An introduction to hifi, Pt.16

Audio amplifiers — 1

Signal switching, phono preamps, tone control

Having, early in the series, discussed the broad concept of domestic hifi sound reproduction and, more recently, examined typical signal sources, the logical next step is to consider the actual amplifier, which receives the source signal from the phono player, tuner, &c, and boosts it to a level sufficient to drive the loudspeakers.

by NEVILLE WILLIAMS

In mass-produced receivers, record players and tape players, the audio amplifier is commonly built right into the unit, being no more elaborate or costly than is necessary to satisfy the immediate requirement.

In the context of high fidelity reproduction, however, the amplifier is more likely to be a physically separate unit and designed on a more generous basis, with a view to obtaining the highest possible quality of sound reproduction, consistent with acceptable cost and complexity.

In the discussion to follow, frequent reference will be made to technical terms and concepts explained in two earlier chapters in this series. Readers may care to check back over them by way of revision:

Hifi Stereo: what it means in simple terms (March 1986, p.10); and

Hifi facts and figures (April 1986).

Fig.1 depicts in block schematic form the major sections — or functions — of a modern high fidelity amplifier. The boxes are drawn with double borders as a reminder that they represent stereo units providing for two identical signal channels. The input and output signals are assumed to be stereo but, to avoid visual clutter, no attempt has been made to show the connecting lines and switch functions in duplicate.

In most domestic situations, hifi amplifiers are used in conjunction with a range of signal sources and normal practice is therefore to provide such amplifiers with multiple input sockets at the rear and an associated signal selector switch on the front panel.

In the early days, one could get by with three selectable inputs: Phono, Radio and a spare channel marked "Aux" (Auxiliary) but this soon expanded to four: Phono, Radio, Tape and Aux.

Subsequently, when compact cassette decks won acceptance, not just as signal sources but for recording and copying as well, it became common practice to provide tape input and output facilities for two such decks, in some cases with supplementary switching that enabled tape/-tape copying to proceed simultaneously with normal use of the radio and phono channels.

As if that was not enough, DAT (digital audio tape) players have now appeared on the hifi horizon, along with the possible further need to route audio signals from TV/video equipment through the sound system. Tomorrow's signal select function may well need to cope with a multiplicity of inputs, such as indicated in Fig.1, along with signal feeds back to the recording devices.

The provision of so many stereo signal select/feed options poses a very real practical problem for the designer. The one-time use of a rotary switch and multiple stereo pairs of shielded wire leads is much too clumsy, necessitating either pushbutton electronic switching using CMOS devices, or else unshielded tracks on a PC board combined with extraordinary care with layout (see the Playmaster 60/60 amplifier, May 1986).

Input signal levels

With the sole exception of the phono player, all of the signal sources nominated in Fig.1 contain internal electronic circuitry, which processes the signal as appropriate, compensating it to a nominally flat frequency response and boosting it to a convenient level for input to the amplifier.

It has become accepted practice over the years to provide a nominal output signal level of between 0.5V and 1.0V RMS, with an effective source impedance ranging from a few hundred to a few thousand ohms. Output signals of this general order are often (but rather loosely) described as being at "line" level, signifying that they are ready to feed directly to an amplifier.

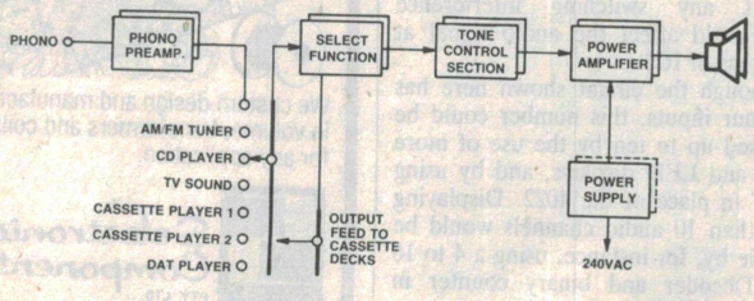

Fig.1: The major sections in a hifi amplifier. Switching on the front panel selects the desired signal and passes it through a tone control system to the main amplifier which, in turn drives the loudspeakers.

