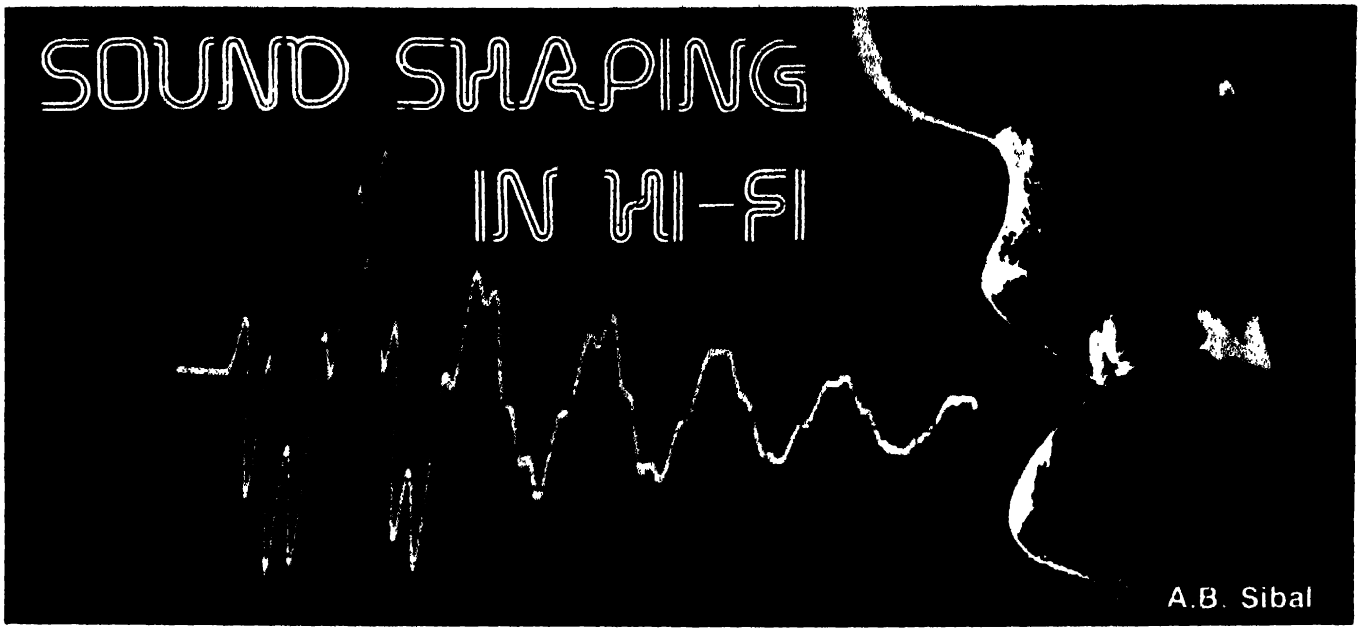

The sound we actually hear from a hi-fi system is not entirely the result of the system's performance. We hear the result of many interacting factors which are the subject matter of applied and practical acoustics.

In hi-fi reproduction, the listening room is an important factor in estimating the system's performance. But unfortunately, little consideration is given to the listening room. Just as we control the resonance peak of the loudspeaker to get smooth reproduction, so must we tame the resonance of the room so that the system performs properly.

The reflection and absorption properties of the room affect the audio reproduction by producing many irregularities of response. A very live or resonant room is detrimental to good musical reproduction on the basis of cleanness of reproduction of notes. Live rooms affect directivity in that sound is more diffused throughout the room. On the other hand, in a dead room sound intensity decreases because the ears do not benefit from reflected sound. Such a room is wasteful of power delivered to it.

The human ear is not a linear device. The sensitivity of the ear for different frequencies changes as the sound grows softer or louder. A system optimally adjusted for the ear under one condition may be entirely out of balance for another condition. Also, the ear concentrates on a small portion of sound, viz. 400-5000 Hz, to receive good communicative intelligence. The lower and higher ends are not at all necessary for the ear to make sense out of what it hears. In high fidelity, however, we are concerned with more than this. We want the lowest throb and the highest tweet. The lows and highs belong to the musical intelligibility of sound.

