

A Class-A Amplifier for Home Constructors

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PART ONE – Design Considerations and a Circuit Description

Since my first articles in Hi-Fi News (November 1967, March 1968) many readers and various 'disciples' have come to recognise the benefits of Class A operation in power amplifiers for high fidelity use. I still find it strange to recall that so many horrible sounding transistorised amplifiers found their way into the market without cries of objection except from an enlightened few. Perhaps it is significant that certain correspondents of HFN can be placed among these protesters, including Norman Hulley, Geoffrey Horn and Harry Leeming, although these gentlemen do not always agree with the solutions proposed by the writer.

So much has been said on the subject, with Wireless World having published two Class A designs (1, 2), as well as an experimental circuit appearing in this magazine (3) – not to mention months of correspondence in both journals – that Frank Jones has insisted it is time for HFN to publish a Class A design for constructors (I haven't insisted, a number of readers have – RFJ). I apologise that it has taken so long to come to press, but several factors, including the desire that the very latest design should be proffered, have caused delay. For newcomers it may be appropriate briefly to recapitulate some of the arguments for and against Class A operation in power amplifiers.

By definition Class A is a method of operation in which all the output devices (transistors or valves) operate on the linear portions of their transfer characteristics all the time, drawing constant mean current from the power supply irrespective of the signal. A Class B amplifier, on the other hand, is one in which the output devices usually split positive and negative portions of the signal waveforms between them, each operating from a point of low initial standing current. Thus, the current drawn by the stage, from the power supply, rises with the signal drive.

Because the transfer from one output transistor to the other (in a transistor amplifier) is very difficult to control, all forms of Class B operation often result in greatly increased distortion at low signal levels, known as crossover distortion. Unfortunately the thermal limitations of early power transistors forced the industry into the adoption of heat-saving Class B designs with the attendant disadvantages of high order harmonic distortion and excessive intermodulation, resulting in 'transistor sound'.

It should be recalled that Class B was never used for high-fidelity valve amplifiers although it was, and still is, used where maximum power is required as in public address applications. One objection to Class A operation is that more heat is generated. Since constant power is consumed, and the power is either dissipated in the loudspeaker or as heat in the transistors, a Class A amplifier is coolest under full drive conditions (more of the power consumed is dissipated in the loudspeaker) while a Class B design is coolest with no drive. However, provided the design is sensible, heat is no real disadvantage and is still less than in a comparable valve amplifier since there is no dissipation from heaters. A transistor can also be swung further than a valve and this means that the theoretical figure for efficiency can be more closely approached.

The advantage of Class A is that distortion reduces with decreased signal drive and is virtually nil at low powers. In practice only musical peaks need 10 W or more, the average power fed to the loudspeakers being under 1/2 W at normal serious listening levels and much less for background music. With Class B operation distortion levels at best remain constant, and in some cases get worse, with reducing signal levels – a very sad state of affairs. Also, because of the discontinuities in the transfer characteristic, stability of the amplifier under transient conditions, into practical loads, is hard to achieve and maintain.

Of course, to the listener/amplifier purchaser, most of these considerations are irrelevant since he is simply buying a box which will do a job, and the manner in which it performs that task is less important than the results.

Class A amplifiers are now available with the same power output and facilities as their Class B counterparts and at about the same price, so economic arguments are not overwhelming.

Subjectively, despite the low level distortion considerations, Harry Leeming has taken the writer to task for not adopting a 'hard enough' sell. He suggests that the non-technical customer might gain the impression that Class A amplifiers sound better only at low volumes, and goes on to comment 'Certainly in my experience this is not true. The important thing about a Class A amplifier is that on loud orchestral passages the instruments do not seem to merge together' and Mr. Leeming's comments are very typical of the observations I have received.

Design inevitably started round the parallel push-pull circuit and it seems appropriate at this point to consider the action of this circuit. If all D.C. isolating components are ignored and the A.C. circuit and the circulating current shown, we observe fig. 1.

However, the final circuit arrived at need not necessarily be thought of in this way. It may be considered as a simple common emitter stage with a driven device for its collector load – the device driven so as to be helpful in preserving maximum efficiency. Mr. Linsley Hood's Class A amplifier is similar, except in so far as his output stage is more akin to the White cathode follower (fig. 2). His upper transistor is the 'driven' element acting as an emitter follower with the second device as the emitter load but driven again in such a manner to vary itself in a beneficial way. These concepts can of course only be realised in Class A for their very nature insists that both devices be always operating – load current sharing can never take place.