In turn, domestic amplifiers are commonly designed with a "line" input sensitivity of around 0.25V (250mV RMS). In other words, they are capable of delivering full rated power with an input of that order. Most amplifiers therefore have gain to spare which, if nothing else, may be reassuring to the user.

When fed with a larger input signal (typically 0.5V or more), the amplifier volume (or gain) control must therefore be set well back to reduce the signal input — and the amplifier output — to the desired level. With typical amplifiers and signal sources, a comfortable listening level is commonly reached with the volume control at about the "10 o'clock" setting. With the control at "12 o'clock" (half rotation) the volume level is usually quite high, the system running into overload somewhere beyond that.

There is no cause for concern if, with some signal sources, the volume control needs to be advanced further than suggested above, provided the system can still be driven to the desired level. It is simply a case of a somewhat smaller input signal requiring extra amplification.

If, on the other hand, normal listening level is achieved at "8 o'clock" and full volume at "10 o'clock", there is cause for concern. Not only may the volume control be unpleasantly critical in use, but it could be that the signal being fed to the amplifier is of greater amplitude than it is meant to accommodate — with the possiblity of overload and distortion on peaks, irrespective of volume control setting.

This situation is most likely to arise with compact disc players, some of which have a nominal output signal level of around 2.0V RMS and no provision to reduce it. The logical course is to insert a resistive attenuator pad in the respective left and right channel signal leads to reduce the level by about 3:1. Details of a switchable attenuator for this purpose were given in the January 1986 issue (see Fig.2).

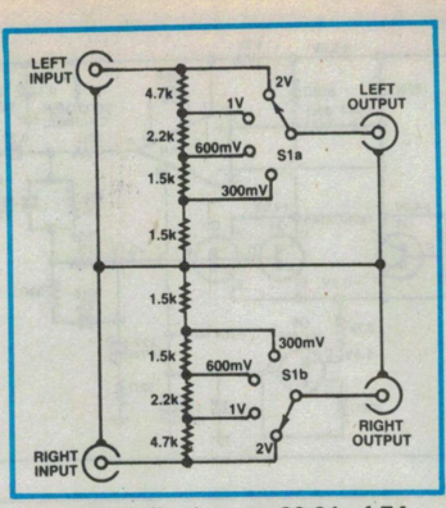

Fig.2: Described on pp.90-91 of EA for Jan 1986, this switchable stereo attenuator can reduce the nominal 2.0V output from a CD player to 1.0V, 600mV or 300mV, thereby avoiding the risk of input overload distortion.

Phono preamplifiers

Phono decks — or "black disc" players present a problem of the reverse kind in that, lacking any in-built signal processing circuitry, they deliver a signal which is neither flat, in terms of frequency response, nor adequate in terms of amplitude. (For a detailed discussion of phono decks, see chapters 4 and 5 in this series, May and June 1986).

It is true that piezoelectric (crystal and ceramic) phono cartridges can deliver a signal that is reasonably flat and reasonably close to "line" level but, with rare exceptions, their overall electrical performance and mechanical characteristics fall well short of high fidelity equipment standards.

Virtually all hifi phono cartridges are therefore of the "magnetic" variety, with a nominal signal output of around 5mV. Moreover, the frequency response is far from flat, being down by about 18dB at 30Hz and up by almost the same amount at 15kHz. To correct this situation, the signal needs to be processed through a preamplifier compensated to the so-called RIAA characteristic which must:

(1). Amplify it to nominal "line" level — around 0.5V RMS; and

(2). Boost the bass end by up to 20dB and cut the treble in a similar manner, to achieve an overall characteristic which is hopefully flat, within about 1dB, from 30Hz to 15kHz. (See Fig.3).

