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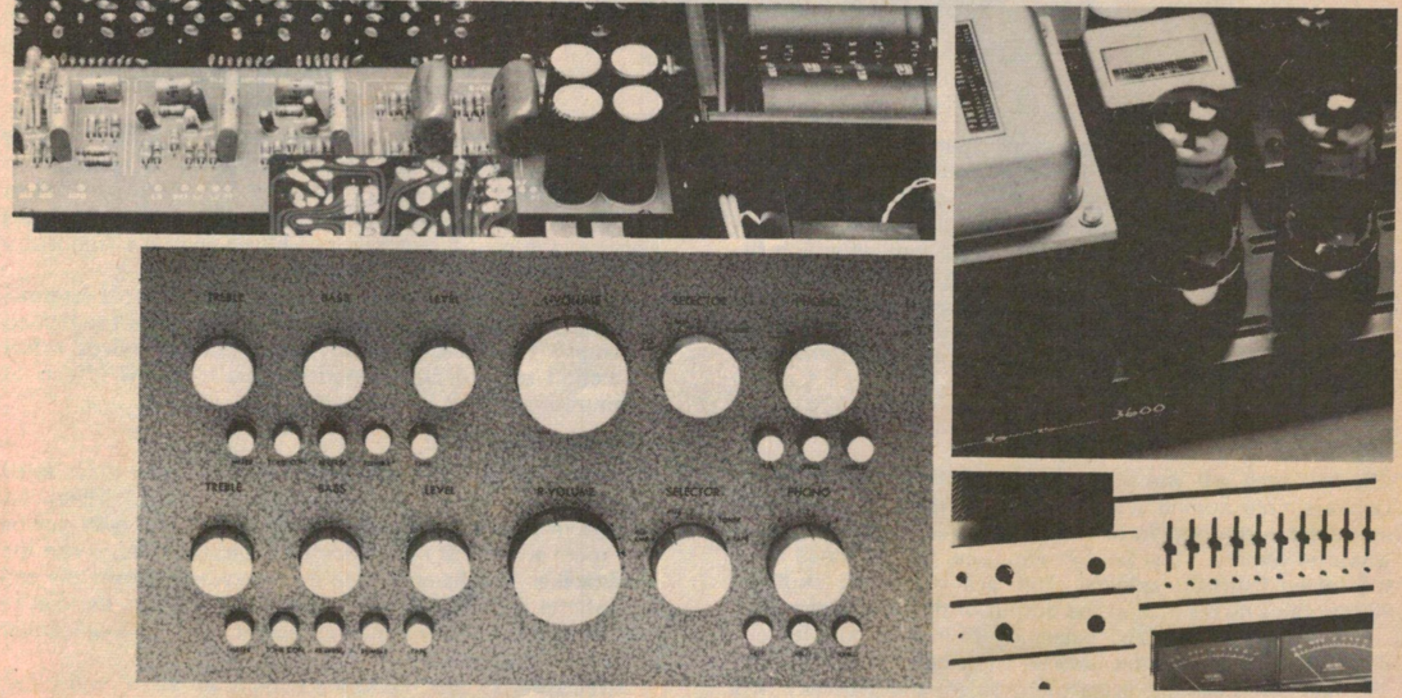
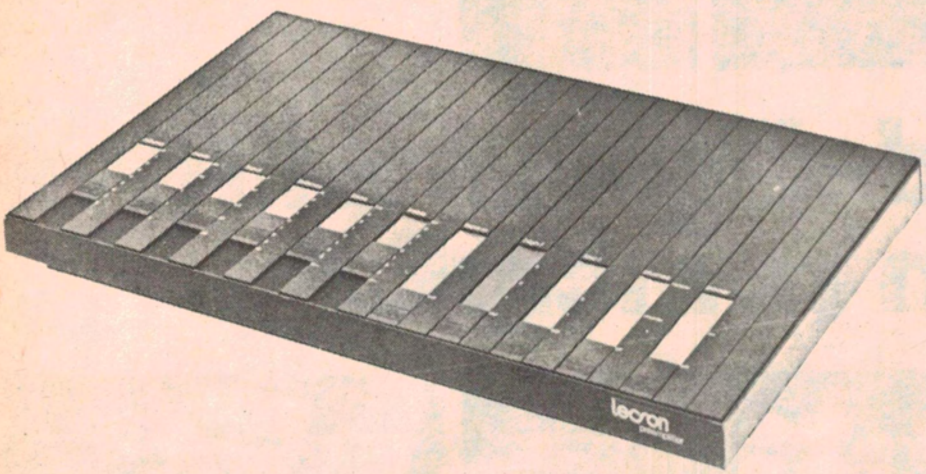
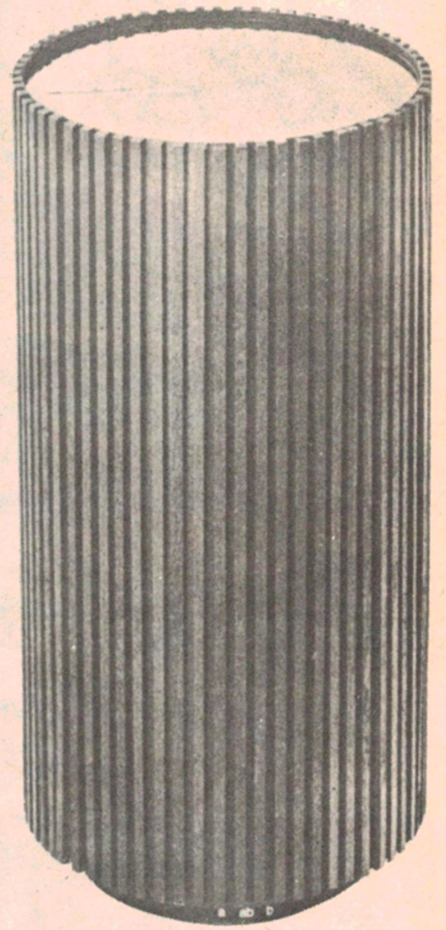
Ultra-Fidelity Amplifiers -design principles



Ultra-Fidelity -design principles

Audio amplifier design has come a long way since the introduction of semiconductors into hi-fi. Stan Curtis, who has been responsible for such excellent examples of the art as the Cambridge Audio and the Lecson, explains here the black arts of ultra hi-fi design.





CAREFUL listening tests have shown that while an amplifier that measures badly is *unlikely* to sound good one that measures well *cannot* be guaranteed to sound good. Thus it is apparent that the traditional measurements of power distortion and frequency response need supplementing by new and more powerful laboratory tests. Such tests should more closely relate to the conditions prevailing when the amplifier is driving realistic loads and using music signals rather than sine-waves, which of course represent only one special case.

Balancing Act

The first such test was popularised by Peter Walker of Quad. It is a simple nulling system which attempts to cancel the input and output signals of an amplifier. With full cancellation whatever remains must be distortion, i.e. signals added to or subtracted from the original. The ideal or perfect amplifier will produce no residual at the output of the nulling circuit.

In practical terms the balancing of this circuit is very difficult if a significant degree of accuracy is required. Thermal drifts can aggravate the problem and generally it is difficult to set up for more than one amplifier type as usually the whole phase-balance network needs to be recalculated and readjusted each time. However this simple circuit is useful for showing just how often amplifiers are clipping the signal in the course of a piece of music and how frequently some amplifiers slew-rate limit the signal.

However, with such high current capability it is essential that the amplifiers have speaker muting to prevent switch-on "thumps" (or more accurately, earthquakes) and dc offset protection to protect the loudspeakers from the effects of 20 amps of pure dc!

Offsetting Long Tails!

Dc offset has been a major problem with many dc coupled amplifiers (i.e. those having no output capacitor). The offset voltage measured across the output terminals should not be any more than ±50 mV. Once this voltage starts to rise the loudspeaker is subjected to a dc bias which moves the coil out of the central position. This in turn causes the coil to heat up and the power-handling capability of the loudspeaker to be restricted.

Eventually (and often sooner) the loudspeaker will blow.

Amplifiers

Many amplifiers have an offset voltage that is acceptable when the amplifier is first switched on but which starts to increase as the amplifier heats up. Such amplifiers are subject to thermal drift and this drift is normally due to a component mismatch in the circuit. The conventional amplifier, with a long-tailed pair at the input, is "theoretically" free of thermal drift as these will be automatically compensated for by the DC feedback.