But early ideas centred round the more conventional approach with the upper transistor fed in anti-phase to the lower, necessitating a bootstrapped phase splitter (it is necessary to add the final output waveform to the drive waveform to the upper transistor in order to achieve a nett difference of the required drive waveform across the upper transistor base emitter circuit), see fig. 3.

It was then found how difficult or rather tedious it is to design an amplifier. The distortions and instabilities resulting from including that phase splitter were quite revealing! It was in the course of playing with the normal circuit that the idea developed of driving the upper half with a waveform proportional to the current supplied by the lower half. Each output transistor was augmented by an emitter follower driver and diodes have been added to the bias chain to provide some temperature stability – fig. 4.

The above output stage, when associated with the necessary "front end", was found to give all that was desired of it at that time. Load stability was excellent and remained excellent when negative feedback was supplied and when a few extra "tricks" had been added. These tricks included various H.F. response killing circuits to prevent supersonic and R.F. oscillations, and careful choice of the value of the first transistor coupling capacitor. Unlike a valve where the grid may be considered isolated from the rest of the device, the base of a transistor may not. The front end coupling capacitor must be large enough to provide the correct L.F. response (and it must be right first time for it is outside the feedback loop) but small enough to prevent any L.F. instability which occurs when the complete amplifier is fed from a very low impedance source as can happen in practice. This capacitor is C5 on the final circuit diagram.

It will be seen that the problem of driving the upper transistor is now considerably simplified. The waveform developed across R_d is added to the waveform at the upper transistor emitter as R_d is already connected to the upper transistor emitter, thus providing an ideal drive condition for the upper transistor. Furthermore, the method seems to reap some additional benefits in practical listening tests with practical loudspeakers. Clearly if the lower transistor is "made to work harder" due to load peculiarities, the additional waveform developed across R_d automatically makes the upper transistor work less hard in compensation and vice versa. A fixed drive phase splitter would not work quite like this.

Experience has shown that the original circuit had certain weak points. The first was that the dissipation of the power output stage was affected by the HT rail owing to the nature of the bias circuit, and excess dissipation was proportional to the square of the excess HT voltage. In addition, the diodes in the bias chain had to be selected for forward voltage, or individual preset resistors had to be used, and temperature stability was not completely taken care of by these components.

There were, as well, a number of unnecessary decoupling and coupling components, while for general purpose use it was thought desirable that the power amplifier should not be dependent on the design of the preamplifier from which it was fed.

The latest circuit is shown in fig. 5, and a detailed description of its functions follows. Not surprisingly the circuit is similar to that of the A21 Series 2 amplifier which is a commercial Class A amplifier that has been developed by the author's company.

The two channels operate from completely independent DC supplies, and the mains transformer has two separate centre-tapped secondary windings each feeding a full-wave rectifier circuit, delivering 36 V at the steady load current of 0.85 A.

Supplies to a preamplifier, if required, are taken from the 36 V rails and stabilised to 16 V with the Zener diode. The zener diode also has in parallel a 1,000 uF capacitor to provide additional ripple smoothing.

The first stage around TR1 is outside the feedback loop and is an emitter follower circuit to offer a high input impedance at the amplifier input, and to provide a relatively constant low impedance as seen looking back into it from the following stage. In between this emitter follower and the first stage proper of the amplifier is an H.F. filter network comprising R7 and C4. The purpose of this filter network is to restrict the rise time of any waveform which may be fed in to greater than 1 uS which is beneficial in providing general overall stability. It is unlikely that an input waveform with a rise time exceeding this figure could ever be applied to the amplifier except under laboratory test conditions, but the inclusion of the filter is felt desirable – and it has no aural effects. TR2 is a conventional grounded emitter amplifier, and provides a large part of the forward gain. TR3 is an emitter follower providing a low impedance drive source to the output stage.

This stage follows the principles previously explained, except that the quiescent current and the mid point voltage are now independently set. TR4 acts as the current controller as the potential developed across the lower output transistor emitter resistor R26 is fed via the potentiometer VR2 to its base, the potential resulting at the slider of VR2 must always be approximately 0.6 V. If the current in the output stage rises, then TR4 is turned 'on' harder and the bias potential applied to the base of TR7 is reduced, tending to restore the current to its correct value, similarly, if the output stage quiescent current is lower than its correct value, TR4 is turned 'on' less, raising the bias potential at TR7's base and thus increasing the quiescent current towards its correct value once more. R22, connected in parallel with the upper half of the potentiometer, is a 'fail safe' resistor in case the slider of potentiometer VR2 should become detached when the full voltage across R26 will be applied to TR4's base and the quiescent current will stabilise at a value of approximately 0.6 A.