Prior to the arrival of solid-state devices, designers had no choice but to use valve type preamplifiers but, with such a low level of signal, especially at the bass end, it was difficult to provide the required degree of amplification without the signal being compromised by noise inherent in the valve circuitry, by 50Hz hum injection from the cathode/heater wiring, and by microphonic effects resulting from vibration of the valve electrodes.

It was especially difficult to achieve a sufficiently low noise level with a phono preamplifier built into the same case as the rest of the amplifier and power supply. It became quite common practice, therefore, to accommodate the phono preamplifier in a small metal box connected by cable to, but isolated from, both the phono deck and the remainder of the amplifier.

Solid-state preamplifiers

Being small in size and without a heater circuit, transistors seemed to offer a way around these difficulties but they proved to have problems of their own, which took some time for manufacturers and designers to sort out: junction noise, low input impedance, limited frequency response, and overload on peak signal levels.

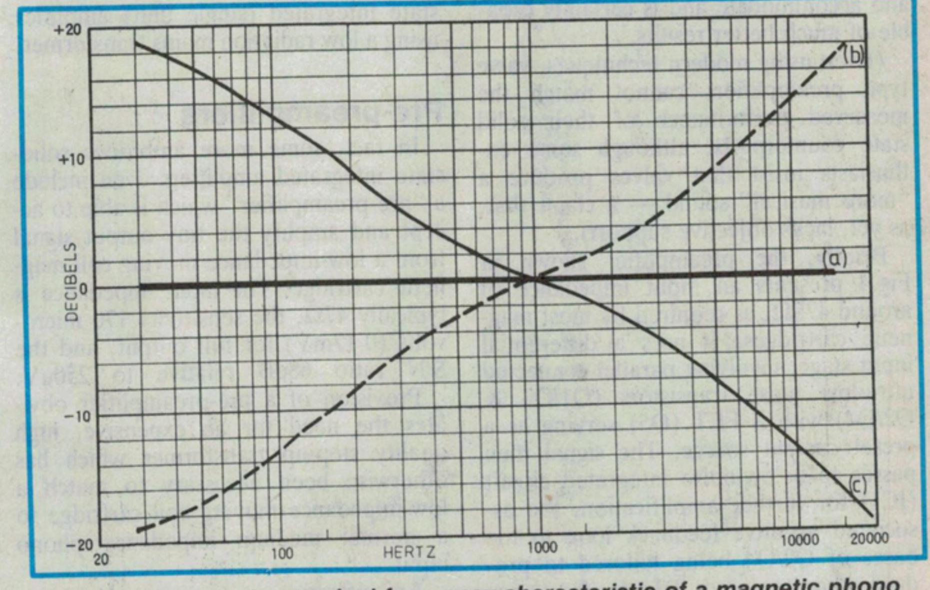

Fig.3: Curve (b) is the nominal frequency characteristic of a magnetic phono cartridge, and (c) the required response of a compensated preamplifier. Ideally the two curves would together produce (a) but, in practice, a resultant within 1dB from 30Hz to 15kHz would be good.