However, this is on the assumption that the first two transistors (or FETs), forming the long-tailed pair, are perfectly matched.

The input offset voltage (upon which the output offset voltage is dependent) is related to the base-emitter voltage V_{BE} of each transistor.

$$\text{e.g. } V_{OS} = V_{BE1} - V_{BE2}$$

This difference can be made almost insignificant by using a dual-transistor or a monolithic integrated-circuit differential stage where matching is provided by the simultaneous adjacent fabrication of the two transistors. With discrete transistors, however, a close match is unlikely.

Similarly unbalanced output loading or mismatch of the collector resistors also increases the offset voltage. These mismatches also worsen the linearity (and hence the distortion) of this stage. Thus well designed amplifiers usually use 1% tolerance resistors in these positions and adopt balanced circuitry throughout.

The offset voltage is considerably reduced by the application of local dc feedback that occurs when emitter resistors are fitted. In this case;

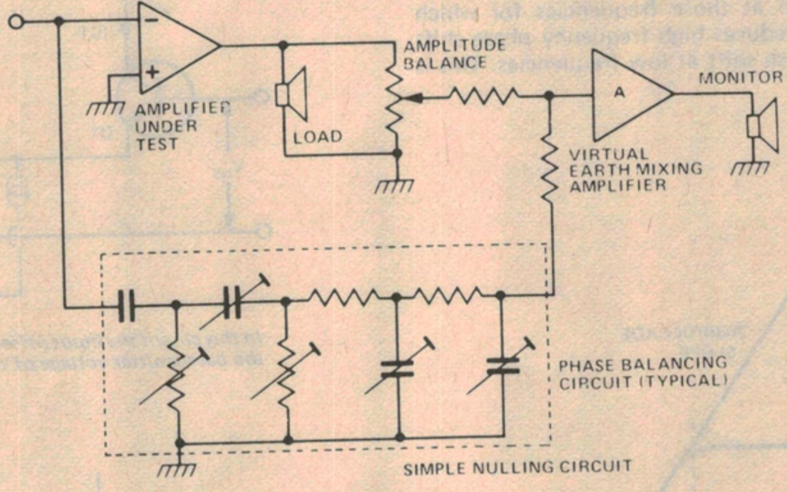
$$V_{OS} = V_{BE1} - V_{BE2} + I_{E1}R_{e1} - I_{E2}R_{e2}$$

and so by adjusting the balance between R_{e1} and R_{e2} with a trimpot a balance can be achieved.

Emitter Resistance

Note that $R_e = R_E + r_e$ is the total external emitter resistance and r_e is the transistor dynamic emitter resistance. Thus it can be seen that in the earlier typical example of a stage without emitter resistors, an imbalance of r_e and r_e will cause a worsening of the offset voltage. More importantly it can reduce the common mode rejection of the stage.

Of course the presence of emitter resistors also lowers the ac gain of the stage. For reasons to be discussed later this is not such a bad thing. This gain can be recovered by using bypass capacitors.



Block diagram of the Peter Walker balancing test.

Clip-on Off Set

Another situation where abnormal dc offset voltages occur is following a clipping overload. When many amplifiers are driven into clipping, the dc voltage of output rises towards one of the HT lines and then when the signal comes out of clipping the amplifier takes a finite time (often several seconds) to recover with the output dc voltage often oscillating between a positive and negative voltage before finally settling back to its nominal zero. Of course, when the amplifier is driven into clipping the normal negative feedback system ceases to control the amplifier.

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Thus the dc instability is indicative of poor low frequency stability in the amplifier. Some of the worst (but not all) amplifiers in this respect, have separate ac and dc feedback loops and so have big electrolytic capacitors (decoupling the ac loop) which take time to charge and discharge.

The old Cambridge P100 amplifier had this problem and the effect on the reproduction of a loud bass note can be imagined. Regrettably many amplifiers still suffer from this problem.

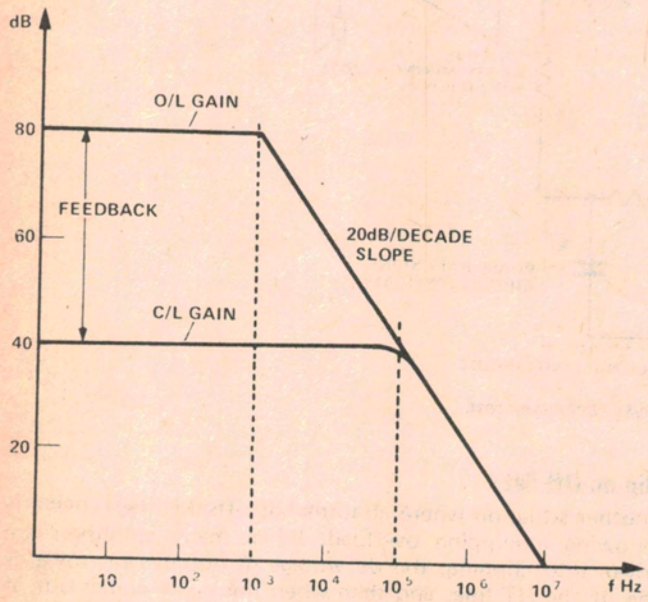
Quite often some amplifiers go unstable without their owners becoming aware of the problem. Sometimes the oscillation may be moderate in level and at a very high frequency; the only symptom being that the amplifier seems to run hotter and next-door's electric drill causes more TV interference than before!

Compensation Phase

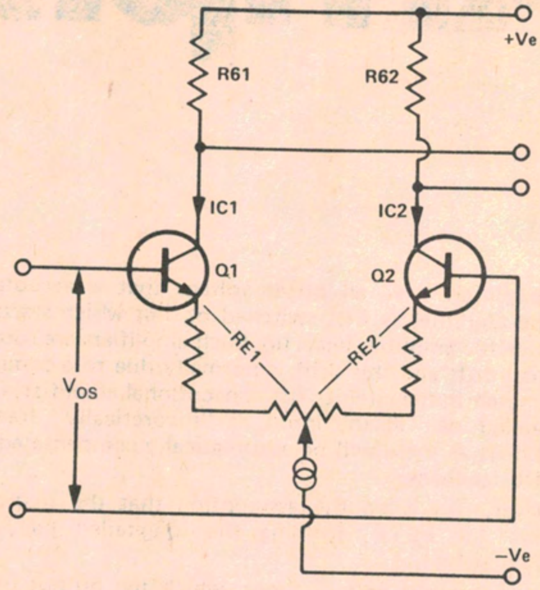
To know why some amplifiers are potentially unstable it is necessary to understand the principles of phase compensation. Much of the low distortion characteristics of amplifiers is achieved through negative feedback. If the phase shift around the feedback loop reaches 360 at any frequency at which the loop gain (i.e. the overall amplifier gain) is unity the result is a self-sustaining oscillation at that frequency.

The phase-inversion to provide negative feedback produces a stabilizing 180 (eg. "out of phase") phase shift, but an additional 180 can be developed in the amplifier.

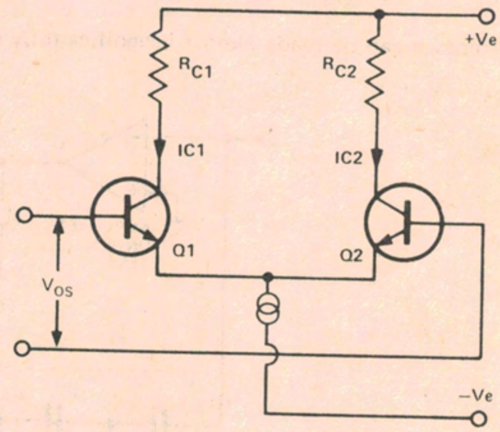
The phase shift developed through an amplifier is the combined phase shift of its several stages, and it usually develops 180 at higher frequencies. To ensure frequency stability under feedback conditions, phase compensation *reduces* the amplifier gain at those frequencies for which phase shift is high and it reduces high frequency phase shift by accepting a greater phase shift at low frequencies. This is



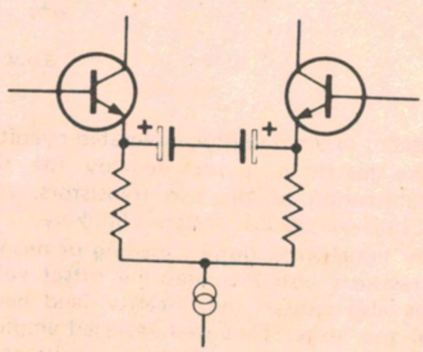
In the case shown in the diagram (unconditional stability) the open-loop response of the amplifier is stabilised by rolling it off at a slow 20 dB/decade slope with a single pole at 1 kHz. This amplifier would be stable with any amount of resistive feedback. However it will be seen that at higher audio frequencies the amount of feedback available reduces and so the distortion of the amplifier will increase. For this reason many amplifiers are of the "marginally stable" type.