But there is not much a listener can do with his living room. Designing or changing the living-room decor for this purpose may not always be possible. Nothing also can be done to modify the physiological hearing process.

So, to overcome the above shortcomings, it is much better to shape the response of the audio system so that the overall response takes the above factors into account.

Need for sound shaping

The number of controls on today's hi-fi equipment is indicative of the realisation of the need for sound shaping. Basically these controls are all means by which the listener may alter the colours of his musical picture. Purists have been debating that the listener should not have the privilege of editing or auditing of music presented to him. The listener should not corrupt the artiste's interpretation of the work, they feel. To the purist reproduction means exact duplication of the original. The purist is a lover of music.

But there is the listener who prefers to recreate the sound that is satisfying to his ears. To achieve this, he adjusts the many controls on his preamplifier, amplifier and speaker. He is a lover of sound. There is no doubt that both the 'music lover' and the 'sound lover' appreciate good reproduction. Both need sound shaping circuits and devices though for different reasons.

Tone controls

The most common and popular technique of shaping the audio response of a system is the Baxendall tone control. This circuit, in one form or the other, is used the world over. This is a control more for the lover of sound than for the lover of music. It does not allow any scope for selective adjustments of audio response except for raising or lowering signal level at bass or treble frequencies. Both bass and treble controls are generally designed to provide $\pm$ 10 dB of boost and cut (at 100 Hz for bass control and 10 kHz for treble control) relative to 1 kHz.

Let us see the working of such a circuit in the light of an operational amplifier working in the inverting mode (Fig. 1). If in such an amplifier a feedback resistor is connected from the output back to the inverting input and a series resistor is connected between the signal source and the

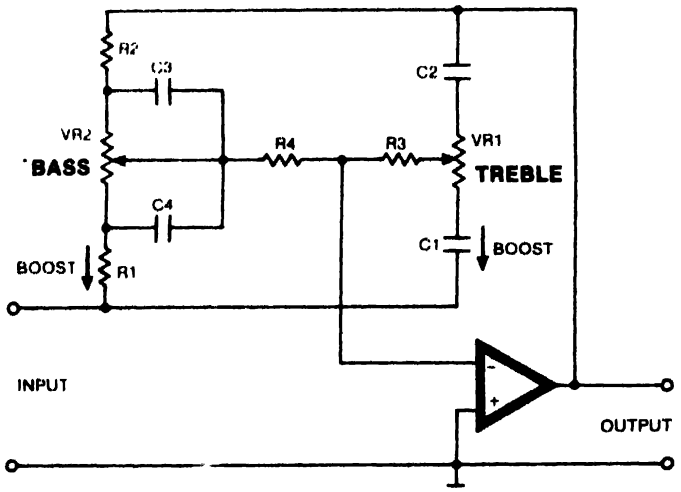

Fig. 1: The schematic diagram of the Baxendall tone control.

inverting input, the voltage gain of the amplifier is equal to the feedback resistance divided by the input series resistance. The voltage gain is therefore controlled by feedback impedance given by the tone control components and by the input series impedance.

Treble control. The treble control employs VR1, C1 and C2. The feedback resistance is through C2 and the top end section of VR1; and the input impedance is given by C1 and the bottom end section of VR1. At low and mid frequencies, C1 and C2 which are equal in value have a reactance which is high as compared to VR1 and so adjustment of VR1 has very little effect on the relative values of feedback resistance and input series resistance which virtually remain the same. At higher frequencies reactance of C1 and C2 is comparable to VR1 and at the highest frequency this reactance is very low compared to VR1 so that adjustment of VR1 has a marked effect on sound output.

With the slider of VR1 towards C2 end, the feedback resistance becomes much lesser than input series impedance, giving significantly reduced gain or treble cut. When the slider of VR1 is moved to C1 end, the feedback resistance is much larger than the series input impedance producing increased gain or treble boost. With VR1 at the centre position, the two impedances are equal, a flat response with unity high frequency gain is obtained.