C12 connected in parallel with TR4 is to ensure an initial slow rise of current when the amplifier is first switched on. This is safer for the amplifier itself and prevents a thunderous plop in the loudspeaker. As TR5 and TR6 may be considered as a compound emitter follower, the potential of TR5 base determines the potential of the 'mid point' to which the loudspeaker circuit is connected. It is desirable that this point is approximately half way in potential between zero and the +36 V line in order to permit the greatest possible excursion of the output waveform. VR1 is connected in the manner shown again to provide a fail safe facility, for even if the slider of VR1 should become detached the potential cannot vary greatly from its nominally correct value.

Overall negative feedback is provided by R13 connected between the main output point and the emitter of TR2. The gain of the amplifier is effectively determined by the ratio of R13+R12 to R12. R13 is shunted with C8 to provide an HF roll off above 20 kHz which is beneficial in maintaining overall stability with reactive loads. The inclusion of such a phase shifting network, along with other phase shifts caused by stray capacities etc., necessitates the inclusion of further killer networks which comprise C15 with R28, 29, 30 and 27, between TR8 collector and the 0V line, and R33 with C18 across the loudspeaker output terminals. It will be noted that the latter network becomes disconnected when the loudspeaker switch, SW2, is opened, but this is of no consequence as the main purpose of R33 and C18 is to prevent spurious oscillations when capacities of the order of 5,000 to 20,000 pF are connected directly to the output terminals, which may result from 'odd' connecting wires. No such capacity can be effective when using the headphone socket, as the series resistor R31 is quite sufficient to hold off such capacity.

The loudspeaker output is connected through C17, a capacitor of 2,500 uF value, which is amply large enough to maintain a sensibly flat L.F. response down to at least 30 Hz. The loudspeaker switch SW2 is included in case it should be desired to switch off the loudspeakers when using headphones, which are connected to the jack socket and fed through the network R32-R31.

The values of R32 and R31 have been empirically chosen as providing the best results with various headphone impedances, and it will be found that 8 ohm or even 300 ohm headphones work equally well. However to maintain damping on 8 ohm phones an alternative network is described on the circuit diagram.

It will be observed that the 0V line of the amplifier is not connected directly to the chassis, but through a 100 ohm resistor R34. In a normal practical case this zero line will eventually find its way to earth via the control unit earthing, and this method of connection ensures that a mains earth may be connected to various units employed with the amplifier without causing any hum loops.

Inputs and control unit power supplies are connected via a 5-pin DIN plug and socket. The socket is mounted on the amplifier chassis and uses the conventional connections of pins 2, 3 and 5 for common left hand and right hand inputs respectively. Pins 1 and 4 are used for left- and right-hand control unit supplies. Loudspeaker connections, to be described in detail next month, have been made using 4 mm sockets, though there is no reason why DIN loudspeaker sockets should not be used if so desired. Mains output connections are made through a conventional Bulgin chassis mounted plug and a three-way socket. Double pole mains switching is used in the interests of safety, and the mains transformer primary circuit is fused, which protects against transformer failure, shorted turns or secondary or rectifier failure. The fuses in the DC power supplies protect against amplifier failure. There is no necessity to protect further against misuse from the point of view of overload or incorrect loads such as short circuits. The current controller TR4 will always maintain the output transistor quiescent current at its correct value and thus the maximum dissipation at its correct value, no matter what happens. This is not an invitation to maintain full drive into short-circuits, but the amplifier is sensibly short/open/reactive circuit proof.

It will be observed that in certain places paralleled components have been used. There is no reason why R1 and R2 should not be replaced with one resistor of 1 W rating value 1 K, or C7 and C6 should not be replaced with one capacitor of value approximately 200 uF, or R28, 29, 30 and 27 should not be replaced with one non-inductive resistor of value 2.2 to 2.7 ohms. However, paralleling components in this manner often proves cheaper to the constructor – worth investigating.

