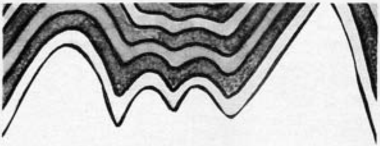
using an equaliser



Although there are many different types of equaliser, they all perform the same basic task, namely the correction of deficiencies in the frequency response of one's speaker system and/or listening environment. As such they represent an extremely useful tool in the quest for 'perfect' hi-fi. Unfortunately, however, equalisers are all to often misused, and in extreme cases actually do more harm than good. The following article takes a look at the various types of application for which equalisers are most suited, and also explains how to get the best out of this versatile instrument.

The great advantage of an equaliser is that, unlike conventional bass and treble tone controls, which can provide only a fairly limited amount of boost or cut at the extremes of the audio spectrum, it is possible to iron out (equalise) peaks or dips in a response over the entire range of audio frequencies. Not only that, but with a *parametric* equaliser, the centre frequency, Q and gain of the equaliser filters can all be tailored to exactly compensate for non-linearities in the response of any given system.

Although the use of equalisers was originally limited to professional sound recording studios, their undoubted benefits have led to an increasing number of amateur applications: dedicated hi-fi enthusiasts, having lavished considerable attention and expense on cartridges, pick-up arms, turntables, amplifiers and loudspeakers, are now resorting to equalisers to 'upgrade' the last link in the audio chain, namely the listening room.

Unfortunately, however, many amateurs fail to make the most of the facilities offered by a sophisticated parametric equaliser, and simply end up using it as a sort of 'super-duper' tone control, twiddling the knobs to get a bit more bass here, less treble there and so on. This article is therefore intended to provide a few insights on how to achieve effective room equalisation, whether it be for domestic or PA-system applications.

Equalising your living room

In recent years the subject of room equalisation has become something of a fad. Various audio design consultants and well-known manufacturers of audio equipment have conducted extensive research into the response of domestic listening environments. Bruel and Kjaer, for example, offer a comprehensive measurement and equalisation system for listening rooms, whilst Philips loudspeakers are specially designed to compensate for the deficiencies of the 'average living room'. The subject of room equalisation, with particular reference to the effect of the placement of loudspeakers, has been discussed in a spate of recent articles, and numerous hobbyist magazines have produced designs for (graphic) equalisers. There is no doubt that people are now generally aware of the effect of the shape and contents of the listening room on the reproduction of the audio signal.

That the room has considerable effect is hardly surprising, especially when one considers how much care and attention is paid to the internal construction of loudspeakers (bracing ribs, damping

1

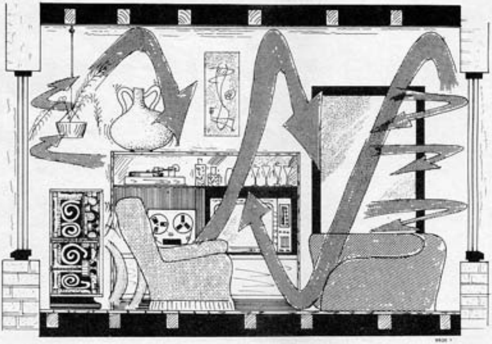


Figure 1. Considerably more attention is spent on the internal design of loudspeakers than on the interior of one's living room, despite the fact that the latter has a profound effect upon the sound of the music signal being reproduced.

materials, air-tight seals etc.): in a sense, the listening room is simply a giant loudspeaker cabinet, in which the listener sits. However, as a rule little or nothing is done to improve the response of the room. Of course it is possible to take such steps as to change the curtains, fit wall-to-wall carpeting, experiment with different loudspeaker placings, swop the furniture around etc. Although whether the living room will remain liveable-in is another question!

A simpler solution to the problem of 'upgrading' your living room is to employ an equaliser, which will compensate for the inherent deficiencies in the room's frequency response.

Assuming, for example, that the room in question has the response shown in figure 2a. Using an equaliser the response of the audio system can be tailored to look like that shown in figure 2b, i.e. the inverse of the room's response, with peaks at 1600 Hz and 4 kHz, dips at 50

2

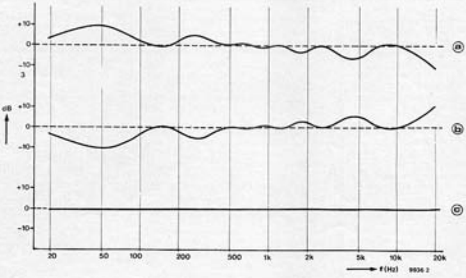
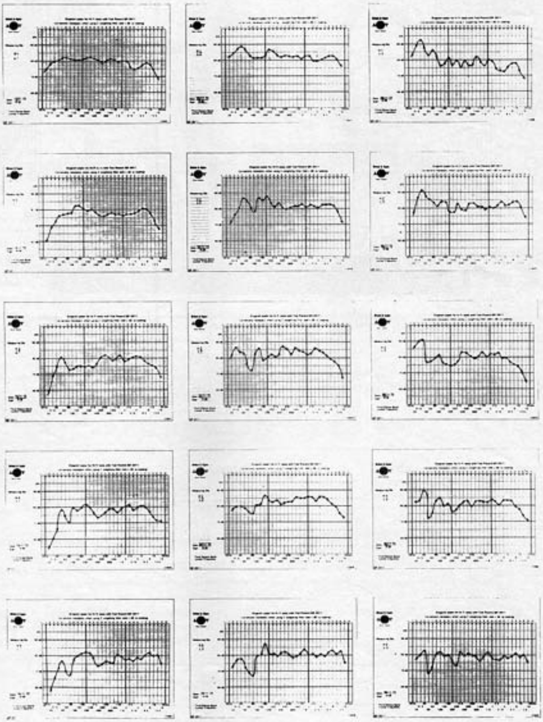


Figure 2. An example of how, in principle, it is possible to obtain a uniform frequency response with the aid of an equaliser. The irregular response of figure (a) is smoothed out by setting up the inverse response (shown in figure (b)) on the equaliser filters. The result (figure (c)), in theory at least, is the desired perfect reproduction.



