to 3kHz region is applied. About 2dB is sufficient to provide the more distant perspective without destroying the sound quality. It may well be that as techniques progress other such tricks will follow. All that is intended at this stage is to get away from the rigid idea that a uniform axial response is necessarily the best.

So far general trends have been discussed and it has been assumed that perfect units were available. As all designers know this is far from the truth and the question arises as to how far departures from the ideal can be made without perceptibly degrading the sound quality. This is also important from the aspect of listening in rooms which after all is where most listening is done. It has been shown that it is the direct sound from a loudspeaker that is predominant, but of course if a loudspeaker is close to a wall, then the near images may form part of the "direct" sound and will produce irregularities in response.

The loudspeaker can be regarded for this purpose as a two channel device with the two channels in parallel. To start with let us examine the case where the main channel has a uniform response and the other has a resonant circuit of variable Q (narrower than a critical band) whose output at resonance adds to that of the main channel and whose amplitude relative to it can be varied. The varying degrees of audibility at different frequencies and for differing Q has been established for pink noise in the form of the relative levels for the peak of the resonance and in the main channel. Now for *loudness* the energy in the critical band is summed and it appears as though this relation roughly holds too for degrees of colouration. Only roughly, for the actual law varies with the degree of colouration itself as shown in Fig. 8. It will be seen from this figure that there is a regular variation in the law with the degree of perception. The law for the "just perceptible" condition is close to the power law and the curve marked "definitely perceptible" is at about the limit of perceptibility for *programme* and is therefore the one we are most interested in. Note that the horizontal axis is not Q but reverberation time and that the vertical axis is dilution. This variation in law with perceptibility is in accordance with the findings of Kryter and Pearsons[12] in relation to the noisiness of a tone in noise and they also show that as the ratio of tone increases the noisiness increases faster than the total r.m.s. value of the critical band concerned. The general slope shown in the figure is confirmed in subsequent work by Moulana. The height of the corresponding irregularity in frequency response is shown in Fig. 9 for the "definitely perceptible" condition.

When narrow peaks are subtracted from the main channel, conditions are very different. Whereas for additive peaks the just perceptible condition was approached, as the amplitude was reduced, slowly and rather indefinitely, for the subtractive condition the colouration suddenly disappeared and it was immediately evident that a cancellation was taking place. This effect was, as would be expected, shown up in the standard deviation of the results; in one case the test team even returned the ultimate of zero spread. The implication of this effect is extremely important as it shows clearly that the subjects were in fact listening to the steady state condition; for it is evident that the time function could not be cancelled in this way. This is a very important distinction, as much earlier unpublished work by the author and supported by other unpublished work also at the BBC by Gilford[13] has shown an anomaly, namely that under certain conditions, which are not at all clear, the law of dilution with Q for a given perceptibility can go in precisely the opposite direction, that is the higher the Q the more obvious is the colouration. It seems highly likely that in these latter conditions it is the time function which is being observed.

Fig. 10 shows the height of an irregularity for a subtractive peak for the "definitely perceptible" condition which again closely corresponds with the "just perceptible" condition for programme. The curve is very different from the additive condition and the results are more nearly like the audibility of tone in wide band noise. In both cases given here dilution appears to be the fundamental factor rather

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than height of irregularity and the unfortunate conclusion is arrived at that even for the steady state condition the audibility of a narrow peak cannot be assessed unless the relative polarity is also known.

Now the question arises as to what happens outside the critical bandwidth. The loudness function is known to be different and this might also apply to colouration. The audibility of a resonant circuit with a Q of about 3, i.e. roughly ⅓ octave wide was examined for various frequencies of resonance using *programme* designed to be critical over the frequency band being tested. The results are given in Fig. 11 which shows the height of the peak for the just perceptible condition. During these tests, after first identifying the frequency of colouration, the subject was permitted to switch the resonance in and out of the circuit and to reduce the height of the peak until it was only just audible. Under these conditions the height of the peak is roughly independent of frequency except at the bass. Here we are, of course, inside the critical bandwidth, but this does not seem to be the essential factor, as experiments with octave bandwidth circuits show a similar shape curve. The standard error for the points in this curve is roughly ½dB.