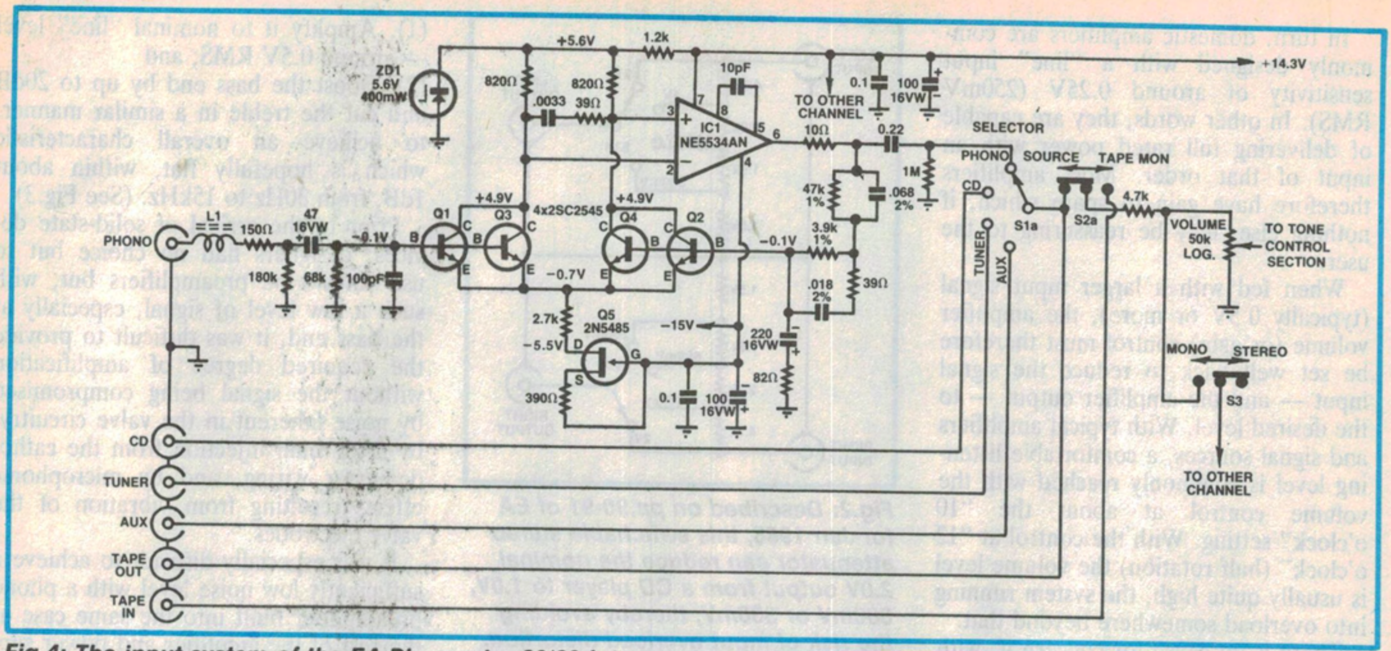

Fig.4: The input system of the EA Playmaster 60/60 integrated amplifier, lifted from the full circuit on p.37 of the June 1986 issue. It includes the circuit of one channel only of the compensated phono preamplifier.

But those problems have long since been overcome by the development of better transistors and better components generally and, more particularly by ultra-low noise transistors, ICs and circuit techniques able to reduce noise greatly in the audio passband.

Fig.4 shows the circuit of one channel of a typical modern phono preamplifier, lifted from the full circuit diagram of the Playmaster 60/60 amplifier (EA June 1986, p.37). It is functionally quite complex, compared with a single EF86 stage from the valve era, but with modern components and techniques, it would probably be easier to assemble and accommodate and is certainly capable of much better results.

(Even using modern techniques, valve type preamplifiers cannot match the measured performance of their solid state counterparts, although some enthusiasts insist that valves produce a "more musical" sound — a claim that, as yet, lacks objective support).

Briefly, the preamplifier shown in Fig.4 presents an input impedance of around 47kΩ, as required by most magnetic cartridges. It uses a differential input stage, involving parallel-connected ultra-low noise transistors (Q1/Q3 & Q2/Q4) with a FET (Q5) serving as a preset current source. The signal then passes to a low-noise integrated circuit (IC1) for further amplification, the associated negative feedback loop to the bases of Q2/Q4 being tailored to produce the required RIAA frequency compensation as indicated in Fig.3c.

Following IC1, an RC coupling network (0.22uF/1MΩ) attenuates possible turntable/disc rumble components below 20Hz, before making the signal available to one pole of selector switch S1a. Fewer input options are provided than shown in Fig.1, with connections for one tape deck only. The circuit does, however, include a mono/stereo switch (S3), a facility not previously mentioned.