A page from Bruel and Kjaer application note 13-101, which throws an interesting light on the topic of room acoustics. The frequency responses shown here were measured using 5 different loudspeakers, set up in 3 different living rooms.

3

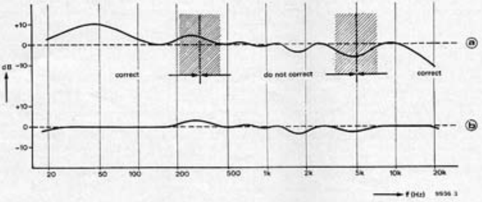


Figure 3. In hi-fi applications it is neither necessary nor indeed advisable to attempt to iron out every single peak and dip in the response. In particular, the band of mid-range frequencies between approximately 300 Hz and 5 kHz is best left untouched, so that the resultant corrected response will look something like that shown in figure 3b.

and 250 Hz and treble boost above 10 kHz. Thus, in theory, the resulting combined frequency response (i.e. that which, so to speak, reaches the ears of the listener) should be the perfectly flat line shown in figure 2c.

Unfortunately, however, as one might expect, things are not quite so simple in practice. The situation is complicated by the fact that the signal which reaches the listener is a mixture of direct and indirect sound. The direct sound is that which travels straight from the loudspeakers to the listener's ears, whilst the indirect sound is that which has first been reflected off the walls, ceiling, floor and furniture. It is the indirect sound, therefore, that is 'coloured' by the acoustics of the rooms. This fact has two consequences:

The relative proportions of direct and reflected sound will vary at different points in the room. Due to path length differences between the direct and indirect signals, either phase cancellation or phase reinforcement may occur, creating nodes and anti-nodes at *different* locations in the room. For this reason it is only possible to equalise the frequency response of a particular listening position. If that position is altered the frequency response will have altered also.

4

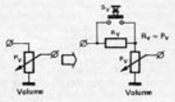


Figure 4. In most cases it is a relatively simple affair to incorporate a switch selectable 6 dB attenuator into a P.A. system. A resistance R_V of approximately the same value as the volume control (P_V) is connected in series with the latter, and a pushbutton switch S_V is then connected in parallel with the resistance.

Secondly, the human ear responds differently to direct and reflected sound, particularly at frequencies within the vocal spectrum between roughly 300 Hz and 5 kHz. The direct sound is recognised as the primary factor determining the 'quality' of the sound source, whilst the reflected sound provides information relating to the listening environment. Excessive equalisation can therefore lead to highly undesirable results, namely strong colouration of the direct sound in an attempt to compensate for a reflected signal heavily influenced by the room acoustics. As already mentioned, careless or over-enthusiastic use of an equaliser can do more harm than good. However the prospective user should not be put off by this fact, since an equaliser can offer tangible benefits to the hi-fi enthuisiast who, for practical reasons, is constrained to listen to his system in a small and acoustically-poor room, with his speakers positioned in non-ideal locations.

The advantages of an equaliser can be illustrated by taking a closer look at the frequency response of a typical living room, as shown in figure 2a. The same curve is shown again in figure 3, with several 'critical' areas emphasised. For the band of frequencies from roughly 300 Hz to 5 kHz, the golden rule is 'leave well alone' (assuming that it is the acoustics of the room and not deficiencies in the response of the loudspeakers which are responsible for irregulatities in the response). However peaks and dips in the response which occur at frequencies outside this band can be flattened out with the aid of an equaliser; at frequencies which are at the junction of these regions (i.e. around 300 Hz and 5 kHz), limited equalisation may be useful in certain cases. What this means for the response curve of figure 3a is this:

- the prominent resonance at around 50 Hz can be completely eliminated (that this also results in an improvement of approximately 10 dB in the signal-to-noise ratio is an added bonus).
- The smaller peak at around 250 Hz lies in a transitional area, thus partial equalisation is possible, if desired. The most sensible procedure is to audibly compare the results obtained with and without equalisation.
- The barely perceptible 'bump' at 150 Hz is really too small to be worth considering; furthermore it lies right in the middle of the critical mid-range of frequencies and should therefore be left untouched.
- The dip at around 1600 Hz is likewise inside the critical vocal spectrum which should be avoided.
- The somewhat larger dip at approximately 5 kHz straddles the second crossover area, thus once again a partial or limited equalisation may prove worthwhile.
- Finally, the roll-off in the response above 10 kHz can legitimately be corrected with the equalizer; care should be taken not to apply excessive amounts of boost, however, since there is the danger of damaging one's tweeters (!)

After the above corrections have been carried out (and assuming that the dip at around 1600 Hz is the result of the room acoustics and not one's loudspeakers), the overall response which is obtained, should look something like

5

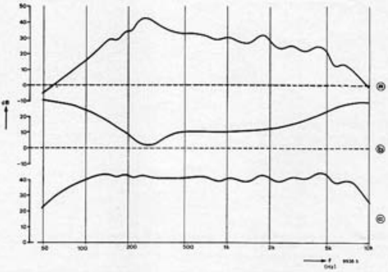


Figure 5. The frequency response of P.A. systems is frequently fairly poor. That shown in figure 5a is a typical example. With relatively simple equalisation, however, (figure 5b) one can obtain a response like that shown in figure 5c, which in practice improves the quality of reproduction to a quite amazing degree.

that shown in figure 3b — and hopefully there should be a correspondingly discernible improvement in the resulting sound!