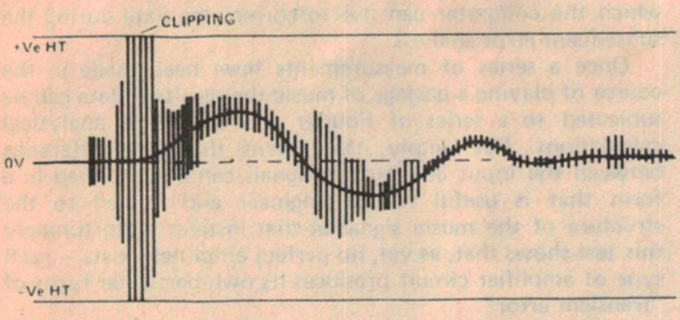
Differential pair with variable emitter resistances balanced by variation of the potentiometer.



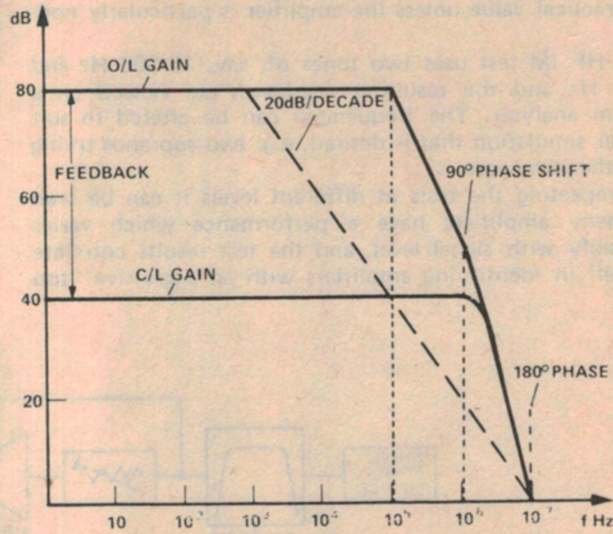
In this circuit the input offset voltage is related to the base-emitter voltage of this transistor.



Recovering lost gain by use of bypass capacitors across the emitter resistances.



Effect of a sine wave of varying amplitude as signal upon the dc offset voltage at the output.



In this case the amplifier has a fast roll-off which allows an improved closed loop performance at higher frequencies but without careful compensation they are not stable under all conditions of feedback. Once the phase shift reaches 180° the amplifier will become unstable so it can be seen that our example is only marginally stable.

accomplished by adding response poles and zeros in the form of resistor-capacitor networks (real or inherent in the transistors) in the amplifier circuitry.

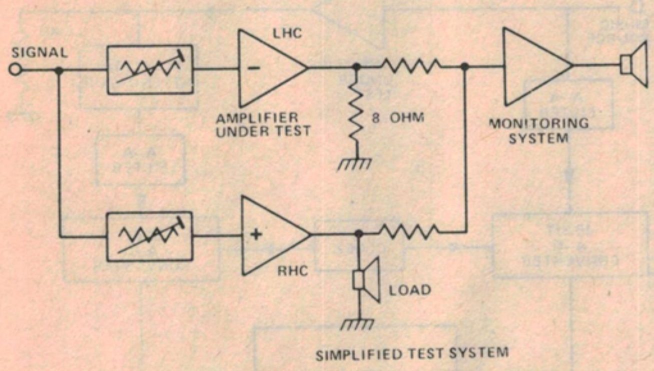
Equally important, to the owner of an expensive pair of loudspeakers, is the problem of high-frequency instability. These days very few high quality amplifiers are so unstable that they break into oscillation. However, quite a few respected units are on the edge of instability and so can potentially become unstable following a shift in operating conditions or of output loading.

Sum Theory

The author used another technique at Cambridge to investigate the changes in amplifier performance that are dependent upon the loudspeaker load. The two channels of a stereo amplifier are driven in mono but one channel is converted to become non-inverting. The outputs of both channels are summed and the resulting signal is monitored. Theoretically both channels should transmit the signal in the same way and (for a given circuit design) any distortion, time aberrations etc. should be the same for both channels. It is often quite possible to balance the two channels (driving 8 ohm resistive loads) so that the residual is inaudible. However when one 8 ohm load is replaced by a real "live" loudspeaker the residual betrays problems caused by the new load. In a refined form the test works well and reveals two interesting things;

i) the two channels of average amplifiers are rarely identical

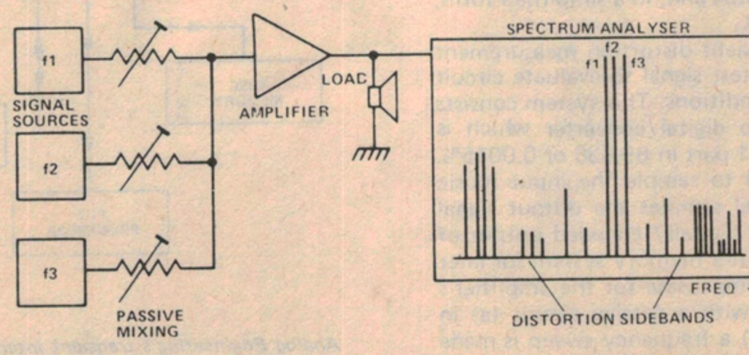
ii) some amplifiers work better in the inverting mode than in the non-inverting.



Using one channel as an inverting amplifier to monitor distortion produced by the design.

IM High

The conventional IM test uses an LF (50 Hz) and an HF (7 kHz) tone in a 4 to 1 ratio and then measures the sum-total of the sideband (e.g. distortion) components. This is of



Intermodulation distortion testing using three frequencies.

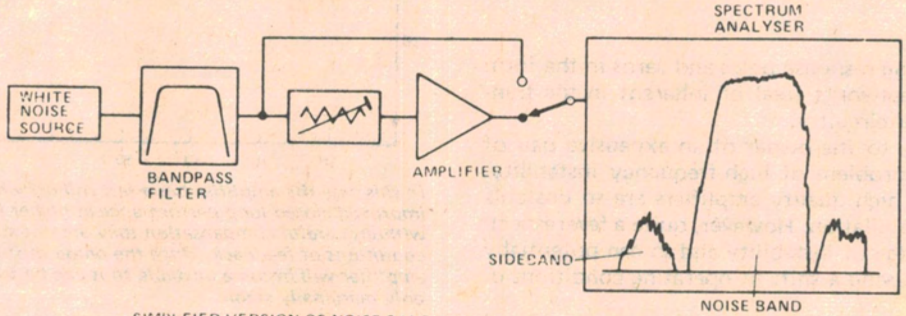
little practical value unless the amplifier is particularly non-linear.

The HF IM test uses two tones of, say, 15 000 Hz and 15 100 Hz and the resulting side-bands are viewed on a spectrum analyser. The frequencies can be altered to suit whatever simulation that is desired, e.g. two sopranos trying to sing the same note.

By repeating the tests at different levels it can be seen that many amplifiers have a performance which varies appreciably with signal level, and the test results correlate very well in identifying amplifiers with an aggressive "top end".

which the computer can use to correct the data during the subsequent error analysis.

Once a series of measurements have been made in the course of playing a passage of music the resultant data can be subjected to a series of Fourier and coherence analytical calculations. Put simply, this means that any difference between the input and output signals can be described in a form that is useful to the engineer and related to the structure of the music signal at that instant. Unfortunately this test shows that, as yet, no perfect amplifier exists — each type of amplifier circuit produces its own particular types of "transient error".



SIMPLIFIED VERSION OF NOISE BAND TEST

Noiseband testing with a spectrum analyser, the sidebands produced by the amp are clearly visible.