Measured input sensitivity of the phono channel, as shown, is 4.3mV for full output and the signal/noise ratio at 1kHz, relative to 10mV input, is 89dB unweighted, using a routine magnetic cartridge. The latter figure is typical for the phono channel in a modern solid-state integrated (single unit) amplifier, using a low radiation mains transformer.

Pre-preamplifiers

In fact, some more ambitious solid-state integrated amplifiers even include a "pre-preamplifier" which is able to accept and amplify the tiny output signal from a low-impedance moving coil magnetic cartridge. The input impedance is typically 47Ω, the sensitivity 170 microvolts (0.17mV) for full output, and the S/N ratio 68dB relative to 250uV.

Provision of a pre-preamplifier obviates the need for an expensive, high quality step-up transformer which has otherwise been necessary to match a low-impedance moving coil cartridge to a normal medium impedance phono input.

An effective, compensated phono preamplifier has been an essential feature in domestic hifi amplifiers for the past 40-odd years but, ironically, that situation may be about to change, with the rapid swing to compact disc and the further possibility of digital cassettes. Phono decks certainly stand exposed as the one signal source unable to deliver a "line" signal.

With the pressure on amplifier manufacturers to cut costs where they can, the possibility must be seen of the phono preamplifer becoming an external option or, more logically, an integral part of the phono deck itself. Having in mind the motor and arm control electronics that is now relatively commonplace in modern turntables, the inclusion of signal processing circuitry would not present any great hassle.

Tone control system

From the early 1930s at least, most receivers and amplifiers have featured some kind of tone control, most commonly involving a switch or a potentiometer which served to introduce a bypass capacitor across the audio signal circuit. The value of the capacitor was chosen such that it would progressively attenuate frequencies above about 2kHz, producing what was euphemistically called a "mellow" tone.

Critics preferred to describe it as "muffled"!

Listeners interested in quality reproduction largely rejected simple "top-cut" tone controls as inappropriate for a quality system, but their opinions diverged.

Some spurned tone controls altogether on the grounds that hifi components should be engineered, as far as possible, for a flat frequency response

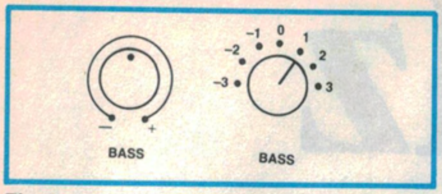

Fig.5: Potentiometers (left) offer smoother tone control than switches (right) but the attendant circuitry must be carefully designed if response in the median position is to be truly flat.

and left that way! Over the years, many amplifiers have conformed to that philosophy, with little more on the front panel other than an on-off switch and indicator, a channel selector and a volume control.

Other enthusiasts have been equally insistent that frequency compensation facilities were desirable, because they would permit the user to compensate for tonal imbalance that might occur anywhere from the signal source to the listening room. Judicious correction could restore full enjoyment of the performance.

That view has obviously prevailed in that, historically, most high quality commercial amplifiers (and systems) have included tone control facilities.

In their most basic form, (hifi) amplifier tone controls involve two multi-position switches, or two potentiometers, controlled by knobs on the front panel, marked respectively "BASS" and "TREBLE", and calibrated to indicate a level response in the median position, progressive boost when turned clockwise, and progressive cut when turned anticlockwise (Fig.5).

Switches were popular initially but potentiometers gained acceptance by reason of their smoother action. The use of calibrated knobs and, in many cases, mechanical indexing, allows particular settings to be identified.

Fig.6 shows the measured adjustment limits of the bass and treble potentiometers in a typical, modest hifi amplifier. With the bass control in the median position, the bass response is flat down to 20Hz, the limit of the graph, to within less than 0.5dB — an entirely acceptable figure.

Turning the bass control clockwise causes the curve to deviate from reference at about 500Hz — referred to as the "turnover" frequency — climbing to a maximum available boost of +14dB at 30Hz. Turning the control fully anticlockwise produces a slightly steeper attenuation curve, as shown.