As the above example illustrates, it is not necessary to make a large number of corrections in order to obtain an 'acoustically' flat response. All that is required in this example is a circuit to provide treble boost, and three variable resonance filters — in fact those facilities offered by the type of parametric equaliser

The following paragraphs describe how to go about actually setting up an equaliser for optimum results in a variety of practical situations.

P.A. systems

P.A. systems used in conference halls and auditoria are usually installed by professionals. However there are many situations such as local community meetings, school prizegivings etc. where smaller halls have to be set up acoustically by comparative 'amateurs'.

The most common problems encountered in this type of case are 'lack of intelligibility', 'not loud enough', and persistent acoustic feedback. Before explaining the main causes of these problems a few preliminary remarks on the nature of P.A. systems would not go amiss. The primary aim of a P.A. system is not to achieve 'high-fidelity' reproduction, but rather optimum intelligibility. Unfortunately, in practice this is often confused with maximum volume. Of course, in some cases intelligibility can be improved by bumping up the volume, but it is often true, particularly in badly designed or wrongly set-up systems, that increasing the output from your speakers simply produces the dreaded acoustic feedback or 'howlround'. One must therefore attempt to (a) make the system less susceptible to feedback, and (b) search for other ways of improving intelligibility than simply winding up the volume control.

To take the problem of acoustic feedback first: most people know that this irritating phenomenon is caused by sounds from the loudspeakers being picked up — either directly or via reflections off the walls, ceiling, etc. — by the microphone(s). These are then amplified, fed back to the loudspeakers, only to be picked up once more by the microphones, and so on until a nasty high-pitched howl is produced (hence the name 'howlround'). In order to increase the volume without provoking this unpleasant effect, the only answer is to ensure that less of the loudspeaker signal is picked up by the microphone(s). This can be done in several ways:

- by using directional (cardioid) microphones, which are less sensitive to sound from the rear.
- by using loudspeakers which also have a directionally dependent response. It is probably not so well known that cardioid loudspeakers exist. By positioning these with their backs to the microphones, acoustic feedback can be considerably reduced.
- by not positioning the loudspeakers right next to the microphones. This may appear rather an obvious point, but it is surprising how many people fail to observe this elementary precaution.
- by setting the output level of those speakers which are nearest the microphones lower than that of speakers situated further down the hall. Many loudspeakers already have a facility for reducing the output level; in those that do not it is a simple matter to incorporate a small value series resistor to provide the desired level of attenuation. This step may at first appear a little self-contradictory, however it allows the amplifier volume to be turned up without significantly increasing the feedback signal to the microphones.
- at any given time, do not have more microphones switched on than is necessary. If there is only one person speaking, then one microphone is all that is required. Switching additional mikes on will simply increase the chance of feedback.
- ensure the volume control is adjusted correctly! This may also appear to be rather an obvious point, but in practice it is often more difficult to observe than it may seem. The following couple of tips should help:

— acoustic feedback is more liable to occur in an empty hall than in a full one. For this reason it is often sufficient to adjust the volume control so that the system is just on the point of 'howlround', with an empty hall. Once the hall has filled up the volume setting should prove spot-on.

– The difference between a correct volume setting and one which is just on the verge of howlround is about 3 to 6 dB. It is often possible to tell when a system is on the verge of howlround by the fact that it sounds decidedly 'echoey' – the effect is slightly similar to that obtained with artificial reverberation units. One can capitalise on the above fact by incorporating a switched 3 to 6 dB attenuator in series with the volume control (see figure 4). With the attenuator switched out of circuit, one first adjusts the volume control unit the P.A. system just starts to howl-round (bear in mind that acoustic feedback builds up gradually), then one simply switches in the attenuator, and the system should be ready for use.

Once acoustic feedback has been reduced to a minimum, the next step is to attempt to increase the intelligibility of the P.A. system without recourse to the volume control. There are basically two main ways of doing this: reduce the amount of reverberation generated in the hall, and improve the quality of the sound itself. The former point basically boils down to improving the acoustics of the hall by installing heavy curtains, thick carpeting, etc., and unfortunately is normally fairly expensive. The second measure, i.e. improving the reproduction of the speech signal is where electronics, in the shape of an equaliser, come in. It is not generally appreciated that the *quality* of the reproduced sound signal plays an important part in determining its intelligibility. It has been proven time and again in practice that a flat frequency response over a reasonably wide spectrum – roughly 100 Hz to 10 kHz – will lead to a considerable improvement in the intelligibility of the average P.A. system. Unfortunately, however, there are a number of prevalent misconceptions regarding the ideal frequency response and how to obtain it. These have led to the appearance of such monstrosities as bass cut 'speech switches' which roll off the response below 200, 300 or even 400 Hz, special 'speech' (loudspeaker) cabinets, which often have a truly horrific response, and speech microphones (whose response is sometimes little better that that of the loudspeakers). All that is needed is for the bass tone control on the amplifier to be set to minimum and the 'presence filter', which, more likely than not, has also found its way into the P.A. system, to be switched in, and one has all the ingredients for a full-scale acoustic disaster!

Figure 5a shows the measured response obtained from such a set-up, with the tone controls set to their mid-positions(!).

Using a simple parametric equaliser, the attempt was then made to iron out the grosser irregularities by employing the filter response shown in figure 5b. The resultant overall response is shown in figure 5c. What cannot be shown however is the amazing improvement in the intelligibility of the sound signal as a consequence of this measure. Whereas previously the speaker could barely be understood in an extremely quiet environment, after the equaliser had been used every word was clearly intelligible even with the noisiest of audiences.

