ADVANCED PRE-AMPLIFIER DESIGN

If that was an "Advanced preamplifier design" in your November 1976 issue then I can only hope that when it is fully developed it will look different from the circuit published.

First a few fundamentals:

1. Magnetic cartridges give output voltages dependent on the velocity of the needle; keeping the recorded amplitude fairly constant with frequency, the record makers therefore force the output of the cartridge to rise at +6dB/octave.
2. Normal cartridges today, because of development in magnetic materials (stronger, smaller magnets), give outputs of much more than 2mV, around 10mV at 1kHz for 5cm/s velocity.
3. If the disc is cut with an overhead of +20dB (peaks of 50cm/s) and the frequency is 20kHz not 1kHz, giving another rise in output of +20dB, then you can see the signal at 20kHz can be 1V.

Reality is not as bad as this since the spectral density of music is not constant with frequency and falls off at high frequencies. However, outputs from cartridges do rise to 200mV peaks and do have fast slew rates.

Mr Self's talk of overload margins is a little confused when he compares amplifier performance. If the normal operating (0dB) level of an amplifier is 10mV input then to cope with 1V inputs there must be no limiting of distortion anywhere before a gain control for signals of +40dB above normal. This is best known as an overhead of 40dB and *is required at 20kHz* relative to 1kHz.

Now RIAA amplifiers have some peculiar problems coping with the high transient signals from magnetic cartridges just because the output does rise with frequency: this rise causes a high spectral density of high frequency signals and high slew rates. The prime requirements in the input stages are therefore wide bandwidth (to give fast slew rate) and low transient distortion when handling the excess high frequency spectral density.

Mr Self's preamplifier does little for either of these: the open loop bandwidth is not clearly defined. If the second stage is guessed at 100 then the stage has a —3dB point of 3kHz. The bandwidth of the amplifier is further limited by the input capacitor (1n5) and by the output loading network R_1/C_1 on the output: in fact what can the amplifier drive into C_1 at 20kHz to give a respectable overhead margin?

More problems!

The input impedance will fall rapidly to high frequencies because the output signal is fed back via 10nF to the emitter of Tr_1 then by 1.5nF to the input itself. Therefore the magnet won't be given a chance to generate the correct h.f. signals, for to do so it must have a resistive load right up to 20kHz.

More problems!

The first two transistors are connected in a classic phase shift oscillator configuration. I have often had this configuration burst into l.f. oscillations when fed from a low impedance (which a cartridge has at l.f.). The reason is simple: there are two phase shift networks, first the r_e of Tr_2 and the decoupling capacitor 22μF ($\phi = 90°$ below 10Hz), the second the resistor 220k and the input capacitor 1μF ($\phi = 90°$ at 0.1Hz). Thus towards l.f. even if the circuit has insufficient loop gain to oscillate (it fortunately has by a factor of about four) it will have a characteristic l.f. peak of a few dB.

All amplifier designs of this type have some sort of l.f. peak; it could be suppressed by increasing the 22μF to 2200μF, thus reducing the feedback by 100 times, or best of all don't use this configuration.

Actually the component values don't seem to have been chosen consistently: the input capacitor of 1μF has a f–3dB point of about 1Hz but the decoupling capacitor has a f–3dB point about 100Hz which is rather the wrong way round to achieve a proper control of the l.f. response.

Only one more eyebrow to raise on the input amplifier! I quote, "insufficient cut at frequencies above 10kHz" (to give the correct RIAA which should be 6dB/oct. fall from 2.1kHz to >50kHz). I shudder to think what is happening to this amplifier's phase response with all the "tricky dicky" empirical networks hung on it. This really is the last straw....

Shall I go on to the l.f. amplifier? O.K., I will. But first some comments on the system.

I don't agree with the gain control where it is, the amount of gain following it is over 65dB at maximum bass boost. No matter how good the noise performance of Tr_4/Tr_5 some l.f. hum and noise will be present at the output *all* the time. By all means vary the input preset gain to allow for high output cartridges but the system volume control must be later on in the chain, or does Mr Self have another control on his power amplifier? The l.f. boost amplifier is a nightmare: why not use any one of the perfectly good op-amps available (L148T1, TBA231)? Why use a design with an obviously wide bandwidth and enormously high gain to do a job that a lower bandwidth, lower gain (more stable) amplifier can do? There isn't, you see, the problem in this stage of lots of h.f. spectral density and fast slew rates — this has all been removed by the input amplifier!

