Audio preamplifier with no t.i.d.

Passive equalization eliminates transient intermodulation

by Yuri Miloslavskij, Institute of Constructional Physics, Moscow

In valve circuits, device linearity received much more attention than it does with transistors. Now, everything relies on feedback. Or one decreases steepness of the output characteristic of bipolar transistors with the help of an additional transistor.

But you may use a single transistor with better linearity. The problem of optimizing the number of stages is associated with having sufficiently linear transistors of optimal frequency properties, gain, noise and other characteristics. Designers of semiconductor devices should pay serious attention to designing such components. Their use would give even better results than the circuit described provides.

THE GREAT MAJORITY of published designs for electromagnetic pickup preamplifiers realized with valves or semiconductor devices have a feedback correction circuit according to RIAA covering one amplification stage but more often the entire preamplifier. In other, rare, cases preamps have a common negative frequency-independent feedback in individual units with the correction in a passive circuit. By such a circuit design superior results have been often achieved as to the harmonic and intermodulation distortion, though such attainment will never help to improve the sound to its original state as may be aimed and because some factors do not lead to any improvement at all.

As to the playback frequency response it is clear that this, along with non-linearity increasing with increase in frequency, contributes to the emphasizing of difference tone distortion, and the greater the frequency shift between the basic components and their associated difference tones the stronger is such emphasis.

So-called transient intermodulation distortion has not been measured as a result of novelty and complexity of this measurement, the lack of a unified and standard measurement method[1] and the lack of quantitative information on perceptibility; in some cases it has been completely ignored.

Listening tests also reveal specific distortions in preamplifiers, particularly in i.c. types. This was the reason for getting back to valve circuits. Temporal problems have been involved also by some other trends in the amplifier design (among other things as a result of the quest for the realisation of the entire preamp as a d.c. amplifier).

But the return to valve circuits is hardly justified, even though it leads to a decrease in transient intermodulation. As to semiconductor amplifiers the situation for most of their technical parameters is more favourable, i.e. frequency properties, noise characteristics, and power consumption. The common tendency is questing for non-linear distortion of the order of 0.001% or even less (this aim is difficult to attain by valve circuits and has not really been attempted). It is known that under contemporary conditions such a distortion level can be achieved only by the one method. But will it lead to the desired results?

To try and answer this question the following must be taken into account:

- Often a recording level relating to particular frequencies at velocity 14 cm/s is taken for the rated recording level—at l.f. the recording level is limited to a certain amplitude of groove displacement. (It would be useful to plot a frequency response of the maximum velocity for example on the basis of the data given in reference 2; this relationship is also necessary to provide a logical method for distortion measurement.) It's known that during recording some signals exceed this rated level. But the probability of appearance of a signal having, for example, a level 9 × 3.54 cm/s is not likely to be high (though this statement is not strictly exact because some firms make records under a high level condition). In addition, only few pickups can reproduce such signals.[3]
- Distortions introduced by recording process and disc production technology (harmonic distortion, difference and additive tones, intermodulation distortion) at velocity 14 cm/s are less than about 1% (at velocity 25cm/s no more than 4% (!) harmonic distortion at 1kHz is guaranteed on B&K test-record QR2010).
- Distortion introduced by the reproduction process because of certain factors (tracking distortions, pinch-effect, angle distortion, refs 4 to 7) at corresponding recording velocities exceeds distortion introduced by recording process, and at some frequencies may be as great as 10% and even more for harmonic distortion, 3 to 4% and more for intermodulation distortion[3,8]. Additive and difference tones distortion is of the same order as harmonic distortion.
- The hearing square and cubic non-linearity ($P_2 = \alpha_1 P_1 + \alpha_2 P^2_1 + \alpha_3 P_1{}^3$) shows itself as the subjective presence mainly of difference tones, as a rule exceeding hearing thresholds under masking. Masking effect consists in masking of some tones including distortion products by louder tones under certain conditions. For example the white noise in the bandwidth 20Hz to 20kHz masks the tone if the sound pressure level exceeds the spectral density noise level by 17 to 30dB. (Refs 9.)
- Studies performed in the USSR during late 1950's and 1960's revealed that more than 50% of sound producers and conservatoire students could not notice 2% square or cubic distortion in organ and violin solo performances.[10] It was also revealed by these studies that cubic distortion is more noticeable than square distortion as one would expect.
- Distortion introduced by tape recorders during recording, by the final mixing of sound tracks, by loudspeakers and other permanent factors act as an essential addition to distortion introduced during reproduction.