Practice has proven that an equaliser is an extremely useful and effective tool for obtaining clear and readily comprehensible reproduction when working in halls with difficult acoustics. However, the way in which an equaliser is used in P.A. applications differs from that when employed with domestic hi-fi systems. It has already been stated that, when equalising the response of an audio chain and/or listening environment, the band of frequencies between roughly 300 Hz and 5 kHz should be left well alone. In the case of a P.A. installation, however, almost exactly the opposite is true: precisely this range of frequencies between 300 Hz and 5 kHz – or to be more accurate, the slightly broader band of frequencies between 100 Hz and 10 kHz – should be corrected with the equaliser. The extremes of the audio spectrum are of little significance for the intelligibility of the resultant speech signal.

Furthermore, whether the response of the reproduced signal is *completely* flat or not is also of secondary importance. For example dips in the response of up to 4 or 5 dB will often have little audible effect. The crucial factor as far as P.A. systems are concerned, is the presence of large resonant *peaks* in the response, since the highest peak effectively determines the maximum setting of the volume control which can be used without causing howlround. Consequently, the equaliser should be employed to ensure that all the peaks in the system's response are on the same level. This process is illustrated in figure 6. Although, at first sight, the response curve of figure 6a may appear to be slightly better, in practice superior results will be obtained with the curve in figure 6b. Of course, as it stands the latter response is far from perfect, and with judicious filtering it is possible to achieve the optimum response shown in figure 6c.

For those readers who are still less than convinced as to the advantages of an equaliser in this type of application, it may be worth pointing out that the cost of a (home-built) equaliser is nothing compared to the price of new microphones or speakers.

Electronic music

A less common but nonetheless important area of application for equalisers is in electronic music, where their flexibility and tone-shaping capabilities make them a useful addition to electronic synthesisers and organs. In direct contrast to the procedure

6

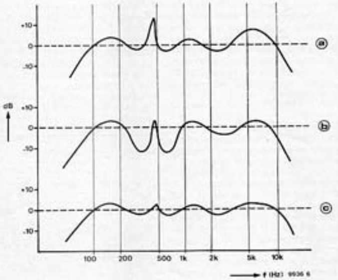


Figure 6. In the case of P.A. systems the equaliser should be set up such that all the peaks in the response have approximately the same amplitude. Although the curve in figure 6a may at first sight appear the better of the two, the fact is that the response of figure 6b will give superior results in practice. That is of course not to say that the latter represents an ideal case; using the same filters it is also possible to obtain the response shown in figure 6c.

7

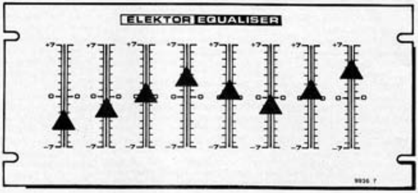


Figure 7. The 'graphic' equaliser owes its name to the fact that by arranging the (slide) potentiometers for the filter controls in a row across the front panel they provide an immediate graphic representation of the frequency response of the equaliser.

adopted in domestic hi-fi and P.A. applications, the filter parameters are not preset and thereafter left untouched; rather the filter settings are varied constantly as demanded by the (live) performance of the passage of music being played. For this reason the filter controls on the equaliser must be well-calibrated and ergonomically designed — a precondition which has led to the popularity of graphic equalisers, where the pattern of the slider potentiometers on the front panel provides immediate visual feedback regarding the overall filter response (see figure 7). However that is not to say that parametric equalisers are unsuited for this type of application — quite the reverse. Their greater scope (control of all the filter parameters) renders them much more flexible and affords the skilled user the possibility of achieving a wide range of different effects.

Setting up an equaliser

Before discussing the specific problems encountered when attempting to equalise the frequency response of domestic hi-fi and P.A. systems, there are several general points which can be made.

Firstly, and most importantly, it is essential that the frequency response which is to be corrected is already known. At the risk of sounding repetitive, fiddling around with the equaliser controls and 'playing it by ear' will almost certainly produce little in the way of tangible benefit and more likely than not will do more harm than good. However, measuring the frequency response in question is not such a fearsome undertaking as one might imagine and worried readers should banish any ideas about expensive Bruel and Kjaer measuring equipment that might be needed. In fact all that one requires is the audio spectrum analyser described elsewhere in this issue, a little patience, and a certain understanding of what one is trying to achieve. The point here is that exceptionally precise filter settings (within ± 0.5 dB) are not necessary, nor does one have to have an *absolutely* accurate picture of the frequency response. It does not matter whether a particular peak or trough happens to occur at *exactly* 225 Hz — what is more important is that irregularities in the frequency response can be detected (without necessarily knowing their precise location) and then corrected. Frequency response curves such as those shown in figures 2, 3, 5 and 6 may well be be interesting for the audio consultant or engineer, but as far as the hi-fi owner is concerned the only thing that counts is the sound reaching his ears!

The measurement and correction procedure for a domestic listening room can be carried out in a number of ways, although in each case the general principles involved are the same. The choice is basically one of ancillary equipment, whether one uses a measurement microphone, headphones, test records etc.

Setting up an equaliser for a P.A. system is somewhat simpler in that it only makes sense to utilise the existing microphone(s) to obtain the results of the spectral analysis. Since this step in fact forms the basis of the various procedures which can be adopted with domestic hi-fi systems we shall examine it first, before going on to discuss how to obtain the best results from an equaliser in domestic audio applications.

8

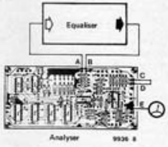


Figure 8. Before the equaliser is incorporated into the P.A. system it must first be adjusted for a flat response. This can be done with the set-up shown here.