Bass control. This control also works in a similar fashion. At high and mid frequencies C3 and C4 with low reactance virtually short VR2. Since R1 and R2 have same value, the feedback impedance at these frequencies is equal to series input impedance and the adjustment of VR2 has no significant effect. At bass frequencies, compared to VR2, reactances of C3 and C4 are higher and affect the output.

In this low frequency range moving VR2 towards R2 makes the feedback resistance lower than series input resistance, resulting in reduced voltage gain or bass cut. When VR2 slider is at R1 end the voltage gain is increased, resulting boost. With VR2 at its centre position, unity gain flat bass response is obtained.

R1 and R2 limit the maximum amount of boost and cut respectively. R3 performs a similar limiting function in the treble circuit. R4 helps to minimise the interaction between the two controls.

Tone controls add to the noise and distortion problems of the system. Tone controls are acceptable only if they are essentially quiet and imperceptible in operation and can be individually switched out of the circuit when not required. They should be part of a stereo preamplifier that has the lowest noise and distortion figures. They are not effective in compensating room acoustic problems, loudspeaker errors and unsuitably balanced recordings. It is noteworthy that high quality preamplifiers are today without tone controls the world over.

Equalisers

Equalisers are popular audio accessories these days. They are sort of supertone controls that allow the user to change small portions of audio spectrum so that overall response is neutral. They provide the ideal way of affecting any frequency modification when a specific need arises. Equalisation can be defined as a process designed to make the audio response as flat (equal) as possible at all frequencies.

Equalisers tame two types of resonance – those with gain and those with loss. And contrary to popular belief, an equaliser is called upon on more occasions to cut signals than to boost them.

Equalisers can be broken down into two sub-categories, graphic and parametric. Graphic equalisers divide the audio spectrum into a given number of bands with individual boost cut controls for each band. The centre frequency of each band is fixed. The family of response curves generated by this type of equaliser resembles a series of peaks and valleys.

In the parametric equaliser, the amount of boost or cut introduced, the central frequency of the band of equalisa-

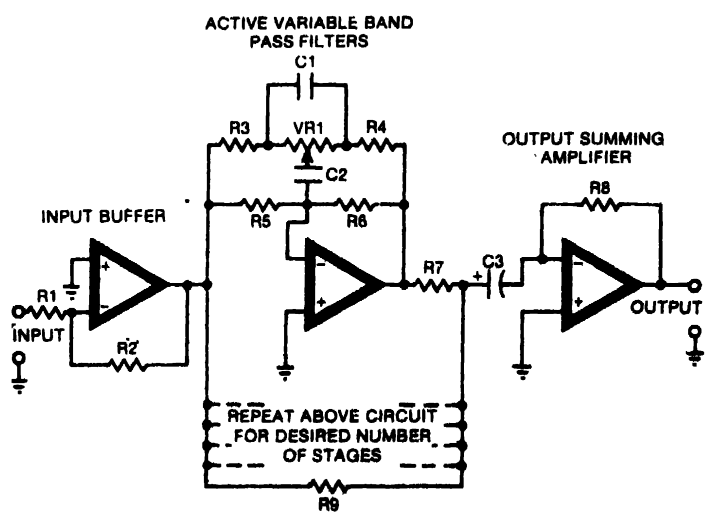

Fig. 2: The schematic diagram of the graphic equaliser.

tion and bandwidth within which equalisation is applied, are all independently variable. The parametric equaliser, providing these three variables to the listener is more versatile. It gives him the ultimate control over the sound reproduced by the hi-fi system.

Graphic equaliser. This is the most popular type of equaliser because of the fact that it is easiest to adjust. A graphic equaliser section is shown in Fig. 2. It consists of an input buffer, a desired number of active variable band pass filters and an output summing amplifier. The input buffer is a standard unity gain stage for impedance matching between the preamplifier and the equaliser section. The active filter section is actually a number of individual active filters with the same feedback design. In all the stages the values of C1 and C2 are responsible for setting the centre frequency. For VR1 normally linear pots are used. The setting of VR1 gives the degree of boost or cut desired in that stage.

The final stage is the summing amplifier buffer stage to sum the individual filters. R9 is necessary to maintain a unity gain configuration at the output with all pots set to the flat position. It subtracts the original signal from the sum by a factor dependant upon the number of filter stages. R9 is selected according to the number of bands required.