Based upon the above-listed considerations one can obviously conclude that in numerous cases sound coloration by preamplifiers and power amplifiers is caused by some special, and if you like pathological, non-linear distortion not properly pertaining to the electro-acoustic transducers and to the ear.

Transient intermodulation distortion as well as distortion severely increasing at low levels and large non-linearity of high order may be attributed to the above-mentioned special non-linear distortion.

Large high-order non-linearity results in large amplitudes of high harmonics, and of difference and additive tones of high order (an increase of side components of intermodulation spectrum). This means lack of steepness in non-linearity attenuation, i.e. insufficiently fast approach of coefficients α_4 to α_n to zero in

$$V_{out} = kV_{in} + \alpha_2 V^2_{in} + \alpha_3 V^3_{in} + \ldots + \alpha_n V^n \ldots \alpha_2 \text{ to } \alpha_n \to 0,$$

$$V_{in} = b_1 \sin\omega_1 t + b_2 \sin\omega t \ldots$$

$$\alpha_k = \alpha_k(b,\omega).$$

Therefore the problem consists of finding a more optimal (logical) way of obtaining specified parameters (quantitative values) $\alpha \ldots \alpha_n$, t.i.m. distortion, signal-to-noise ratio, f_{upper}, and by as simple a method as possible.

In designing the preamplifier it was proposed that distortion introduced by disc reproduction would not practically increase correspondingly with frequency and velocity and that distortion not properly pertaining to the electroacoustical transducers would not be introduced. It was expected that signal-to-noise ratio with pickup connected would be about 69dB(A) referred to an input level of 2mV (3.54cm/s) at 1kHz (ref. 11) and no less than 50 to 55dB in linear band of 20Hz to 20kHz (though it is not necessary to aim at such a high figure as 69dB at all because of worse signal-to-noise ratio of discs).

Based on the above-mentioned considerations I propose a circuit design for a preamplifier for electromagnetic pickups with bipolar transistors which produces practically no transient intermodulation distortion (below).

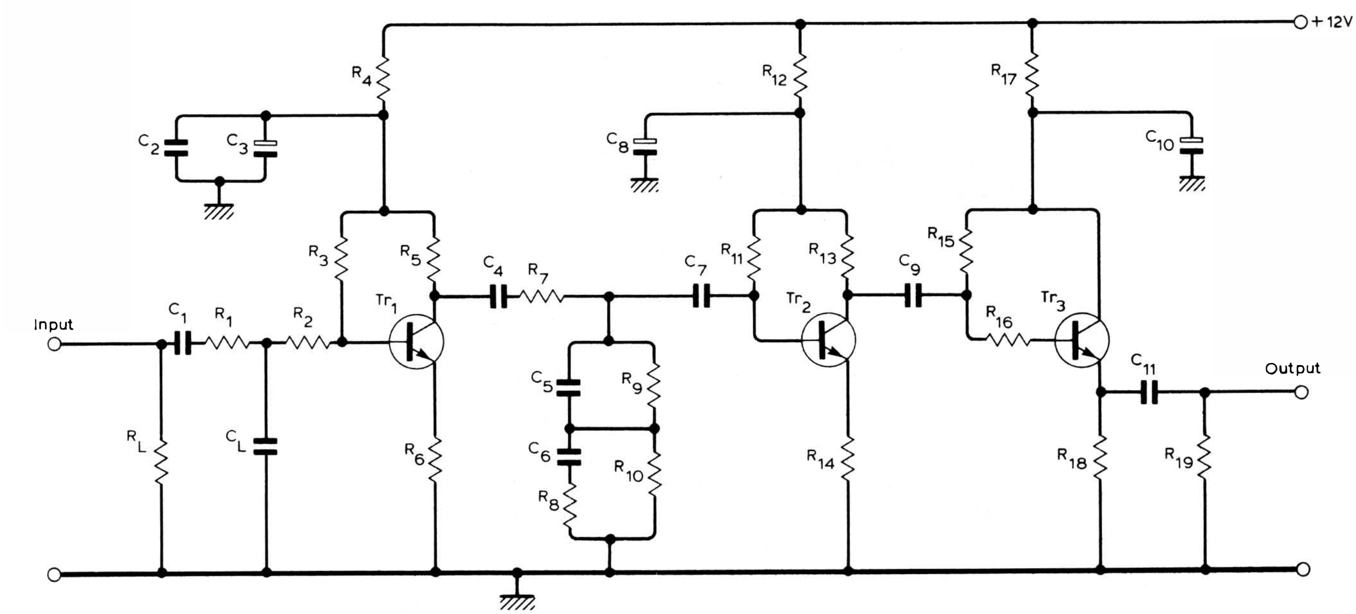
There is no common negative feedback; correction according to RIAA is accomplished in a passive circuit. The design is comparatively simple but it calls for high-quality components. The number of components and amplifier stages is practically minimal. The preamplifier provides a gain of 34dB at frequency 1kHz which more than twice exceeds the allowance required on the basis of data given in reference 2.