Dynamically Noisy

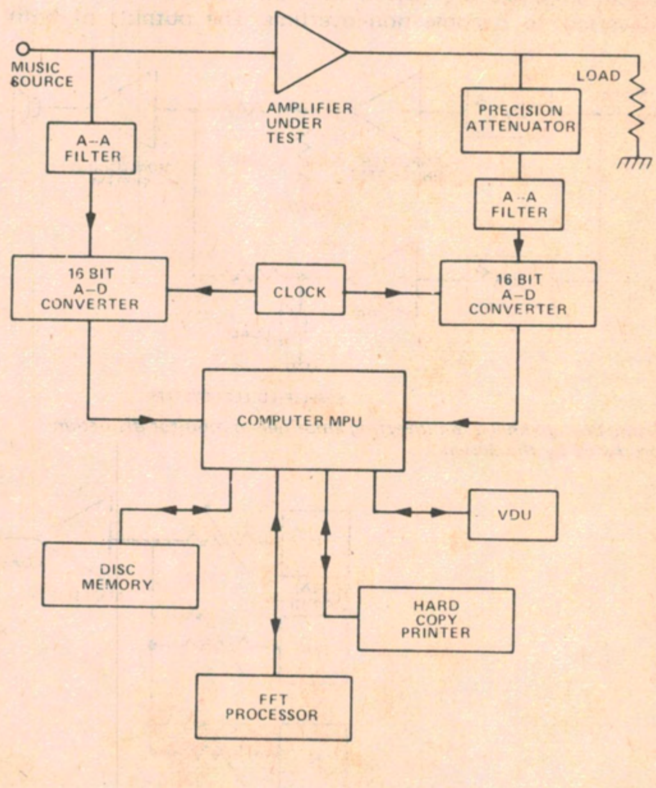
The second test is similar but attempts to measure the, amplifiers' performance under more varying "dynamic" conditions. A white noise source has a harmonic and amplitude structure which is variable and random and thus provides a better simulation of a music signal than does a sine-wave. The noise signal is passed through a bandpass filter to define its frequency response. The bandwidth and centre-frequency can be altered to suit the investigation as can the overall operating level. The output of the amplifier is fed to a spectrum analyser where the out of band components can be studied. Again this test is very useful for studying the effects of different loudspeaker loads but more significantly for subjecting the amplifier to random momentory "clipping" overloads.

A Channel and A Log

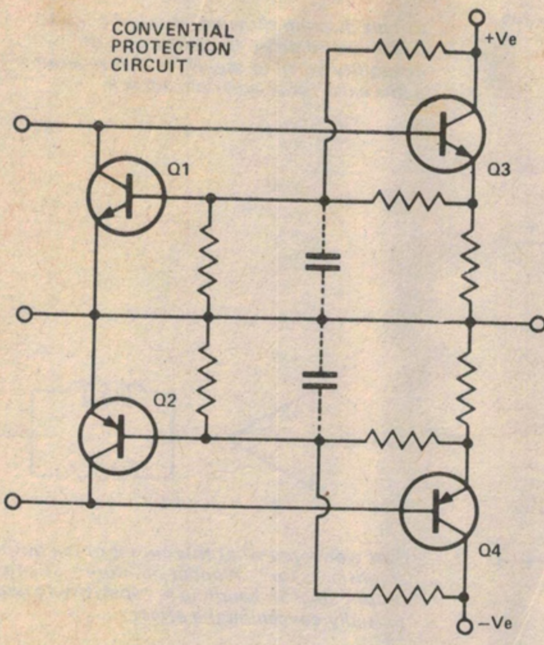
Possibly the most complex type of testing in use is a form of input and output signal comparison used by Analog Engineering Associates of the USA and, in a simplified form, by Mission Electronics in the UK.

AEA have developed a transient distortion measurement system that uses a music as a test signal to evaluate circuit performance under dynamic conditions. This system consists of a dual channel analogue to digital converter which is designed to have a resolution of 1 part in 65,536 or 0.0015%.

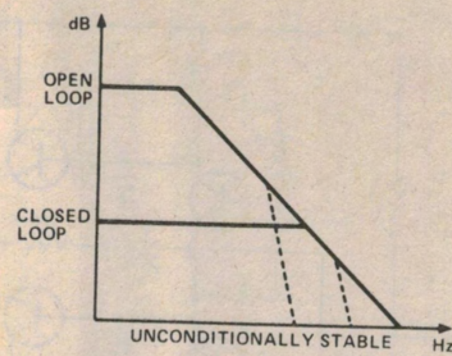
One channel of this is used to sample the input music signal whilst the second channel samples the output signal via a precision attenuator. The digitally encoded output of the convertors is fed to a computer memory system for later analysis. Instead of trying to compensate for the amplifier's phase and frequency response with a passive circuit (as in the earlier simple nulling circuit) a frequency sweep is made through the amplifier to generate a "transfer function"



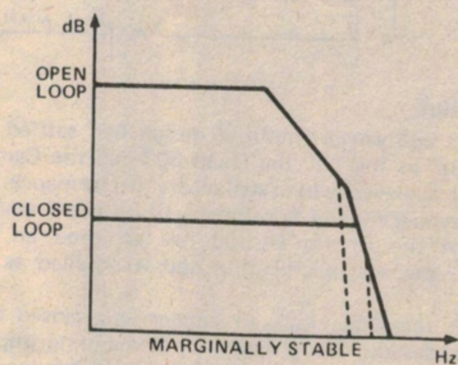
Analog Engineering's transient intermodulation distortion measurement system, used in Britain by Mission Electronics.

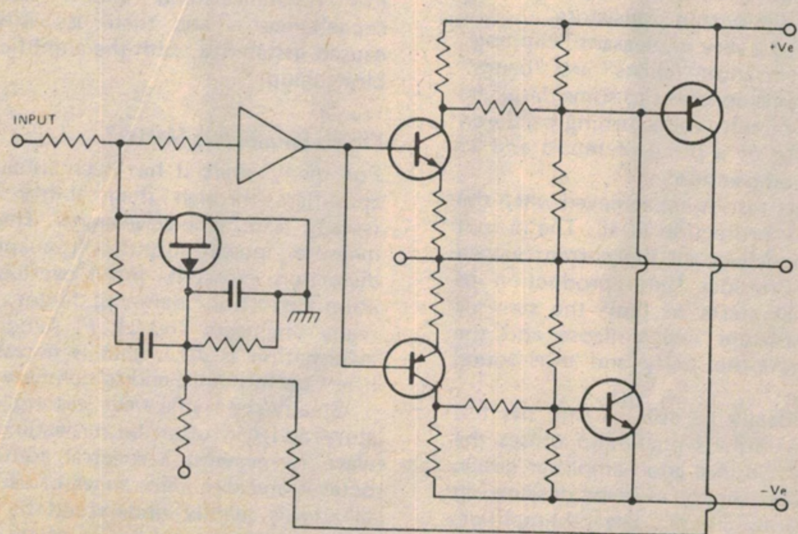


A study of the circuit of a conventional V - I protection circuit will show that as the protection transistors turn on they become a 'non-linear resistor' across the bases of output transistors Q3 and Q4 and as such create unpleasant distortion. One solution tried by some companies was to slug the bases of Q1 and Q2 with a capacitor to provide a time delay to prevent the protection operating except during a sustained short-circuit.



Above: Effect of adding an extra pole at the output of an unconditionally stable amplifier, such as might be added by a complex cross-over network. Below: Same condition applied to marginally stable type. Phase shift now borders on 180°, i.e. oscillation.





In this protection circuit the FET starts to turn-on when full-power is delivered into a 2 ohm load. The main advantage over a conventional protection circuit is that the limiting is "soft" (i.e. very gradual) and thus audibly acceptable and secondly that the distortion is much lower — and still only about 0.1% at limiting.