Each channel of the equaliser is controlled by separate slide potentiometers and the array on the front panel gives an approximate indication of the response inserted to modify the audio system frequency response. Human hearing is relatively insensitive to frequency response errors of less than 1/3 octave. So, if equalisation is to be taken seriously, the professional graphic equaliser will have 24 to 31 bands at 1/3 octave spacing. However, for reasons of economy, the 10-band equaliser with 1 octave spacing is the most popular for home use.

Parametric equaliser. A simplified block diagram to describe the principle of parametric equaliser is shown in Fig. 3. It shows a unity gain inverting amplifier inter-connected with state variable band-pass filter. A potentiometer is connected between the input and the output of inverting amplifier to give adjustable boost and cut. The centre frequency and filter bandwidth can be selected or adjusted by means of two potentiometers incorporated in the circuit of the band pass filter.

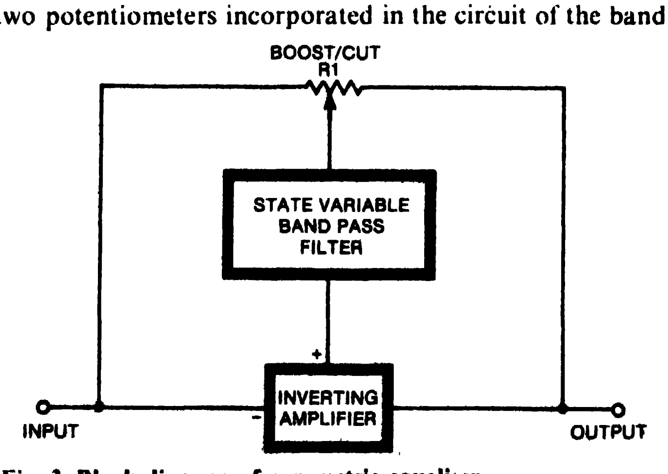

Fig. 3: Block diagram of parametric equaliser.

When the wiper of R1 is at the left extreme of its travel (input side) the output of the band-pass filter adds to the input signal, resulting in a boost within the filter pass band. When the wiper of R1 is at the right extreme of its travel (output side), the band-pass signal is subtracted from the input signal, attenuating input signal within the pass band of the filter.

At the mid point position of R1 wiper, no signal is routed to the band-pass filter as the output of the inverting amplifier is equal and out of phase with the input signal at the wiper. This means the band-pass filter will give no output and hence has no affect on inverting amplifier which therefore exhibits a flat frequency response.

A simplified schematic of one channel of such an equaliser is shown in Fig. 4. IC1 and associated resistors make the inverting amplifier. The state variable band-pass filter is composed of a differential amplifier (IC2 and its associated passive components) and two active integrators in cascade (IC3, IC4 and associated components). The bandwidth of filter can be varied by adjusting potentiometer R1 and the centre frequency selected by adjusting dual potentiometer R2. R3 is used for boost and cut.

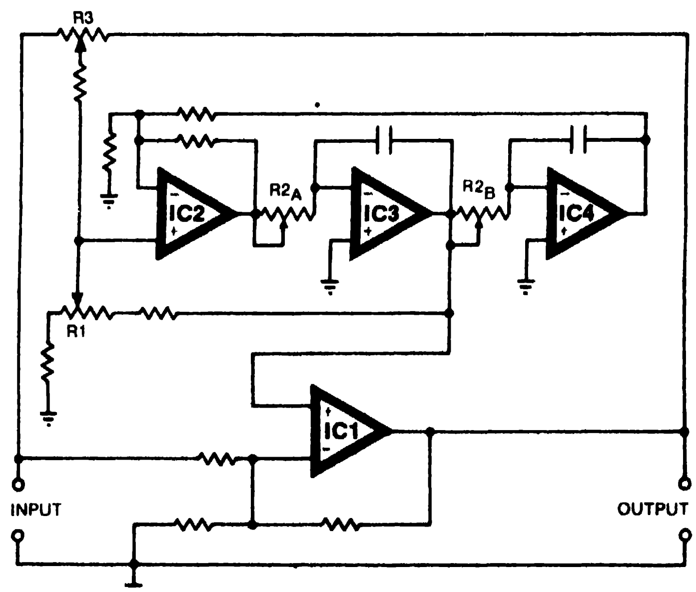

Fig. 4: Simplified schematic diagram of one channel of parametric equaliser.