P.A. systems

It goes without saying that, as far as possible, the performance of the P.A. system should be optimised before the equaliser is introduced. That is to say that the positioning of the microphone(s) and loudspeakers should be carefully chosen; ideally, cardioid microphones should be used, and, if necessary, the output level of the frontally situated speakers lowered. Only when no further improvements of this nature can be achieved should the equaliser be brought in. The setting-up procedure discussed here assumes that one possesses a parametric equaliser and the audio spectrum analyser

The procedure followed with an octave or third-octave graphic equaliser is broadly similar; any differences will be mentioned as they arise.

1. The first step is to adjust the equaliser controls to obtain a linear frequency response. This is done by connecting the noise generator direct to the equaliser input and the analyser filter and display to the output of the equaliser (figure 8). The analyser filter should be adjusted for

maximum Q (1/12 octave bandwidth). With this arrangement it is a simple matter to trace and correct any peaks or dips in the response which are caused by the equaliser itself (the filter sections of a graphic equaliser should be adjusted one at a time).

2. One next has to find a suitable point in the amplifier at which to connect the equaliser. If the amplifier has a monitor input, then in most cases one need look no further (see figure 9a). Figures 9b and 9c however, illustrate how it is possible to incorporate a monitor switch oneself.

9

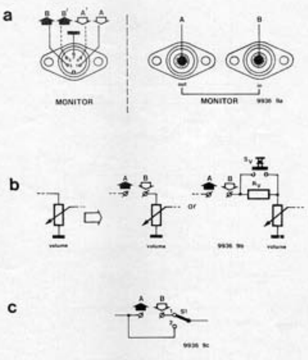


Figure 9. It is necessary to find a suitable point in the amplifier at which one can connect the equaliser. In general this will be in the region of the volume control. If the amplifier already possesses a monitor input then this can be used.

3. The output of the equaliser should then be connected to point B in figure 9, the noise generator connected to the equaliser input, and the analyser filter and display to point A in figure 9. This arrangement is depicted in figure 10.

4. The frequency response of the system can now be measured; first of all however, it is important that the potentiometer control which sweeps the centre frequency of the analyser filter up and down the audio spectrum has been provided with a (calibrated) scale (from, say, 1 to 10). If several microphones are used in the P.A. system under test, only the main mike, i.e. the one used most often, should be switched on. The results obtained can be plotted to form a graph such as that shown in figure 11a. The points most worth plotting are the highest values of a peak and the lowest of a dip. If an octave or third-octave equaliser is used then the analyser filter should be varied stepwise in octave or third-octave increments. The readings obtained for each frequency band are then plotted as shown in figure 12a.

5. Using a ruler one then draws a line approximately mid-way between the highest peak and lowest dip (see figures 11 b and 12b); this represents the theoretically ideal response to which one is approximating.

6. The Q of all the bandpass filters in the parametric equaliser are set to maximum (if a graphic equaliser is being used points 6 to 13 are omitted) and using the analyser filter the first peak or dip in the measured response is located; in figure 11b for example, this is the peak between measurement points 2 and 3. Since it is a peak, the first equaliser filter is set for maximum cut and the centre frequency of the filter slowly adjusted until there is a (fairly sudden) drop in the analyser reading. The centre frequency of the equaliser filter is then fine-tuned until the reading on the analyser display is at a minimum. Finally, the attenuation of the filter is reduced to the point where the meter reading coincides with the theoretically uniform response.

7. The analyser filter is then tuned up the audio spectrum until the next irregularity in the response is encountered. If, as in figure 11b, this is a dip, the second equaliser filter is set for maximum boost, tuned in to the appropriate frequency, and the gain of the filter varied until the desired reading on the analyser meter is obtained. If further deficiencies in the frequency response exist, this procedure is then repeated with the remaining equaliser filters.

10

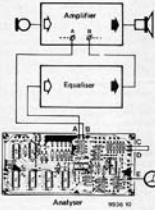


Figure 10. Once the equaliser has been adjusted for a flat response and a suitable connection point in the amplifier has been found, the analyser and equaliser are connected as shown.

8. The next step is to tune the analyser filter to the frequency at which the bass response of the system begins to roll off sharply. This point is indicated with an arrow in figure 11b. The Baxandall bass control on the equaliser should then be set for maximum cut, and its 3 dB point adjusted until the meter reading falls to 0.7 of its original value.

9. The turnover frequency of the treble filter in the tone control network is adjusted in exactly the same way. Were one to measure the resultant overall response (not that this is necessary), it would look roughly like that shown in figure 11c.

10. The centre frequency of the analyser filter is now tuned down to the point just below that at which the turnover frequency of the bass control was adjusted. The gain of this filter should then be increased until it coincides with the theoretical 'flat' value. The same procedure is performed for the treble control.

11. The analyser filter is tuned to a frequency on the 'flank' of the first

11

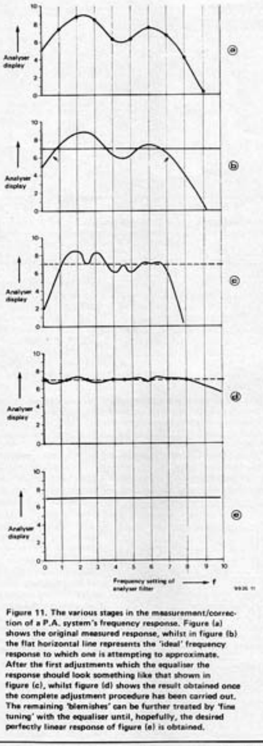


Figure 11. The various stages in the measurement/correction of a P.A. system's frequency response. Figure (a) shows the original measured response, whilst in figure (b) the flat horizontal line represents the 'ideal' frequency response to which one is attempting to approximate. After the first adjustments which the equaliser the response should look something like that shown in figure (c), whilst figure (d) shows the result obtained once the complete adjustment procedure has been carried out. The remaining 'blemishes' can be further treated by 'fine tuning' with the equaliser until, hopefully, the desired perfectly linear response of figure (e) is obtained.

peak or dip in the response and the Q of the first equaliser filter is reduced until the reading of the meter at this point reaches the nominal 'ideal' value. This procedure is repeated for the rest of the equaliser filters.