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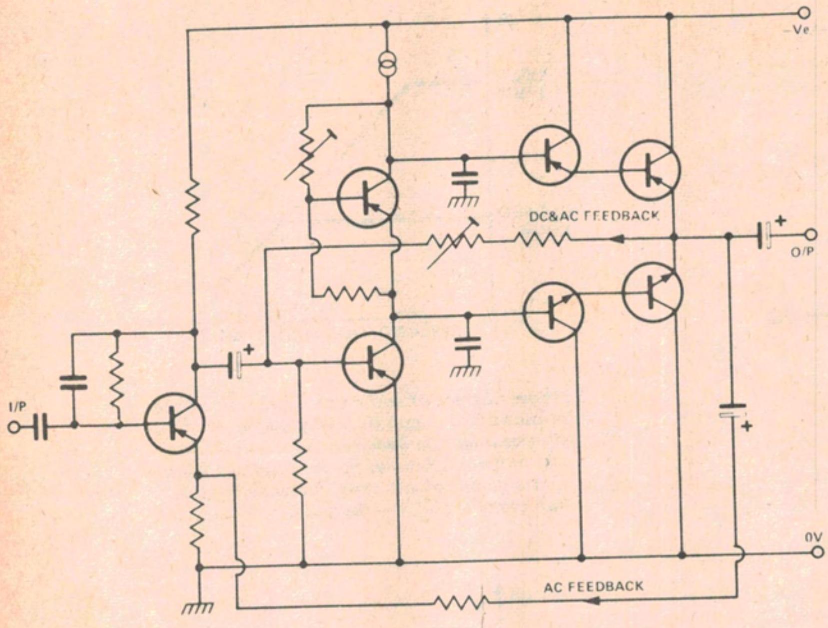
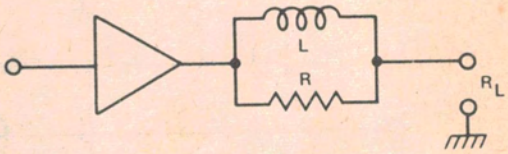


Circuit diagram showing a typical circuit which would prove to be prone to dc instability when in use. Note that separate paths exist for ac and dc feedback.



Ever wondered what this circuit in the output of an amplifier is for? Wonder no more — it's to aid the output stage in handling a capacitive loading by partially cancelling the effect.

Out of The Rut

A few years ago power-amplifier design had settled into a satisfying rut. In the UK the Quad 303 and the Cambridge P-Series had achieved very satisfactory performance figures and they were generally considered to be good amplifiers. In the USA the Crown DC300 has achieved an almost theoretically perfect specification and was hailed as "State of the Art".

However, the first crack to appear was caused by new loudspeaker designs. Some had very demanding impedance curves which in some cases presented a two ohm load to the amplifier. Such a low value of load (almost a short circuit to some minds!) operated protection circuits in many amplifiers, limiting the current to protect the output transistors.

The operation of these caused a very unpleasant "clipping" sound in some cases and even stranger "clicks" and "bangs" in others. Thus alerted it became apparent to some designers that conventional protection circuits were turning partly-on quite frequently in the course of a piece of music and so giving a sort of premature clipping action.

Without any doubt the best results are achieved when the output stage is devoid of any protection at all. The output stage should be designed to deliver all the current a load demands without limiting. Consider the reproduction of a bass drum. If the amplifier starts to limit the start of the "thump" the sound pressure will collapse and the bass-drum will appear to have no body and thus sound unrealistic.

The output-stage should ideally be able to sink the full energy of the power-supply until its regulation causes the current to limit progressively. So in a good amplifier design the output-stage and the power-supply must be designed as a single item and not as separate circuits. Several amplifiers *are* designed like this. The Lecson AP3 Mk II, the BGW models 500 and 750, and the Mission Power Amplifier. The Lecson AP3/11 can, for instance, deliver nearly 20 amps to the load before the mains fuse blows and the BGW model 750 even more.

If the amplifier now has to drive a capacitive load eg. electrostatic speakers, or complex crossover networks; another pole is added at the output.

In the case of the unconditionally stable amplifier the only ill-effect will be some "ringing" in the closed loop step response — but in the case of the marginally stable amplifier it may go completely unstable. The most popular "belt and braces" solution to this problem is to fit a resistor-inductor network at the output to "cancel-out" the effect of the capacitive loading.

It is interesting to note that some marginally stable amplifiers omit those components as most speaker cables have sufficient resistance and inductance. However, some of the new "Super-Cables" (Litz and Lucas, etc) have a very low resistance and almost no inductance but some capacitance — and their use with certain amplifiers has caused instability, with the amplifier (or speakers) eventually blowing-up!

Which Parameters Matter?

For many years it has been usual to specify and compare amplifiers through their ability to handle a continuous (steady state) sine-wave signal. Thus such a signal is used to measure power-output, frequency response, harmonic distortion, crosstalk, input overload capability, intermodulation distortion, damping factor, and gain! Unfortunately many engineers and Hi Fi pundits still believe that such information is ALL that is necessary to quantify an amplifiers performance and to compare it with others. Not so!

Steady-state sine-wave testing can tell only part of the story and can often be misleading. Music contains complex wave forms with a spectral content of greater than eight octaves and dynamic ranges of up to 100 dB. Yet such complexity is readily understood by the human brain which, in mastering the subtleties of spoken language, has evolved the ability of extraordinary auditory sensory perception. The music signal, as with all audio signals, can be considered in terms of two variable qualities — the frequency domain, and the time domain.

The frequency domain has monopolised engineers' thought

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for so long — even the most complex music signal can be represented by a Fourier analysis.

This mathematical equation lists separately each frequency making up the signal, (together with its phase and amplitude). However, a Fourier analysis is only complete in the case of simple waveforms, with more complex waveforms it becomes only a convenient approximation.

To make a Fourier analysis of a signal the components of that signal have to be analysed over a period of time such that complete cycles of the lowest frequency can occur. Thus we take consideration of the time domain.

Where steady-state signals are concerned the time domain is not normally considered, as the signal is of a continuous unchanging nature between any two periods. If the "time window", during which the signal is Fourier analysed, is reduced progressively it becomes apparent that an accurate spectral analysis becomes less possible. It can then be seen that the important characteristics of the signal are amplitude and rate of change. In other words its envelope.

What Do We Want?

What is required is the amplification of an audio waveform in such a way that the ear can detect no degradation.

Let us consider ways in which such degradation can occur. The waveform envelope can be distorted by amplitude changes of any component or by changes in the phase relationship of the component harmonics.

Experimental work has established that changes in the relative amplitudes of the harmonic structure of the waveform are readily detectable.

Other work has shown that the qualitative characteristics of a complex sound depend upon the phase relationships of the component harmonics. It would seem that as a phase difference must be interpreted as a time delay between the component parts of the signal, then a sufficient phase shift in a system must eventually become audible as these component parts are moved in respect to each other in time. In practice large phase shifts are very audible and indeed telephone lines are often phase and delay corrected to render speech intelligible. However, establishing an acceptable degree of phase shift is extremely difficult.

Following the arrival of "linear phase" loudspeakers great controversy has raged over whether phase shifts affect sound quality. A study of the experimental work performed to date shows that

1. It seems to be very difficult to replicate someone else's experiment.
2. It seems, on balance, that where recurrent waveforms (steady state) such as sine-waves (and instruments producing a "continuous" although decaying tone) are concerned; then quite large phase shifts, between the extremes of the frequency band, have no identifiable effect on sound quality. However, a phase non-linearity on the leading edge of a true transient appears to be audibly more perceptible, particularly on speech and percussive sounds.

Bandwidth and TID

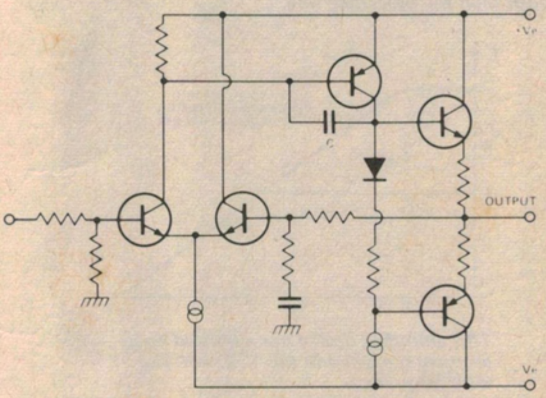
Transient signals cause many problems of which phase linearity is but one. Other problems include; instability and ringing, clipping, slew-rate limiting, and transient intermodulation distortion.

Transient intermodulation distortion (TID or TIM) is much in vogue but is often misunderstood. TID most commonly occurs when an amplifier, with overall negative feedback over several stages, is driven by a large enough signal whose frequency (or equivalent rise time) is above the open loop bandwidth of that amplifier.