The design here has the following major problems:

1. The open loop gain depends on the transistor h_{FE}s (very variable).
2. The open loop compensation is not calculated to ensure good transient response and/or stability. Is it calculated?
3. The response of the network 270k + 22k + (1n5//12nF) does not give anything like the correct l.f. response for RIAA. This should start to fall at 50Hz, all 20dB at 6dB/oct. to 500Hz then go flat to >20kHz. Mr Self's circuit, if he wants to know, starts to fall at 37.4Hz and falls at 6dB/oct. for 24dB.

Finally, the tone control is the usual "Baxandall" horror, for two reasons. The first, the lift and cut of ±15dB is too large, giving audible phase shift problems, and anyway whose power amplifier can handle more than 10dB? The other reason is that the bass lift and cut varies both amplitude and frequency at once. On top of which there is the absurdity of providing selected treble roll frequencies alongside completely unknown and variable bass-roll frequencies!

O.K. I am willing to accept the challenge, if *Wireless World* is. [Yes — Editor.] I will describe my alternative version of preamplifier, with details of each design decision and performance objective.

Until then, Mr Self...?

A. J. Watts,
SGS-ATES (United Kingdom) Ltd,
Aylesbury,
Bucks.

Mr Self replies:
To deal with Mr Watts' main points in the order that he makes them:

He is correct in stating that the outputs from cartridges have high frequency peaks and large slew rates, and that this represents a potential problem in the design of RIAA-equalized disc input stages. However, if the treble-cut portion of the RIAA curve is incorporated in the first stage, in the form of frequency-dependent negative feedback, the falling high-frequency gain means that the signal the stage puts out is substantially "tamed" and so enormous slew rates are simply not required; the open-loop bandwidth of the published disc input stage is quite adequate.

He is wrong in stating that the closed loop bandwidth is limited by the 1n5 input capacitor; this component, in conjunction with the associated 820Ω resistor, forms an r.f. attenuation network to prevent breakthrough of radio signals, and has no effect within the audio band. This is because the input stage is in a series feedback configuration, and hence almost the same signal voltage appears on the emitter of the first transistor as at the base, due to the high open-loop again; hence at audio frequencies the capacitance is "bootstrapped" and has no effect.

Similarly Mr Watts is incorrect in saying that the input impedance of this stage will fall significantly at high audio frequencies. A.c. feedback is returned to the emitter of the first transistor, and not the base; this series feedback raises the input impedance of the stage, in accordance with the elementary laws of feedback, so that it has a negligible effect on the impedance seen by the cartridge, which is completely defined by the parallel combination of the 68k and 220k resistors. This gives a constant impedance across the audio band.

The first two transistors are not connected in a classic phase shift oscillator configuration; this requires three RC networks, not two. Hence the circuit cannot oscillate at low frequencies, though it is possible for diminishing phase margins at low frequencies to cause an l.f. hump, if the d.c. feedback time constants are poorly chosen. This is why the input and decoupling time constants are markedly different. I would prefer not to comment on Mr Watts' phase-shifts and frequencies as of course a single pole cannot ever give a 90° lag; it can only approach it asymptotically.

If a low gain input stage is used to allow a very high overload margin, then there will always be a problem in persuading the stage to give less than unity gain at the highest extremes of the RIAA curve. The extra treble cut network (560Ω and 6n8) does not alter the overall phase response, as its extra phase lag is compensated for by the falling phase lag of the input stage due to the h.f. gain levelling out at unity. Since we are dealing with a minimum-phase system (in the sense of having no all-pass filters), then the amplitude/frequency response completely defines the phase/frequency response. In other words, if the RIAA curve is correct, then the phase response will be indistinguishable from that of a more conventional circuit using only one treble-cut time constant.

And now to the next stage...