12. Theoretically, the equaliser should now be set up correctly and the response curve of the system should resemble that shown in figure 11e, i.e. flat over the range of the spectrum analyser. Unfortunately, however, this will rarely be the case in practice, and it will be necessary to repeat the above procedure from point 4 onwards in a slightly modified form. The reason for this can be explained if one looks at the curve shown in figure 11d, which represents the probable frequency response obtained so far. The curve exhibits the following faults:

– The turnover frequency of the bass tone control is too low, with the result that the response slopes too sharply at this point. The remedy – increase the turnover frequency and reduce the gain slightly.

– The centre frequency of the first (equaliser) bandpass filter is too high, the consequence being that the filter introduces too much attenuation and has too large a bandwidth. Each of these filter parameters should therefore be adjusted.

– The second bandpass filter is correctly adjusted, however the centre frequency of the third is slightly low, causing over-attenuation and resulting in too small a bandwidth.

– The turnover frequency of the treble control is too low, causing the response to roll off at high frequencies; once again this should be corrected.

13. With an octave or third-octave (graphic) equaliser the adjustment procedure is considerably simpler; this is in fact one of the main advantages of this type of equaliser. A filter with switchable centre frequency (in steps of an octave or 1/3 octave) is employed as analyser filter. The adjustment procedure consists simply of setting up each frequency band in turn and varying the gain of the corresponding equaliser filter until the analyser reading coincides with the nominally flat value. As expected, the resultant response curve (see figure 12c) has a certain waviness, which is unavoidable when using a graphic equaliser. However this is of only minor importance in this type of application.

14. Irrespective of the type of equaliser which is employed, the adjustment procedure, once completed, should be checked with the aid of the following test: The system should be set up as for normal use, i.e. the equaliser is connected to point A in figure 9 and the pink noise generator removed. The analyser filter and display, however, are left connected to point A (see figure 13) for the time being. The volume control of the amplifier is then turned up to the point where acoustic feedback just starts to occur. Using the analyser filter it is a simple matter to detect the frequency at which the signal is oscillating, whereupon the gain of the corresponding equaliser filter should be reduced a fraction.

If the equaliser has been optimally set up, the system should no longer oscillate at the same frequency. If, however, it should continue to do so, then it means that the equaliser has not been correctly set up and the adjustment procedure should be repeated point for point.

16. If more than one microphone is used in the P.A. system, the above procedure is only carried out with the

main mike. The response obtained with each of the other microphones is measured separately as described in point 4. Should these all prove to be reasonably flat, the system is ready for use as it stands. If this is not the case, however, then one of the following steps may prove necessary. If one mike has an irregular response and it is of a different type to the main mike, then one should consider replacing it. If the discrepancies are only minor, then basic equalisation (one equaliser filter per mike) for each microphone may be adequate. Bear in mind that a dip in the response of the other microphones is less important than the presence of a peak. Finally, a compromise solution is also possible: i.e. one switches on all the mikes and adjusts the equaliser for the optimal response.

In conclusion it is worth pointing out that all the above measurements were carried out using a pink noise test signal. This type of signal source was in fact chosen for a very good reason. Were the response of the system measured using e.g. a sinewave generator, the response shown in figure 5a would look something like that in figure 14. The response is characterised by countless dips and peaks separated by little more than a couple of Hertz and varying in amplitude by between 20 to 30 dB. These very sharp dips and peaks are intrinsic to the response and cannot be corrected. If attempting to equalise a response measured using a sinewave generator the important thing is to align the tops of the peaks; the average and minimum amplitude levels are of minor importance, since, as already mentioned, it is the signal peaks which determine at what point the system succumbs to acoustic feedback.

Although the measurements obtained with a sinewave generator are more accurate, they are also considerably more time-consuming. In addition, when plotting the response of a system, there is the added diffi :ulty of ensuring that one is recording only the peak signal levels.

12

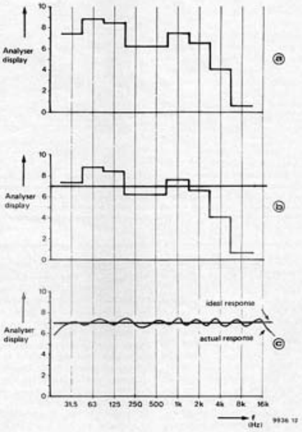


Figure 12. With octave and third-octave graphic equalisers the response can only be varied in octave or third-octave steps, hence there is little point in measuring the response of the system more accurately than this. Figure (a) shows the measured response with an octave/third-octave analyser filter; in figure (b) the nominal 'flat' value is drawn in, whilst figure (c) shows the response obtained with the equaliser optimally adjusted. The 'waviness' of the response is an inherent result of employing a graphic equaliser and cannot be rectified. However in practice it has little effect upon the final sound quality.

13

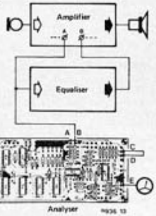


Figure 13. With the set-up shown here it is possible to check the performance of the P.A. system after equalisation.