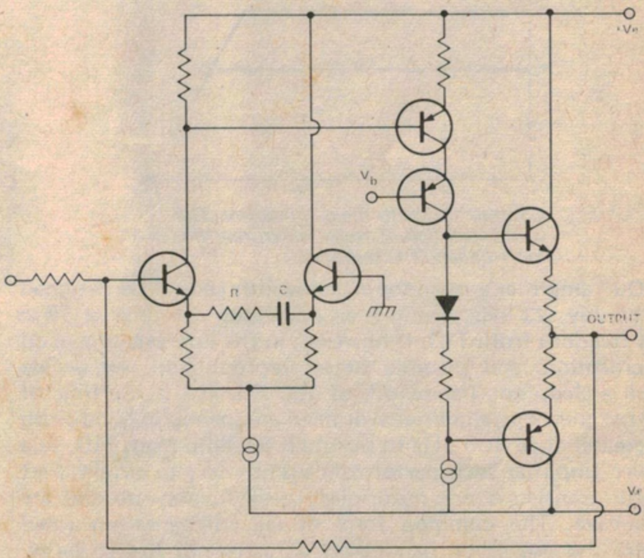
Because the feedback loop is fed from the output of the amplifier, there is no effective feedback until signal current flows at the output, i.e. during the open-loop rise time of the amplifier.

Very large signals occurring in the intermediate stages of the amplifier cause those stages to distort or even to clip. With some amplifiers this clipping can cause the stage to latch-up for a time until the operating conditions restabilise. Thus not only is the leading edge of the signal severely distorted — in some cases it is removed completely.

TID is therefore a form of overloading that is dependent upon both amplitude and time. It is audibly (but at a higher signal level) similar to cross-over distortion, as both effects cause phase and amplitude modulation of the signal due to momentary change in gain. (Remember that at the cross-over point zero, there is no current flow in the output stage and hence no feedback current and so the amplifier is momentarily open-loop.)



Circuit diagram of a typical amplifier circuit which employs lag compensation techniques — provided by C.



Lead compensation: components R and C provide the time constant.

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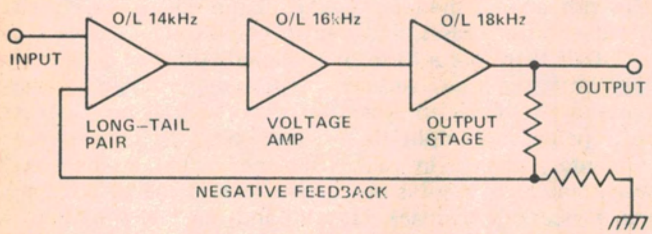
Making Big Bands

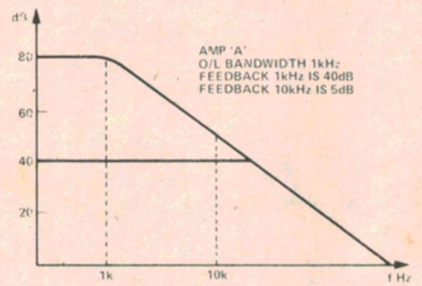
TID can be avoided by designing an amplifier whose open-loop bandwidth is greater than the highest frequency of the input signal. The maximum bandwidth can then be defined at the input by a passive RC filter. Thus if we decide upon a maximum signal bandwidth of 20 kHz than our filter will limit the signal waveform rise-time to T = 0.35.

$$T = \frac{0.35}{20\ \text{kHz}}$$

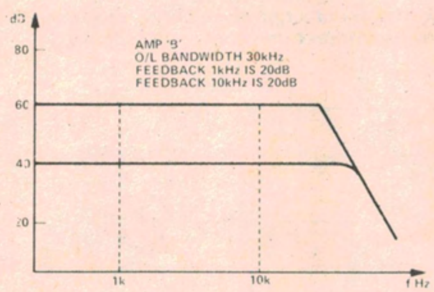
$$\text{i.e.}\quad 17.5\ \mu\text{s}.$$



Third method of avoiding TID. Each stage in the design has a wider bandwidth than the preceding one.



This amplifier design has a limited open loop bandwidth and the THD will rise with frequency.



Contrast this with the graph above. The bandwidth here is much wider, resulting in a more linear THD response.

Our amplifier's open-loop bandwidth should be designed to be, say, 23 kHz, giving it an open-loop rise-time of 15 μs and freedom from TID. If however, in the interests of a good specification, and possibly better reproduction, we decide upon a close-loop bandwidth of 100 kHz (i.e. a rise time of 3.5 μs) then our amplifier will need an open-loop bandwidth of greater than 100 kHz to maintain freedom from TID. In a power amplifier such performance is not easy to obtain. Fast power transistors are notoriously easy to blow-up and are expensive. The common form of lag compensation (used where the open-loop bandwidth is restricted) has to be replaced by lead compensation:–

Another technique is an extension of the first in that the preceeding stage of the power-amplifier is designed to have a lower open-loop band width than the next.

Important or Not?

Many people now consider that TID is unimportant or even that it doesn't exist. This is partly because it is very difficult to measure and only readily visible (in the laboratory) in the "clipping" state. To reach this stage with most amplifiers (but not TID – free designs) there is a requirement for either fast rise-time or higher signal levels or both, – conditions that are unlikely to occur in practice. However, a large degree of non-linearity and hence bad intermodulation will still occur with more realisable input signals. Although this cannot be measured yet (how do you measure say, 5% IM over a period of 5 milliseconds?) it can be predicted mathematically and, just as important, heard. Amplifiers free of TID have a very "open" quality with accuracy of depth.

An amplifier designed with a wide open-loop bandwidth, for low TID, often has other more tangible benefits. The high frequency THD is usually no higher than at the mid-point; in stark contrast to more traditional designs. This is because gain is still available at high frequencies for negative feedback. Such amplifiers also usually have much higher slew-rate.

Slew

Slew-rate defines the speed with which the amplifier can deliver output voltage to the load. For example, if an amplifier has a maximum output of 100 volts p/p and a rise-time of 100 μs, then the amplifier, if it were perfect, should have an output of about 80 volts after 10 μs in response to a suitable square wave input. In other words the output voltage would have risen at the rate of 8 V/μs. However, amplifiers do not generally respond to large changes as fast as their small signal characteristics predict, for circuit and transistor capacitances can be charged only as fast as their driving circuits allow.

In its simplest form the slew-rate of an amplifier defines how fast the output voltage can change for large signal conditions, and it is normally quoted in volts per micro second. The maximum slew-rate of an amplifier is usually limited by the slowest stage in its circuit.

That stage will have an operating current T (as set in the design) and a capacitance C (usually a frequency compensation capacitor)

$$\text{Slew-Rate} = \frac{T}{C}$$

Thus if a transistor stage has a standing current of 100 μA and is compensated by a 43 pF capacitor then its slew-rate will be

$$\frac{100}{33}$$

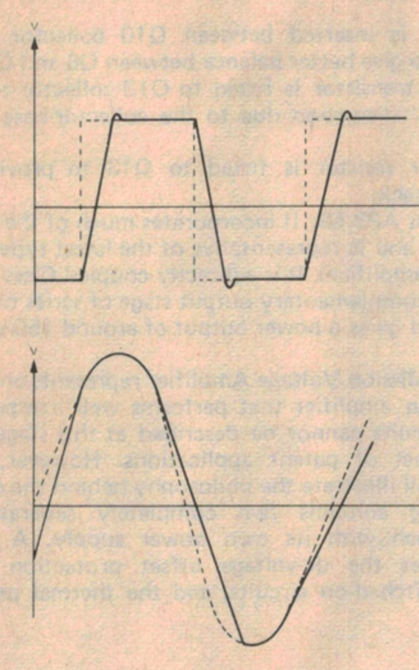
$$\text{i.e. } 3\ \text{V}/\mu\text{s}$$

Depending upon the design some circuits have a different slew-rate depending upon whether their output is negative-going or positive-going. Slew limiting also defines the full-power bandwidth; a figure more commonly quoted by manufacturers.

$$fp = \frac{SR\,(10^6)}{2\pi\,E\,op}$$

E op = peak output swing in volts
fp = Full power bandwidth in hertz.

Thus in a 100 watt (into 8 ohms) amplifier having full-power bandwidth of 20 kHz the required minimum slew-



The effects of slew-rate on a signal passing through an amplifier prone to this fault. Top: a squarewave, note the slight overshoot. Below that, a sinewave. In both cases the dotted line represents the input.

rate would be about 5 V/μs. This is, however, the absolute minimum figure and experience suggests that such an amplifier would have a hard, gritty high-frequency sound. Such an amplifier should have a slew-rate greater than 20 V/μs to be certain of avoiding the increase in distortion caused by the gradual onset of slew-limiting.