If two contiguous circuits are used to form a plateau of twice the bandwidth, centred around a mid-band frequency, a "just perceptible height of 2.6dB is obtained and if the bandwidth is doubled again using four contiguous peaks a just perceptible height of 1.8dB is obtained. It is clear that some form of summation is taking place and extrapolation suggests a minimum audible level for wide band signal of about 1dB. However it is equally clear that, from the point of view of sound quality, this summation does not proceed indefinitely as it does with loudness. To take an extreme example if the entire range is raised by 10dB there is a large change in loudness but, by definition, none in sound quality. Furthermore if the entire spectrum except for the lowest ⅓ octave were raised the effect would not be described as an excess in most of the range but as a deficiency in the bass. One point to be noted was that even with the wide pleateau used, i.e. 1⅓ octaves wide, only one frequency of colouration was heard.

It has been seen that some form of summation is taking place over quite a wide frequency band and it is therefore pertinent to enquire how far apart two peaks must be before they are audible as separate entities.

For this test the same resonant peaks with a Q of 3 were used as before. The observer was instructed to increase the height of the peak at the reference frequency until the colouration was *clearly audible*. Successive peaks were then raised and lowered at one third octave intervals, to a height deemed by the observer to give maximum discrimination, until the frequencies of colouration of the two peaks were separately discernible; once again programme was used appropriate to the frequency range being covered. The results were quite astonishing; the mean values for the team of observers are given in Table 1.

Table 1

Ref frequency (Hz)	125	250	500	1k	2k	4k
Minimum distance apart of peaks, in octaves	1.5	1.3	1.3	1.1	1.1	1.0
Standard error of mean, in octaves	0.08	0.11	0.11	0.13	0.08	zero

The variation with frequency is interesting and may be due to the nature of programme spectrum. It will be appreciated that as the frequency increases the detailed structure of the spectrum becomes more and more random until at high frequencies it is not far removed from modulated random noise. The figure of one octave obtained in this part of the spectrum approaches the corresponding value which is obtained using pink noise as a source, instead of programme.

The remarkable result in Table 1 may possibly be the key to a number of previously unexplained phenomena. For example does it indicate why the irregularities in the sound field of a live room are not separately audible?

However, one conclusion is clear; if the loudspeaker contains a number of low Q resonances spaced closer together than one octave and covering the whole frequency range they should be inaudible. When a test was actually made of a series of peaks at intervals of ⅔rd octave at a level of 6dB above the base line, using a critical material such as speech it was found that they were in fact inaudible even on an A/B test. The conclusion is therefore correct, and the frequency response of such a characteristic is shown in Fig. 12.

If however the peaks are increased to a level of say 15dB, the sound becomes extremely coloured and it is evident from the character of the sound that a regular series is being heard. (The comment about irregularities in a room must include, therefore, the proviso that the frequency spacing is also irregular.) It should be noted however that the series being used is a logarithmic one not an arithmetic one and that the bandwidth of each peak is also a logarithmic function; however the ear still detects it as a regular series. The question therefore arises as to what constitutes a regular series, regular on what scale? Examples of series in loudspeakers include mis-terminated horns, a loudspeaker spaced away from one wall, a folded corner horn and a labyrinth.

If a frequency characteristic is listened to which is uniform up to 1kHz followed by the logarithmic series mentioned above, somewhat imitating a horn, the "fundamental" heard is clearly the lowest peak, in spite of the ⅔ octave spacing. It will be noticed however that the upper terms of the series are relatively inaudible and this prompts the question as to how many terms of a series are necessary to give this peculiar sound quality; experiment gives the answer of only three or four terms. If however the low end of the frequency scale has this series starting at 40Hz finishing at 500Hz and of uniform response thereafter, the position is now reversed. The most prominent colouration is at 500Hz and the lower, so to speak "fundamental," frequencies are relatively inaudible. What then constitutes the "fundamental" of the series? Now let us go further, if a complete series starting at 40Hz and finishing at 20kHz is listened to, using pink noise for convenience, it is found that on a high quality loudspeaker the main colouration is in the 600 to 800Hz region, no less than 15 times the "fundamental"!