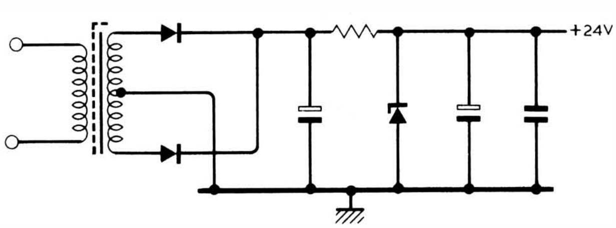
This circuit design is based on a close analysis, leading to the following main conclusions:

1. Bipolar low-noise transistors are now available which provide rather good amplification linearity at collector currents beginning from 100 to 150μA, at the same time as maintaining the required frequency properties at these currents.
2. From the distortion viewpoint the optimal place for a passive correction circuit is between the first and second amplification stages.
3. Signal attenuation by a passive correction circuit does not cause trouble, provided the operating conditions of the subsequent (and the first) stage are optimally selected and a low-noise transistor used.
4. Analysis of circuitry for a preamplifier having one voltage amplification stage with either passive or negative feedback RIAA correction) leads to unattainable technical requirements, especially for moving-coil cartridges.

Circuit description

The first amplification stage provides 75μs-time constant referred to 2mV per 3.54cm/s and its input resistance is decreased by resistor R_L to the required standard value (30,47 or 100kΩ). The required collector current in the first amplification stage is set by R_3. A bias circuit with fixed base current (rarely used) was selected mainly on the basis that the input resistance of the first amplification stage should be at least 100kΩ, and that contemporary low-noise low-power silicon transistors have collector reverse current of less than 1nA (V_c = 0 to 5V, T = 20°C). At temperatures of 25 ±20°C instability of collector current is less than ±7% (with





	2mV/3·54 cm/s	200μV/3·54cm/s	110μV/3·54cm/s
C_L	50-250p		220n–10n
R_L	see text	see text	100
C_1	2μ2	22μ	22μ
R_1, R_2	100		7R5
R_6	1k	100	27
R_5	75k	33k	33k
R_4	24k	6k8	6k8
R_{14}	750	330	330
R_{13}	43k	33k	33k
R_{12}	33k	6k8	6k8
R_3	→IC_1 = 200μA	→IC_1 = 500μA	→IC_1 = 500μA
R_{11}	→IC_2 = 200μA	→IC_2 = 500μA	→IC_2 = 500μA

Michael Sagin plans to provide printed circuit boards for a stereo version for £3 inclusive of V.A.T. and postage. His address is 23 Keyes Road, London, N.W.2.

Tr_1	BCY 55	C_5	220n	R_7	25k
Tr_2	BCY 55	C_6	47n	R_9	33k
C_2	100n	C_7	1·0μ	R_8	160
C_3	300μ,25V	C_8	300μ,25V	R_{10}	1k4
C_4	1·0μ				

a sufficiently high value of R_3) and may be easily tolerated. Analysis reveals that this instability depends mainly on temperature drift of transistor current gain, and also on voltage source stability, resistance value of isolation filter resistor (for voltage supply), and resistor stability.

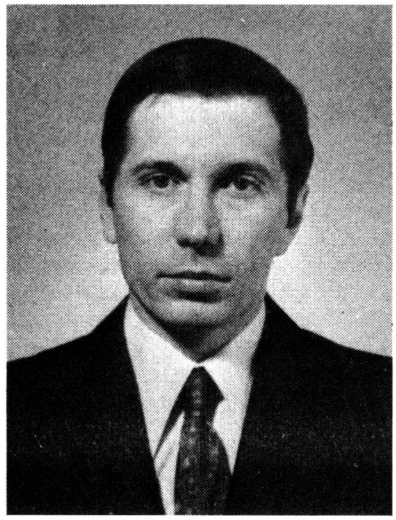
The value of collector current in the first stage (200μA) meets in a compromising way the following contradictory requirements: necessity for utmost minimized noise level on the one hand and obtaining the required distortion level, satisfactory frequency response and a level allowance with regard to peaks on the other.