Installing the equaliser

Equalisers are available as separate audio accessories with built-in power supplies. Most hi-fi systems can accommodate the equalisers through tape monitor loop. Connect the 'tape out' or 'tape record' output of your amplifier to the equaliser's input. Connect the equaliser's output to the 'tape in' or 'tape monitor' jacks of your amplifier. Push the tape monitor switch to 'on' to enable the equaliser. The signal of the preamplifier will now pass through the equaliser before entering the power amplifier.

Adjusting the equaliser

Equalisers can be adjusted in different ways. These vary in convenience, cost and accuracy. Hi-fi addicts familiar with live music can adjust the bands by the ear to match their idea of good sound (lover of sound). Another way is to use a real time analyser, a costly audio accessory, to adjust the equaliser (lover of music).

However, the room system combinations are best equalised by first employing acoustic methods followed by electronic equalisation. First try to reposition the loudspeakers, modify the absorption coefficient of the room as far as possible and adjust the speaker crossover level controls (if any)

Adjusting by the ear. Adding sharp boosts at very low and high ends of audio spectrum allows the listener to compensate for speaker roll-off. A broad dip inserted at mid-band frequency will simulate loudness control to enhance low level listening Most often a lack of deep bass or extreme high is due to limitation of the dynamic drivers.

Never try to force flat response out of the speakers by blindly applying large amounts of deep bass and extreme treble boost. Such efforts may result in an overloaded amplifier, excessive distortion and blown voice coils.

The sound due to resonance which boosts the frequency response is painfully obvious to the listener. The sound from such resonance continues in the room long after the signal has stopped. The equaliser eliminates the problem even if it is not right on the dot frequency of the resonance by reducing the energy that causes it.

Adjusting with audio analyser. The audio analyser allows the frequency response characteristic of the audio signal coming from the loudspeaker to be quickly measured and displayed. The basic unit along with its built-in microphone preamplifier and noise generator is capable of analysing signals in the octave bands.

With equaliser analyser combination one can anticipate an impressive improvement in the sound system. The characteristic of the total sound circuit (including the room) can now be easily tailored to produce an optimally flat overall response. Sound colouration due to unwanted speaker and room resonances will be substantially reduced, producing a subjectively cleaner sound.

By varying the equaliser response in the respective octave bands, the listener will get a number of plateaus on the response curve, all of which can be moved up or down independently of each other, till the display on the analyser becomes as flat as possible.

For achieving optimum equalisation the listener, for given loudspeaker position and room decor, may have to spend hours in doing the equaliser adjustments. It is better to make a cardboard template for these slider pot control settings for equalisers with slide potentiometers. Even if the controls are now altered or tampered by someone else, with the help of the template, the listener can always return quickly to the correct control settings.

In case of equalisers with rotary potentiometers a note must be made of their marking position on the dial or front panel. Such measures will save the listener many agonising hours of adjustment, especially where equalisation is done by the ear.

Conclusion

No doubt today's reproduction is more life-like and realistic but one must recognise the artiste's concern for retaining the originality of sound

But nothing in nature works by itself without being affected by something else. The listening room, loudspeaker errors and unbalanced recordings modify the sound in intensity, tonal characteristics and directivity. To get sound without coloration everything has to be 'right' acoustically and electronically. And even then the listener can only 'approach' the real performance.

The most effective way of this 'approach' is through equalisation. Tone controls at the most are only a useful facility to have but cannot effectively counter resonance problems. Equalisers built with high quality very low noise op-amp ICs (like TL071, TL351, NE5534) which are today available seem the ideal solution to this problem.

Equalisers have to be used intelligently and that too only when a specific need is apparent. Remember, equalisers are called upon to cut signals more frequently than to boost them. □