Mr Watts appears to have overlooked the system volume control at the end of the preamplifier chain; one can hardly have a volume control later in the proceedings than this. Since this control is used for day-to-day volume manipulation, and hence is rarely fully up, the residual hum and noise is attenuated with the signal, as Mr Watts suggests, and the desirable "zero noise at zero volume setting" condition is in fact attained.

If this stage is a nightmare to Mr Watts then I venture to suggest he will find trying to extract the same performance from a TBA231 even more of a bad dream. Integrated circuit operational amplifiers were not chosen as they give an inferior noise performance, due to the processes involved in integrating the input stages, and in general only accept lower supply voltages, hence giving less overload margin. As for the "major problems": 1. The open-loop gain certainly does depend on the transistor current gains. However, since this is the case for every amplifier ever built, I am unrepentant. To return to the laws of feedback, one of the prime motivations of negative feedback is to render closed-loop gain predictable by making the effect of open-loop gain changes negligible.

2. If Mr Watts can calculate the phase and gain stability margins of this stage, then I shall be interested to see his results. I find a flat assertion unconvincing and I imagine others will too.

3. If Mr Watts rechecks his calculations, or better still, measures the actual circuit instead of theorising, he will find that the combined response of the first two stages is very close indeed to the RIAA curve.

As for the tone control stage, I suggest it is probably impossible to design a tone control without phase shift.

As explained in the text, the variable turn-over frequency over the bass control is advantageous rather than otherwise. I fail to see how this makes the provision of switched treble turn-over frequencies "absurd".

In conclusion, I can only say that I would like to thank Mr Watts for the friendly and constructive nature of his comments. I can hardly wait to see his own preamplifier design.

CITIZENS'-BAND IN THE UK?

I note with regret that R. C. S. Withers' organization (UK Citizens' Band Campaign), is advocating the use of 27MHz for a citizens' band service in the United Kingdom ("Letters" December 1976).

Such a service is essentially short range and therefore the selected frequency range should not be one usable for long distance communication when the maximum usable frequency is high.

A u.h.f. band remote from broadcast television and amateur frequencies would be a first choice. Alternatively a v.h.f. band could be used but there would appear to be many demands for the use of v.h.f. for other services.

There exists a Citizens' Band Association which is promoting the establishment of a v.h.f./u.h.f. citizens' band service in the United Kingdom. They have published proposals for a service, including a technical specification.

H. Turner,
Derby.

ADVANCED PREAMPLIFIER DESIGN

The letter from Mr Watts in your February issue and the answer from Mr Self is notable for two factors — the abrasive language of the former's criticism and the surprisingly temperate reply from the latter. Frankly, I, too, could find much to fault in the design but, of course, there are ways of expressing it, aren't there?

My main criticism of Mr Self's design is that it is over-engineered, conceived by a hi-fi enthusiast who apparently has not been too involved in the costing process when putting together the elements of a circuit. The principle of Occam's Razor is also the essence of good design technique. He has also overlooked the simple facts of life — that despite the extremes to which one may go in designing equipment of this type, the aberrations that are inherent in all programme sources available to the domestic user are likely to be far greater than those introduced by even the most modestly designed reproducing equipment.

But Mr Watts is guilty of worse errors, in dealing with pure theory, opinion, and dressing it up as fact. Let me take one example — and since he seems to invite challenges, here's another from me. If he is able to produce for me a *high grade pickup cartridge* capable of the sort of amplitude linearity input when correctly loaded that he insists should be observed in the equalised input stage and will deliver consistently peaks in excess of 200mV, then there is £5 ready in my hand for any charity he cares to name.

Reg Williamson,
Norwich.

ADVANCED PRE-AMPLIFIER DESIGN

In reply to Mr Williamson (Letters, April), I think there are mainly two points to be made. One, that any pre-amplifier should have adequate signal handling capacity in excess of the performance of any pickup cartridge both dynamically and in pure consideration of the amplitude of signals. Second, that as far as I am concerned the two pickup cartridges which are capable of giving peaks in excess of 200mV are the Ortofon SL15 with appropriate transformer and the Decca London cartridge.

The reference to signal peaks of 80 cm/s observed on gramophone records came from the book "Hi-Fi Systems" by G. King where there is a graph illustrating the velocities measured on gramophone records at various frequencies.

I nominate my favourite charity as the Musicians Union!

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