At the high frequency end, the maximum available boost is +12dB at 15kHz

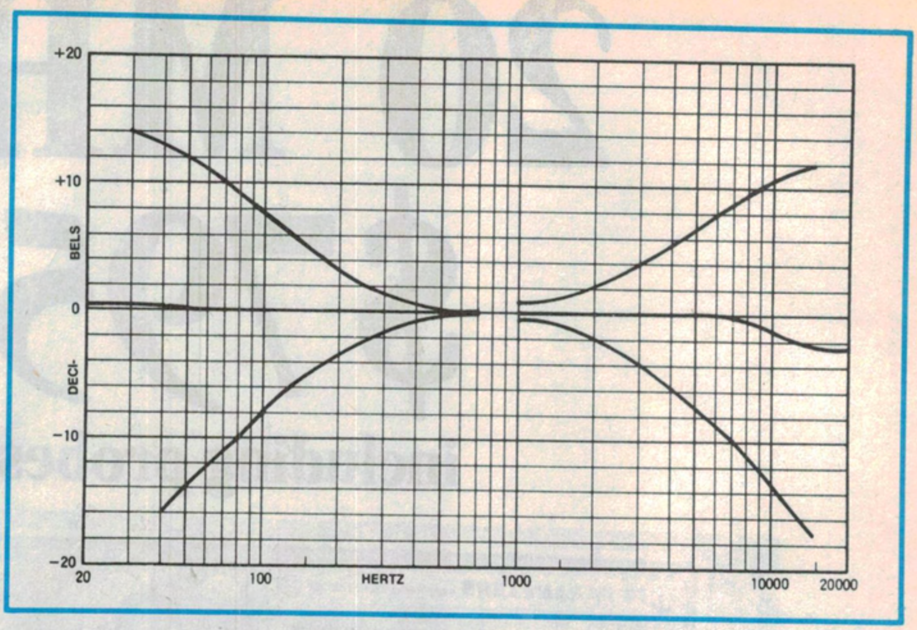

Fig.6: Typical adjustment range of the bass and treble tone controls fitted to a fairly basic hifi amplifier. Ideally, the response should be truly flat with both controls at a clearly identifiable median position.

while, again, the attenuation curve is somewhat steeper. However, with the treble control in the median position, the high frequency response is down by 2dB at 15kHz — not disastrous but certainly open to criticism.

Ideally, the response through a tone control stage should be flat right across the audio band, to within a fraction of a decibel, when the bass and treble controls are set to their visual median position.

Extra knobs and switches

Bass and treble controls with "turnover" frequencies as shown — 500Hz and just over 1000Hz respectively — have the advantage that their effect is subjectively obvious and the user is left in no doubt about their effect. However, it is not possible to boost or cut the very deep bass or the very high treble, without simultaneously boosting or cutting frequencies nearer the centre of the range.

To overcome this limitation, supplementary controls can be provided (Fig.7) which allow turnover frequencies to be selected closer to the extremes of the audio band. To the technically inclined music lover, they can be both meaningful and useful for such tasks as boosting the deep bass, cutting rumble, adding sparkle to the upper treble or selectively reducing top-end noise.

Unfortunately, the cost factor and the potential for confusion for the uninitiated is such that turnover controls are relatively rare.

By way of partial compensation many commercial amplifiers, provide low and high frequency attenuation filters, which can be cut in or out of circuit with simple toggle switches. The low frequency or "subsonic" filter is intended to operate below about 30Hz, mainly to combat rumble components from the phono turntable or disc. The high frequency filter operates above about 7kHz, to take the edge off residual noise or distortion.

Unfortunately, high/low filters often look more impressive on the panel than they sound in practice. Being of relatively simple configuration, they simply do not provide a sufficiently sharp cut-off beyond the turnover frequency to be really effective.