The living room

As in the case of P.A. systems, the most suitable point in the reproduction chain to incorporate the equaliser is the monitor input of the amplifier. If such an input does not already exist, then, as already mentioned, it is a relatively simple matter to incorporate such a facility oneself.

For stereo hi-fi systems a 'stereo' equaliser in the shape of two independently variable mono equalisers is required. Quad fans need not worry, since generally speaking there is little to be gained from using an equaliser for the rear channels.

Once installed there are several methods which can be adopted to set up the equaliser. The simplest is to use the complete audio analyser described elsewhere in this issue in conjunction with a measurement microphone. However other approaches in which only part of the audio analyser is used together with a pair of high impedance headphones are also possible (it is even possible to dispense with the audio analyser entirely!). Each of these methods will be described in detail.

a. Analyser and measurement microphone

The adjustment procedure with analyser and measurement microphone is essentially the same as that adopted with P.A. systems. By 'measurement' microphone is meant a mike whose frequency response is sufficiently flat to ensure that it does not introduce a significant degree of error into the

14

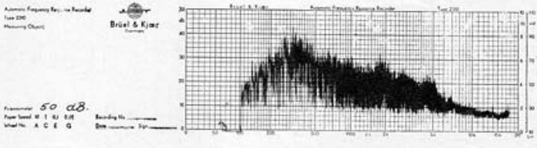


Figure 14. Until now the frequency responses shown have all been 'idealised'. However if the response is measured extremely slowly (15 to 20 minutes for one complete response curve) using a swept sinewave generator then the resultant graph looks rather different from that shown in figure 5a! One can clearly see that there are a large number of quite sharp peaks and dips which are only a few Hertz apart. These rapid variations in amplitude cannot be corrected however, and consequently there is a little point in measuring them. When using a noise generator as a test signal source, one obtains an 'averaged' response curve, which is much more useful when it comes to practical adjustments with the equaliser.

measurements. A good quality microphone of the type intended for use with reel-to-reel tape recorders should fit the bill.

The connections for the analyser and microphone are illustrated in figure 15. The microphone should be situated in the 'ideal' listening position within the room and care should be taken to exclude extraneous noise sources (wives, children etc.!) One then works through the same procedure as described for P.A. systems, but with one notable exception. As already mentioned, any dips or peaks in the response occurring between roughly 300 Hz and 5 kHz should generally be left alone. Until now, however, there has been no need for the frequency scale on the analyser filter control to be calibrated, which means that there is no way of telling where these frequencies occur! Fortunately, however, there are alternative methods of determining this frequency band with sufficient accuracy: e.g. the use of test records which have a number of specified frequencies recorded on them; alternatively one can utilise the knowledge that on a piano (or the 8' register of an electronic organ) 300 Hz coincides roughly with d^1 — the d above middle c, and 5 kHz with e^5 (i.e. four octaves above middle e).

In figure 3a the frequency response exhibited a dip at around 1600 Hz, and it was stated that if this was a result of the room acoustics, it should not be equalised; if however it was caused by the response of the loudspeaker, then it was legitimate to remove the dip using the equaliser. The simplest method of ascertaining which of these two situations is in fact the case is to measure the loudspeaker response in two different rooms. The most suitable room for this purpose (assuming it is large enough!) is the bathroom! However one must of course be extremely careful when using electrical equipment in the vicinity of water taps etc. At any rate, if the same dip in the response occurs when the loudspeaker has been set up in a different room, then one can safely assume that it is the fault of the loudspeaker itself.

Since a stereo equaliser actually consists of two separate mono equalisers, in theory the adjustment procedure should be carried out twice, once for each channel, and in each case with the other channel completely disconnected. In practice, however, it is sufficient to feed the noise signal to the desired channel and simply to turn the balance control on the amplifier to the appropriate end stop. Any crosstalk between channels should be too small to affect the resultant measurement.

15

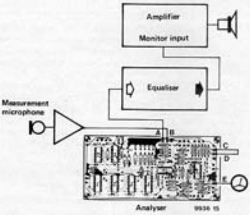


Figure 15. If a reliable measurement microphone is available the arrangement shown here can be used to measure the response of a hi-fi system and living room.

Test records

Certain hi-fi stores stock various test records which often include pink noise test signals. In principle, these can be used in place of the pink noise generator of the audio analyser. The adjustment procedure then becomes slightly more inconvenient, since one must constantly search for the right spot on the record for each measurement; however this in no way interferes with the accuracy of the adjustment procedure.

Sinewave test signal

It is also theoretically possible to use a pure sinewave (whether from a sinewave generator or a test record) as a test signal, however this approach is not recommended. As has already been

16

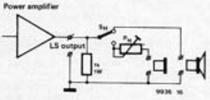


Figure 16. When using headphones to measure the room's response a volume control is necessary to be able to adjust the signal from the 'phones' until it sounds the same as that from the loudspeaker. In addition one must be able to switch between the two, so as to make a direct comparison.

explained, the actual frequency response of the system consists of a large number of very rapid variations in signal level. Were a sinewave generator employed as a test signal source these peaks and dips would be reflected in the measurement. One would then have to determine the 'average' frequency response of the system before one could set about equalising it. A small drift in the oscillator frequency, a fractionally incorrect setting of the controls, could lead to differences in signal level of from 5 to 10 dB. Such is the risk or error using a sinewave test signal that it is best to avoid this approach altogether.

Headphones

There may be those who do not wish to purchase a measurement microphone (and suitable pre-amp) solely for the purpose of setting up an equaliser. If that is the case an alternative solution is to use a pair of high-quality headphones. The adjustment procedure is simplest if one has a pair of 'open' headphones, i.e. which do not acoustically isolate the ears from external sounds. Figure 16 shows how the headphones are connected to the amplifier. This set-up allows one to switch from loudspeaker to headphones and to vary the volume of the headphone signal until it sounds the same as that from the loudspeaker (It is important that the headphones do not muffle or distort the loudspeaker signal in any way).