Unfortunately the higher the power output of the amplifier the greater the required slew-rate as more volts swing at the output in the same period of time and so as our 100 W amp needs 20 V/μs an otherwise identical 50 W amp needs 14 V/μs and a 20 W amp needs only 9 V/μs. But these forms of distortion tend to give subtle audible effects compared to the most common amplifier problem — that of clipping.

Clipping

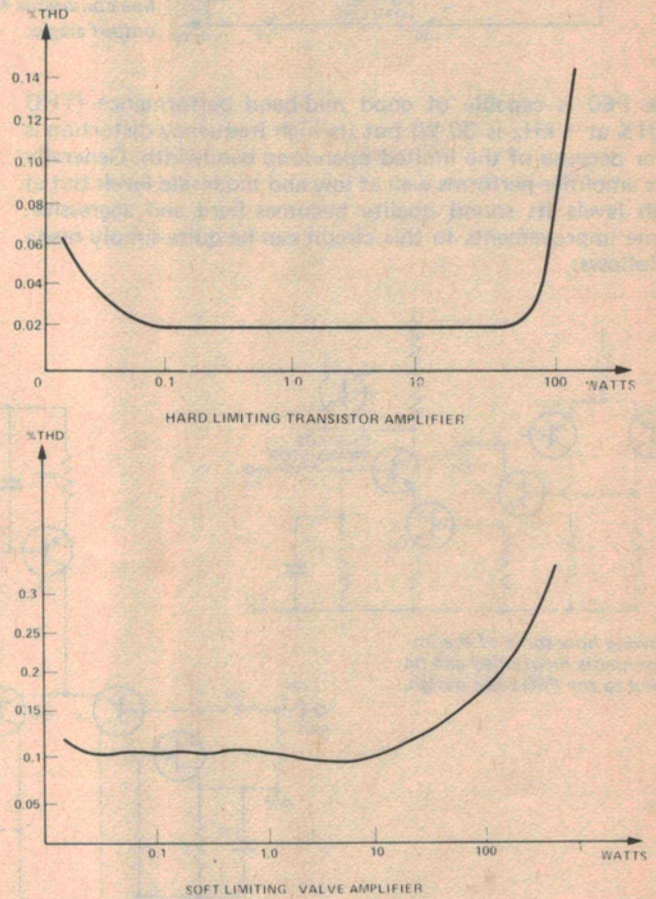
Clipping occurs when an amplifier is overloaded by high level signal peaks. Such peaks occur frequently in much music material and so the manner in which the amplifier clips determines its audibility. A soft, clipping effect where the distortion rises gradually (typical of valve amplifier circuits) is audibly preferable to the hard clipping typical of transistor circuits.

Worse still, some amplifiers tend to suffer saturation effects on clipping and take a time to recover; thus artificially extending the length of time the signal is clipped. The use of overall negative feedback to reduce distortion unfortunately makes things worse. Overall feedback effectively linearises the clipping — the distortion changes from 0.01% (say) to 10%, and quite suddenly too.

Design Procedure

We have covered just a few of the requirements a designer must consider when working upon the design of power-amplifiers. There are many more to be considered to even rough out a design specification before the circuit hardware is considered. The following sequence is mandatory:

1. What parameters are important to prevent audible degradation of the signal?
2. Detail a performance specification that meets the requirements of (1).
3. Decide upon the circuit technology necessary; Bipolar; MOSFET; Valve; Class A; Class B; Switching; etc; etc.
4. Undertake a development programme to produce a prototype.



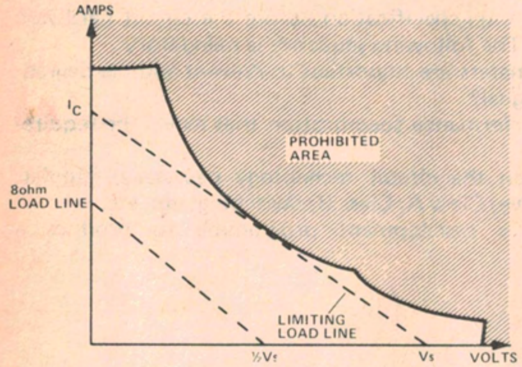
A comparison of the limiting characteristics — in general — of both transistor and valve amplifier types. There is a body of opinion which holds these curves to be the whole truth as to why valve amplifiers are preferred by many musicians.

At this point the designer has to accept that it's a real world and that his performance specification cannot be achieved in a way that is acceptable to accountants, salesmen, customers, customer's wives or whoever else is around. Trade-offs are necessary and much of the "art" is in deciding which defects and degradations are more acceptable than others.

As an illustration of the changes in design approach over the years we will briefly illustrate three designs for which the author has been responsible:

1. Cambridge Audio P60 (P80)
2. Lecson AP3 Mk II
3. Mission Electronics Voltage Amplifier

Ultra-Fidelity Amplifiers -design principles



Illustrating the load line conditions for output stages.

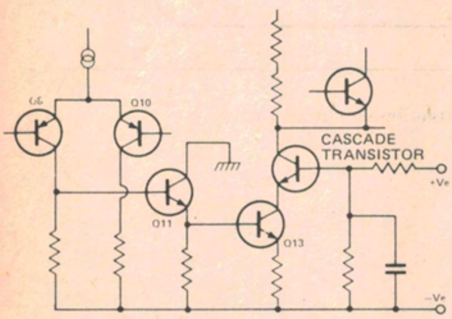
The P60 is capable of good mid-band performance (THD 0.01% at 1 kHz is 30 W) but its high frequency distortion is poor because of the limited open-loop bandwidth. Generally this amplifier performs well at low and moderate levels but at high levels its sound quality becomes hard and aggressive. Some improvements to this circuit can be quite simply made as follows:

1. A resistor is inserted between Q10 collector and the negative rail to give better balance between Q8 and Q10.
2. A cascade transistor is fitted to Q13 collector to reduce "early effect" distortion due to the collector-base capacitance of Q13.
3. An emitter resistor is fitted to Q13 to provide local negative feedback.

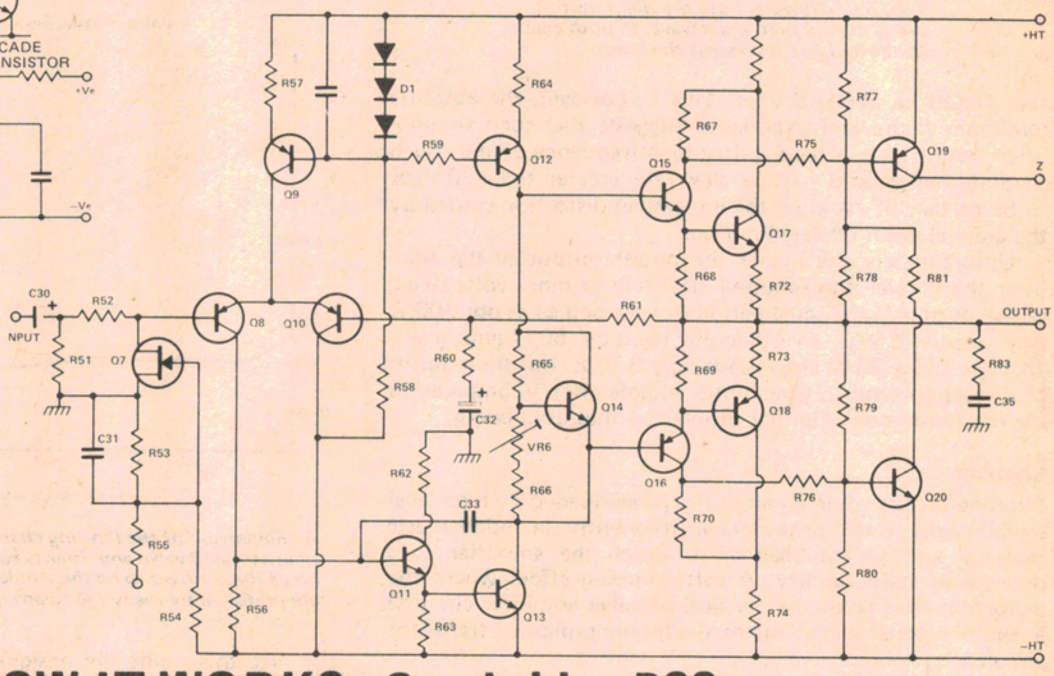
The Lecson AP3 Mk II incorporates much of the thinking in this article and is representative of the latest types of high performance amplifiers. It is a directly-coupled Class B design using a fully complementary output stage of series connected transistors and gives a power output of around 150 watts per channel.