Now how is a series detected? One obvious answer is by means of a scanning technique, and it would appear from the examples quoted above as if the scanning may work in both directions, from the bass upwards in frequency and from the treble down e.g. a triangular wave form. It follows that if there is a scanning mechanism, there also exists a corresponding time series, and it might be expected that this also would give rise to peculiar effects, and this is found to be correct. To take a well known example; if a person claps his hands under a bridge with the arches, say, 100ft apart, a series of pulses with a repetition frequency of about 10Hz is produced, nearly an octave below the lowest frequency we can hear. But what in fact is heard is a noise like a "twang," with a spectrum centred around, say, 1500Hz; no less than 150 times the fundamental! Nor is this an isolated example. In one studio in the BBC, under certain conditions the sound quality could, before remedial action was taken, become very hard. Reverberation time measurements give no clue to this effect at all. On one occasion however the audience balcony, which was rarely used, was entered, and on clapping, a flutter of less than 10Hz frequency was heard, and the connection was appreciated. That studio was being modelled[14] at the time and measures were taken to remove the flutter in the model. When corresponding modifications were carried out in the real studio the hard quality disappeared.

As a final example, in a sound control

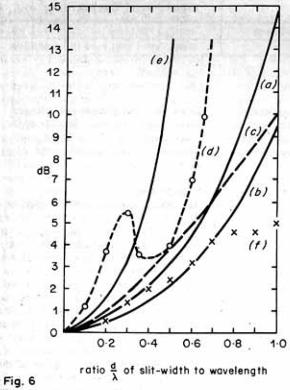

Fig. 6

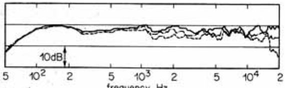

Fig. 7

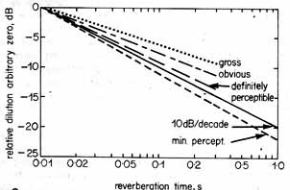

Fig. 8

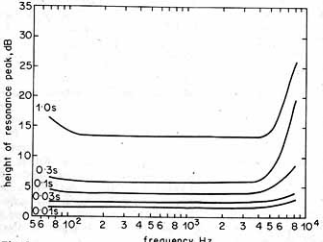

Fig. 9

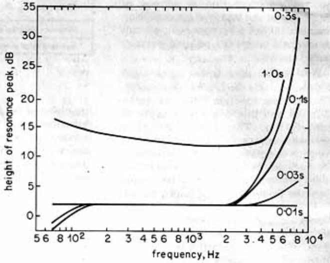

Fig. 10

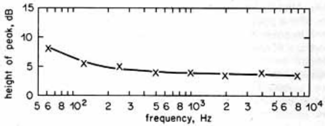

Fig. 11

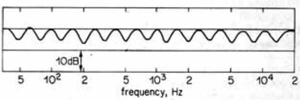

Fig. 12

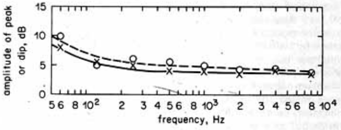

Fig. 13

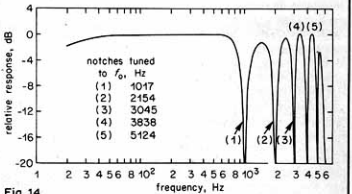

Fig. 14

room attached to one of the television studios a loudspeaker was suspended near a corner, and complaints were made of the sound quality. It was clear from a visit that the quality was indeed very peculiar and "tunnelly"; moreover, it varied considerably throughout the room. To check that it was not caused by the loudspeaker itself, this was lowered to the floor and it was shown that there the sound quality was quite satisfactory. A frequency response curve was taken with the loudspeaker back in place, taking precautions to eliminate as much of the reverberant field as possible. This curve showed definite evidence of a series. The loudspeaker was then lowered 35cm to try and break the series and a further measured curve showed that this had been successful. Under these conditions the sound quality was completely satisfactory and also now reasonably uniform through the room.[15]

To sum up, it is not at all clear how to define a series; it appears that it can be regular in hertz or octaves, but what about mels, and how regular is regular? Clearly however series should be avoided at all costs as there are no means of knowing in what part of the spectrum the subjective effect will occur.