For extra measure, many amplifiers also feature a "Loudness" switch which has the effect of boosting the bass and perhaps the treble by a fairly arbitrary amount. The idea is to compensate for the subjective frequency loss that occurs when circumstances dictate that music be listened to at an unnaturally low level.

On the plus side, a loudness control

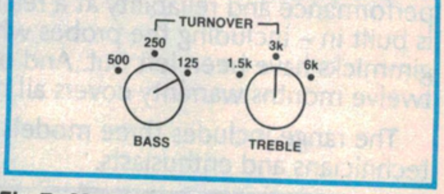

Fig.7: Means to select bass and treble control turnover frequencies — a useful feature for users able to understand their effect.

compensates for low listening level at the flick of a switch; on the minus side, it can completely unbalance the sound if it is inadvertently left on when the amplifier is operating at normal room volume.

The defeat switch

With such an array of facilities to "doctor" the frequency response, it may be somewhat reassuring to find, on many amplifier panels, a "Defeat" switch which effectively cuts the tone control system out of circuit, when set to the position variously marked "Defeat", "Direct", "DC", &c.

Defeat switches provide a ready means of comparing tonal balance with and without compensation but, more importantly, they provide a desirable option for the kind of listener, referred to earlier, who is intolerant of anything but a flat amplifier!

Fig.8 shows the circuit of a typical tone control section, again lifted from the full circuit diagram of EA 60/60 stereo amplifier. Deliberately unpretentious, it provides neither loudness compensation nor hi/lo filters.

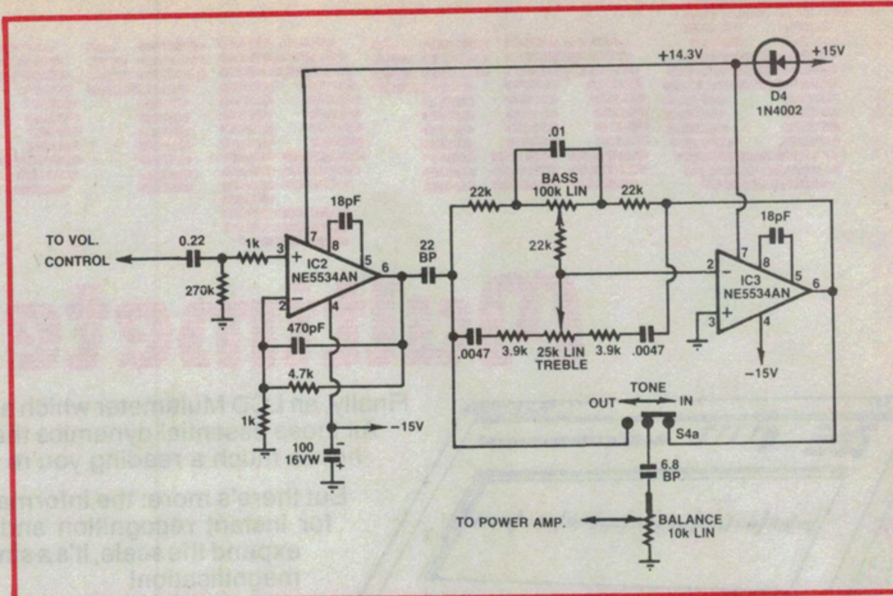

Fig.8: Also from the Playmaster 60/60 amplifier, this tone control section is typical of modern practice. Measured frequency response range is quoted as a symmetrical plus and minus 12dB at 50Hz and 10kHz.

Taken directly from the main volume control at the output of the selector system (Fig.4) the signal passes to the input of IC2 (pin 3) and appears, duly amplified, at pin 6. One feed goes direct to the Tone In/Out switch, while a second feed is routed through the bass/-treble tone control network and IC3.

The signal which is ultimately fed to the Balance control and thence to the main amplifier is either flat or subject to control, depending on the setting of the switch.

(To be continued)