The New Mission Voltage Amplifier represents an attempt to produce an amplifier that performs well irrespective of load. The circuits cannot be described at this stage as they are the subject of patent applications. However, a brief description will illustrate the philosophy behind the design.

The casing contains two completely separate mono amplifiers, each with its own power supply. A separate module carries the dc-voltage offset protection circuits; the delay switched-on circuits; and the thermal protection



Showing how some of the improvements mentioned can be added to the P60 basic design.



Full circuit diagram of the Cambridge P60 power amplifier design.

HOW IT WORKS—Cambridge P60

The P60 power amplifier is of a conventional design but with care being taken to optimise each stage. Q8 and Q10 form a long-tailed pair with Q9 as their emitter current source. Q8 and Q10 must be very closely matched for minimum DC offset and for maximum common-mode rejection to avoid H. T. ripple appearing at the output. The next stage is the Q13 voltage amplifier which is loaded by a current source (Q12) instead of the more common "bootstrapped" resistors. Note that Q13 is buffered from the long-tail pair by an emitter follower (Q11) to prevent any loading of that stage worsening the distortion characteristics.

Capacitor C33 gives lag compensation which defines the dominant pole of the amplifiers. The open-loop bandwidth is quite high (for this type of circuit) at 12 kHz but none the less this amplifier is prone to TID effects. The protection circuit is very unusual in that the output is limited by an FET (Q7). Q19 and Q20 each form conventional V-I summing circuits which monitor the loading on the output stage.

If either Q19 or Q20 turns-on, the gate of the FET Q7 (normally biased-off by R54 to the negative HT) is biased positive and it starts to turn-on. It then acts as a potential divider with R52 and thus attenuates the audio signal. This protection only turns on at the equivalent of 50 W into 2 Ohms load and when it turns on it only adds moderate distortion (0.2% typically) as distinct from clipping.

circuits. Particular attention has been paid in the design to achieving:

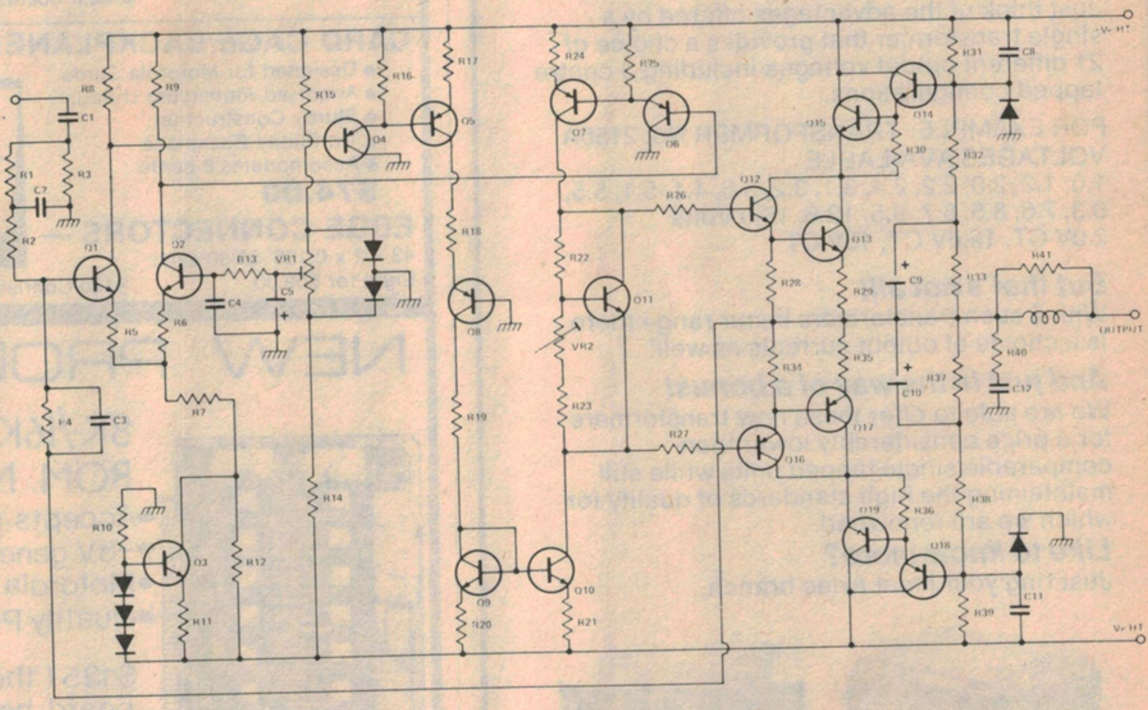
1. Low distortion with a very low order of overall feedback
2. Wide open-loop bandwidth with an excellent slewing rate
3. Minimum time and phase distortion
4. A high transient power capability with virtual freedom from clipping effects.

The output stages have a very high current capability but have no protection circuits, the output transistors being designed to sink the full energy of the power-supply into the load. A patented form of voltage feed to this stage gives the amplifier a short term power delivery capability of about 600 watts (compared to the rated 150 watts 8 ohms). This represents a 6 dB increase in power availability over the rated figure. The voltage amplifing stages are designed to clip softly and this combined with the low-overall feedback gives overload characteristics similar to those of an equivalent valve amplifier.

Conclusion

This feature has discussed just some aspects of modern audio amplifier design. At present much attention is still given to whether an amplifier is designed around bipolar transistors, FETs, valves, or switching transistors. However designers are beginning to appreciate that the major stumbling block is not designing a circuit using any of these technologies but in deciding upon what is the performance specification required *that will give faithful reproduction of the sound source.* Until this problem is solved there will continue to be an element of uncertainty in amplifier design.

The Mission Amplifier referred to in this article is due for release very soon now, and we will be taking a closer and more detailed look at this design — results as soon as possible in ETI.



Full circuit diagram for the Lecson AP3 power amplifier design, producing around 150W.

HOW IT WORKS—Lecson AP3

Transistors Q1 and Q2 form a long-tailed pair differential amplifier with Q3 as the emitter current source. Local feedback is applied in the form of emitter resistors R5 and R6. The base of Q2, instead of being grounded, is connected to a potential divider RV1 which permits the DC offset at the output to be set to zero. The input signal to Q1 is passed through a low-pass filter (R1, C2) which sets the bandwidth to 22 kHz (i.e. below the open loop bandwidth for no TID effects). The bi-phase outputs of the long-tail pair feed a second differential amplifier Q5 and Q7. Transistor Q5 has a constant current load (Q8) whilst is terminated by a current mirror (Q9 and Q10). Transistor Q10 will always deliver the same current as transistor Q9 hence the term "Current Mirror" and the excellent symmetry and balance this stage achieves. Functionally, however, Q10 can be considered as an active load whilst Q7 is a voltage amplifier from whose collector the drive to the output stage is taken. Note that Q5 and Q7 both have local emitter feedback (R17, R24) and that both are buffered from the long-tail pair (Q4 and Q6 emitter followers).

Transistors Q12, Q13, Q16 and Q17 each form conventional Darlington emitter follower stages. Each stage is series connected to a further power transistor (Q14, Q15 and Q18, Q19 respectively) which is permanently biased ON. Their emitter potentials are determined by the ratio of the base potential dividers. This ratio was chosen such that Q13 and Q15 each has half the supply rail across them.

The whole amplifier is in the inverting mode with overall shunt feedback through R4 and C3.

This amplifier is quite fast having an open-loop bandwidth of about 27 kHz. The circuit is stable without the usual compensation capacitors within the loop. THD is low being typically (at 100 W into 8 Ohms) 0.004% at 1 kHz and 0.02% at 10 kHz. The HF distortion can be further improved by selection of transistor Q7 for a device with a low collector-base capacitance.

No conventional protection circuits are used as extremely high power transistors are fitted and these can survive a short-circuit condition in the time taken for the power supply to shut down.