Dips

It is now necessary to consider the effects of dips in the response curve on their own. It would be expected from perturbation theory that unless the hearing system is highly non-linear the magnitudes of dips would be similar to that of peaks, for the just perceptible conditions. Experiments were carried out to determine the just perceptible values for Qs of 3 as for peaks, except that the in/out switch was not used; the reason for this will be discussed later. The results are given in Fig. 13 together with the corresponding values for peaks. It will be seen that the two sets of values are closely similar, such differences as there are being in the direction that general experience would indicate. The depth for two contiguous dips forming a trough in the midband was 3.8dB, and for four contiguous dips was 2.5dB, again both slightly greater values than for the corresponding plateaux.

However, when the experiments with the four continguous dips were being carried out a further effect was noticed which had not been observed before. Particularly when the trough was clearly audible, in addition to the effect of the dip, the high frequency recovery to normal level was also clearly audible. Experiments to determine the narrowest trough for which this effect was noticeable gave a result of 1⅓ octaves exactly the same value as obtained for the minimum distance apart of two peaks for this part of the spectrum. The question arises as to whether this mythical scanning mechanism is again responsible, having the slow decay time we have postulated, a certain bandwidth for the trough being necessary before the fall is great enough to be audible. Furthermore a fast rise time was also suggested and under these conditions it is not surprising that two peaks should give the same separation as a trough.

Now this matter can be taken a little further; if the upper recovery in frequency response of the trough is removed completely so that instead of having a trough there is merely a step, it might be expected that the decay of the scanning mechanism should still register, and it does. Under these conditions the audibility of the spectrum near the step is definitely reduced whilst that somewhat higher in frequency appears to stand out in excess. This latter effect is not a new discovery, it has been known for at least 30 years and was a common feature in early single unit loudspeakers where it was known by the delightfully descriptive name of "disembodied top" as the upper end of the spectrum appeared to be separated from the main body by a gap.

Now narrow crevasses must be examined. It has often been stated that narrow crevasses are inaudible but it depends on the exact frequency of the dip. For example if it falls on the fundamental of a musical instrument the result can be disastrous. However, Professor Hill, formerly of the BBC Research Department, has shown[16] that if the frequency of the crevasse is offset by about a quarter tone from a fundamental, a narrow crevasse can indeed be almost inaudible. Figure 14 shows one extreme example he tested. The high frequency cut off of 6kHz was imposed for other reasons, and the frequencies of the crevasses are not simple multiples of one another. This appalling looking response was tested on subjects using as test material, male speech, piano music and dance music. The subjective mean grading in each case, where one unit represented "slightly worse than the standard" which also had a 6kHz cut off, was 0.6 for speech, 0.8 for music and an *improvement* of 0.3 for dance music; obviously the overall effect was quite small.

A/B testing

Now the alarming fact is that A/B testing may under certain circumstances give rise to completely wrong results when comparing the sound quality of two loudspeakers. If pink noise is used as a convenient source, and a deep narrow crevasse produced in it, it has been shown that the effect will be almost inaudible. If this is listened to for, say, half a minute as if programme were being used to judge a loudspeaker, and then the crevasse is switched out so that a uniform spectrum is produced, the ear will hear a strong colouration at the frequency of the crevasse. It seems that there are two mechanisms at work; the conscious one ignores the crevasse but the subconscious one detects it clearly. When the uniform condition is suddenly heard the subconscious mechanism comes forward and points out that there is now a considerable amount more sound energy at the frequency of the crevasse, and as that condition had been accepted as satisfactory the only conclusion to be reached is that there is now an excess in this region and that the sound must now be highly coloured. Transferring this to loudspeakers it is implied that if one with a crevasse is first listened to then it will probably appear that one with a uniform response is coloured.