Diode D2 connected between TR7 base and the collector of the current controlling transistor TR4 serves a dual purpose in assisting in overload protection and also 'catching' the base of TR7 at TR4 collector potential during switch on, again tending to prevent a large and sudden rise in output stage quiescent current. During normal operation the amplitude of the waveform appearing at TR7 base is much less than the 0.6 V forward voltage of D2. C13 provides a large degree of shunt feedback round TR4 effectively removing any AC component in the voltage fed to its base which tends to appear, especially at low frequencies, when the impedance of C14 becomes comparable to the emitter resistor R27. R26 – 0.7 ohms, is the resistor across which the drive waveform is developed to feed the upper half of the output stage. This drive waveform is applied through C11 and R17 to the base of TR5.

If the circuit is wired UP in conventional manner, the earthy line should follow approximately the series of connections as on the circuit diagram, but the loudspeaker earth return and R12 should come together to the common line at the same point. It is recommended however that the construction details and printed boards which will be described in full detail next month should be used, and impatient readers will find the author less than happy to commiserate if troubles are experienced in the meantime!

REFERENCES

1. 'A Simple Class A Amplifier.' J. Linsley Hood, *Wireless World*, April 1969.
2. 'Ultra Low-Distortion Class A Amplifier' L. Nelson-Jones, *Wireless World*, March 1970.
3. 'An Experimental Class A Amplifier.' D. H. Reed. *Hi-Fi News*, April 1968.

PART TWO – Building and Testing

The majority of the second article was devoted to the design and assembly of a particular enclosure for this amplifier. The enclosure was available in kit form at the time the article was originally published but is obviously now obsolete. I have therefore only reproduced the final section on 'testing and commissioning' – Geoff

Testing and Commissioning

The 'adjust mid-point' trimmers VR1 on each power amplifier panel should be set in their mid-positions, and the 'adjust current' trimmers VR2 on each panel should be set fully clockwise. With no input or loudspeaker loads connected, the unit should be switched on and voltage checks made using a 20,000 ohms per volt multimeter such as an Avo 8. The HT rails measured either at the fuses on the power supply panel or by measuring on the cans of the power transistors TR6 should be 36 V or very slightly higher. The voltages measured at the cans of the lower power transistors i.e. TR8 should be approximately 18 V. Assuming these are correct, the currents in each output stage should now be set up. Switch off the amplifier and withdraw one of the H.T. fuses, and insert an ammeter in circuit in its place. Check that the ammeter connections are not shorting to any other point, and switch on the amplifier. The current should rise fairly rapidly to about 0.5 A and then gradually to about 0.7 – 0.8 A. The 'adjust current' trimmer on the appropriate channel should now be gradually turned until the current reads 0.9 A max. Switch off the amplifier, replace the fuse, remove the other H.T. fuse and re-connect the ammeter; the same procedure is now followed for the other channel. Switch off the amplifier, replace the fuse and switch the amplifier on again, and adjust the 'mid point' pre-set potentiometers VR1 for voltages of 18 (or half the H.T. rail should this be slightly different to 36 V) on each of the lower power transistor cans (i.e. TR8). Check that the voltage at the pre-amplifier supply points i.e. pins 1 and 4 of the DIN sockets is 16 to 18. The amplifier should now be ready for use, if you have further test equipment such as A.C. voltmeter, generator, oscilloscope, etc. the more usual routine tests may, of course, be carried out.

Finally the headphone outlet. The author is in dispute with the Features Editor here since he believes that a moving-coil headphone is a current operated device and does not require a low Z source. For this reason he uses a high Z attenuator, but to satisfy the F.E.'s point of view an alternative scheme is shown on the circuit. The wiring, though, is not as shown, and with the low Z arrangement the output capacitor must feed the change over contact, with the loudspeaker terminals fitted to the S contacts and the 30 ohm resistors to the P contacts. The tinned wire links between the CO and P contacts are then omitted, and a 1.2 K resistor is placed across each pair of LS sockets to maintain a polarising voltage across the output capacitors.

Abridged Specification

Power output into 8 ohm load, prior to clipping	12.5 W
Harmonic distortion at 10 W into 8 ohm load	0.1%
Frequency and power response	30 Hz-20 kHz (+0, -1 dB)
Signal-to-noise ratio	80 dB
Input for 10 W output	420 mV
Input impedance	250 K
Output impedance	0.25 ohms

All measurements refer to one channel, both channels driven.

Figures