Since the switch and volume potentiometer must be operated from the desired listening position, a sufficient length of suitable cable is required.

The connections between the amplifier equaliser and analyser are shown in figure 17.

Once again it is possible to use a test record as a pink noise source in place of the noise generator on the analyser, although it is less convenient. The display or meter section of the analyser is not used with this set-up (no measurement mike), instead one trusts to one's ears to distinguish between signal levels. This does require a certain amount of concentrated listening, however in practice this has proven to work quite well. The adjustment procedure is as follows:

1. The analyser filter control is set to roughly its mid-position, and with the S_H switch (see figure 17) in the 'loudspeaker' position, the noise signal is adjusted to a reasonable room level. If the volume of the noise signal is too high it is not only extremely disagreeable, but there is also a risk of damage to the speaker!

2. Potentiometer P_H is set for maximum resistance, switch S_H is moved to the 'headphones' position, and P_H is then adjusted until the signal from the headphones sounds to be at the same level as that from the speaker was.

3. The frequency of the analyser filter is gradually moved up and down the entire spectrum and the differences between the signal levels of the loudspeaker and of the headphones are noted – loudspeaker slightly louder, much louder, the same, etc. At the same time one should observe at what points the highest peaks (i.e. greatest signal levels) and lowest dips (smallest signal levels) occur. A useful method of recording one's observations is illustrated in figure 18a; figure 18b shows the corresponding frequency response. With this information one can now proceed to set up the equaliser in the manner described above, using the signal level established in point 1 as the nominal 'flat' value. As already mentioned, the band of mid-range frequencies should normally be left unaltered.

Summarised briefly, the remainder of the adjustment procedure is as follows:

4. All the equaliser (bandpass) filters are set for maximum Q. With the aid of the analyser filter the first peak (in figure 18 this lies between test points 1 and 2) is detected, the first equaliser filter is set for maximum cut and its centre frequency adjusted until it coincides with the top of the peak. The amount of attenuation introduced by the filter is then adjusted until the signal level of the loudspeaker and headphones is the same. This procedure is repeated with the remaining equaliser filters for any other irregularities which require correction (in figure 18 the other prominent peaks and dips fall within the

17

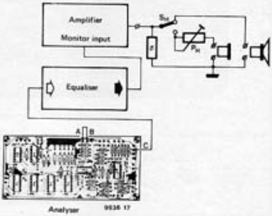


Figure 17. Connections between amplifier, equaliser and analyser when using headphones to measure the room's response.

18

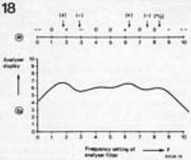


Figure 18. An example of how one can audibly chart the response of the room when using headphones. ++ signifies: loudspeaker much louder than headphones, 0 means both are equally loud, etc. The tops of peaks and bottoms of dips are marked with an arrow. The actual curve which corresponds to this notation might look something like that shown in figure (b).

critical mid-range of frequencies to be left alone).

5. Using the analyser filter, find the frequency at the lower end of the spectrum at which the loudspeaker begins to sound perceptibly quieter than the headphones (just below point 1 in figure 18); set the bass control filter of the equaliser to its lowest frequency and adjust it for maximum cut. Then gradually increase the turnover frequency until the loudspeaker sounds even quieter still. Repeat the above procedure for the equaliser treble control (in figure 18 the reference frequency will probably lie just above test point 9).

6. Set the analyser filter frequency to minimum and increase the gain of the bass control until the 'flat' level is obtained; adjust the treble control in the same way.

7. On the sides of the original first peak in the response there should now be two new peaks. Adjust the analyser filter until it coincides with one of these new peaks and reduce the Q of the first equaliser filter until it has disappeared. If necessary repeat this procedure with the remaining equaliser filters.

8. Finally, sweep the analyser filter up and down the entire audio spectrum and check to ensure that all the adjustments that have been made are correct. It will generally prove necessary in practice to make a few additional corrections or alterations. Once done, the system is now ready for use and can be subject to the crucial test of introducing a suitable music signal and listening to hear (hopefully) the improvement in the resultant sound.

Bibliography:

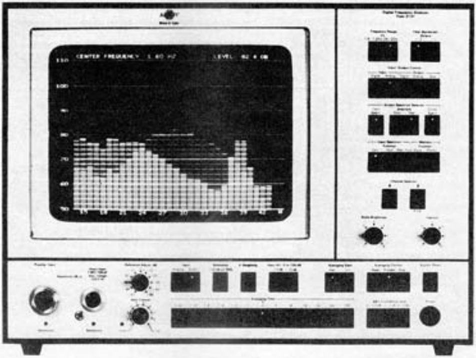
J. Moir: Interactions of loudspeakers and rooms. Wireless world, June 1977.

Bruel and Kjaer: Relevant Hi-Fi tests at home. Paper at the 47th Audio Engineering Society Convention. Also available as a Bruel and Kjaer application note.

Bruel and Kjaer: Hi-Fi tests with 1/3 octave weighted, random noise. Bruel and Kjaer application note no. 13-101.

Philips: Sound equalisation using Philips K and Q-filters. ELA application note 17.8100.35.331011.

2



An example of an extremely sophisticated (and expensive) spectrum analyser used for professional applications. The model shown here is the 2131 Digital Frequency Analyser from Brüel and Kjaer, which splits the audio spectrum up into octave or third-octave frequency bands and displays the corresponding signal levels on a CRT.