Conclusions

There is a real danger of making loudspeaker unit surrounds too compliant as this can give rise to non-linearity distortion of high orders at quite low levels.

Equalization at the bass under whatever name it is called must be applied with full regard for associated power requirements or distortion may occur.

To obtain uniform response at various angles in the mid-band region a narrow-fronted cabinet is called for. Slits can be very useful but their action is obviously considerably more complex than appears at first sight.

The sound quality of a loudspeaker is determined much more by the direct response at any given angle than by the spherical integrated response, and at any rate for stereophonic purposes there may well be a degree of omnidirectionality beyond which it is inadvisable to go.

Fig. 6. Directivity of a slit; response at 60° relative to that on axis.
Fig. 7. Frequency response of a miniature loudspeaker at various angles to axis.
Fig. 8. Variation of law of addition with subjective degree of colouration.
Fig. 9. Height of irregularities due to additive peaks for a definitely perceptible condition, using pink noise.
Fig. 10. Height of irregularities due to substractive peaks for definitely perceptible condition, using pink noise.
Fig. 11. Height of irregularities due to additive peaks having a Q of 3 when listened to one at a time; for just audible condition, using critical programme.
Fig. 12. Response curve showing nature of inaudible irregularities when listened to together.
Fig. 13. Height of irregularities due to dips in response for a Q of 3 when listened to one at a time, for just audible condition, using critical programme; curve of Fig. 11 added for comparison.
Fig. 14. Frequency response of transmission chain used by Prof. Hill.

A plea is made for non-uniform axial frequency response insofar as it assists greatest realism overall.

Additive narrow peaks in response appear to add up on a roughly r.m.s. basis but subtractive ones appear to obey a different law. Dilution of the peaks relative to the main channel appears to be the fundamental factor rather than height of irregularity. Wider peaks of the same relative polarity add upon a rather different basis and the frequency discrimination for colourations is astonishingly poor.

Series are not yet fully understood, but the indications are that they should be avoided at all costs.

Dips in response appear to have little or no effect on contiguous peaks either inside the critical band or outside it.

Isolated dips obey similar laws to peaks, and narrow crevasses can be inaudible if they avoid fundamentals.

A/B tests of sound quality are found to have pitfalls and appropriate measures should be taken where necessary as have been indicated.

In a number of these phenomena there is a suggestion that a scanning mechanism may be at work and that it may operate in both directions, i.e. from the bass up and from the treble zone. This could also account for the fact that if a step in the response curve is produced the corner of the step is always audible whether the step is up or down. This suggestion immediately raises the questions of what is the scanning repetition rate, is the scanning linear and if so on what scale, hertz, octaves or mels, and what are the rise and decay times?

Finally it should be appreciated that only a few of the effects which go to make up sound quality have been mentioned but all these effects appear to be used simultaneously.

The views expressed here are based on experience within the BBC. Some of the conclusions are drawn from limited evidence and not all engineers within the BBC would necessarily agree with all of them. It spite of all that has been said, in the final decision, a good loudspeaker remains a matter of personal choice. However, experiment and analysis help us to make this choice.

Thanks to my colleagues for bearing so patiently in the experiments and to the Director of Engineering of the BBC for permission to publish.

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The following list of manufacturers and their addresses is not a definitive guide to producers of high quality loudspeakers but is provided by *Wireless World* as a help to readers.

Manufacturers

Acoustical Manufacturing Co. Ltd, St. Peter's Road, Huntingdon, PE18 7DB.
Acoustic Research International, High St., Houghton Regis, Beds. LL15 5QJ.
Altec Sound Products Ltd, 17 Park Place, Stevenage, Herts.
Bang & Olufsen (UK) Ltd, Eastbrook Road, Gloucester GL4 7DE.
Bose (UK) Ltd, Milton Regis, Sittingbourne, Kent.
B & W Electronics, Meadow Road, Worthing, Sussex, BN13 1QA.
Cambridge Audio Ltd, Lamb House, Church Street, London W4 2PB.
Cerwin Vega (UK), 281 Balmoral Drive, Hayes, Middx.
Celestion, Ditton Works, Foxhall Road, Ipswich, Suffolk IP3 8JP.
Chartwell Electro Acoustics Ltd, Alric Avenue, London N.W.10.
Eagle International, Heather Park Drive, Wembley, Middlesex HA0 1SU.
Gale Electronics & Design Ltd, 39 Upper Brook Street, London W1Y 1PE.
Goodmans Loudspeakers Ltd, Downley Road, Havant, Hampshire PO9 2NL.
Griffin, H. K. & Co. (Electronics), Siddons Factory Estate, Howard Street, West Bromwich, Staffs.
Gulton Europe Ltd, The Hyde, Brighton, Sussex BN2 4JU.
Hayden Laboratories Ltd, Hayden House, 17 Chesham Road, Amersham, Bucks HP6 5AG.
Hitachi Sales (UK) Ltd, Hitachi House, Station Road, Hayes, Middx. UB3 4DR.
IMF Electronics Ltd, Westbourne Street, High Wycombe, Bucks.
Jordan-Watts Ltd, Benlow Works, Silverdale Road, Hayes, Middlesex UB3 3BW.
KEF Electronics Ltd, Tovil, Maidstone, Kent ME15 6QP.
Lansing, James B., C. E. Hammond & Co. Ltd, Lamb House, Church Street, London W4 2PB.
Leak, Rank Radio International Ltd, P.O. Box 596, Power Road, Chiswick, London W4 5PW.
Lecson Audio Ltd, Burrel Road, St. Ives, Hunts PE17 4LE.
Lowther Acoustics Ltd, St. Mark's Road, Bromley, Kent BR2 9HQ.
Macinnes Laboratories Ltd, Stonnam, Stowmarket, Suffolk, 1P14 5LB.
Marantz, Pyser Ltd, Fircroft Way, Edenbridge, Kent TN8 6HA.
Millbank Electronics Group, Bellbrook Estate, Uckfield, Sussex, TN22 1PS.
Monitor Audio, 347 Cherry Hinton Road, Cambridge CB1 4DJ.
Mordaunt-Short Ltd, Durford Mill, Petersfield, Hampshire, GU31 5BB.
Nordmende, H. Vesshof & Co. Ltd, Unit 4, Blackwater Way, Ash Road, Aldershot, Hants GU12 4DL.
Omal Group Ltd, Omal House, North Circular Road, London NW10 7UF.
Philips Electrical Ltd, Century House, Shaftesbury Av., London WC2H 8AS.
Photax (London) Ltd, Hampden Park, Eastbourne, Sussex.
Pioneer, Shriro (UK) Ltd, Shriro House, The Ridgeway, Iver, Bucks SL0 9JL.
Quad, Acoustical Manufacturing Co. Ltd, St. Peter's Road, Huntingdon, PE18 7DB.
Quasar, Quasar Division, Precision Centre, Heather Park Drive, Wembley, HA0 1SU.
Radford Audio Ltd, Ashton Vale Road, Bristol, BS3 2HZ.
Rank Audio Products Ltd, P.O. Box 70, Brentford, Middx.
Regent Acoustics, Carrington House, 130 Regent Street, London W1R 6BR.
Sansui, Vernitron Ltd, Thornhill, Southampton SO9 5QF.
SMC, Monitor Distribution Co. Ltd, 76 Bedford Road, Kempston, Beds, MK42 8BB.
Sonab Ltd, P.O. Box 4, Oldfield Road, Hampton, Middlesex, TW12 2HN.
Spendor Audio Systems Ltd, Unit 12, Station Road Industrial Estate, Hailsham, Sussex.
Stereostage, Nucleus, 22 Hyde Green, Marlow, Bucks.
Studio Craft, Acoustico Enterprises Ltd, Unit 7, Space Waye, North Feltham Trading Estate, Feltham, Middlesex, TW14 0TZ.
Tannoy Products Ltd, Norwood Road, West Norwood, London SE27 9AB.
Telefunken, AEG Telefunken (UK) Ltd, Bath Road, Slough, Bucks.
Yamaha, Natural Sound Systems Ltd, Strathcona Road, North Wembley, HA9 8QL.