The value of the local negative feedback resistor is selected on the basis of the same compromise. It should be noted that more stringent requirements are placed upon the linearity of the first stage of voltage amplification and upon its independence on frequency, and this approach helps to decrease the level of difference tones derived even from very high frequencies. Harmonic distortion (referred to the second harmonic) should not exceed 0.03 to 0.06% (V = 14cm/s); the upper frequency limit should be as high as possible.

The subsequent amplification stage is realized in the same manner. The collector current value in this stage (260μA) was selected on a similar basis. The next stage is the emitter follower.

The preamplifier is constructed with the help of the following components. Transistors Tr_1 and Tr_2 are the differential assembly BCY55, which provides a low level of infrasonic and low frequency noises, sufficient linearity, high gain in the operating current range, around 200, and has the required frequency properties. For isolation polycarbonate capacitors are selected (several isolating capacitors provide the required steep sharpness for attenuating signals containing infra-low frequencies). Resistors are of the high stability, 1% tolerance, low-noise class. Capacitors in the correction circuit are of polycarbonate type, 2% tolerance. Capacitor C_L is of mica or polycarbonate type; it is selected on the basis of pickup specification and input cable capacitance. Transistor Tr_3 is a low-noise linear silicon transistor, current gain is about 1000.

The preamplifier is supplied by battery or by a stabilized voltage source in a separate casing. The principle circuit diagram of this supply unit is shown on Fig. 1. Stabilized output voltage is 24V, hum level no more than 3mV. The application of this power supply circuit provides a safety margin for the required isolation from conducted interference of various frequencies and hum. This power unit and its isolation filters, does not introduce additional noise and interference at the preamp. output. Electrolytic capacitors of power unit and isolating filters are of solid tantalum type. Other capacitors are of low-inductance film type.



Thirty-two years old Yuri Miloslavkij is presently doing post-graduate work in the laboratory of architectural acoustics within the Insitute of Constructional Physics, Moscow. He first started working in the field of acoustics and electro-acoustics at the Institute of Television and Broadcasting, which he joined five years ago. Yuri Miloslavskij graduated in physics from Saratov University in 1971

Recently moving-coil cartridges find expanding application even in non-professional reproduction, resulting from the obvious merits of that type. With such a circuit design the preamplifier may be directly connected to a moving-coil cartridge merely by changing component values, without giving rise to a deterioration in specification; in this case the preamp. provides a gain increase of 20dB (26dB). Collector current in the first and the second stages is increased to 500μA. The former power unit is used merely by changing the values of resistors in isolation filters. It thus becomes unnecessary to use a special pre-amplifier. The preamp. may be used even for cartridges such as the Ortofon SL20Q and MC20, providing a very low output of 110μV per 3.54cm/s.

Similar modifications enable this preamp. to be fed by tape-recorder head or dynamic microphone signals.

Measurement data at input levels of about 2mV

1. Gain at 1kHz frequency: 34dB (amplification instability in the above-mentioned temperature range is no more than 0.5dB).
2. Signal-to-noise ratio referred to input level 2mV at 1kHz with a magnetic pickup connected: 64dB(A); in linear band 22.4Hz to 22.4kHz: 58dB.
3. Harmonic distortion at input level 8mV (14cm/s) at 1kHz: 0.15%. (In constant output voltage mode harmonic distortion doesn't increase with increase in frequency. Analysis of the level of high harmonics $n = 4$ to 20 with $f_{n=1}$ = 1kHz has revealed that their amplitudes are well below the third and especially the second harmonic; this demonstrates more than necessary attenuation of non-linearity.)
4. Intermodulation distortion at input levels 1mV/60Hz and 8mV(16mV)/7kHz: 0.25%.*
5. Difference tone 1kHz with input level of basic frequencies 10kHz and 11kHz being 8mV and at an amplitude ratio 1/2: 0.3%. Additive tone distortion doesn't exceed harmonic distortion.
6. Distortion increases proportionately to input level with a proportionality factor of about one.
7. Frequency response accords to the RIAA standard in the range of 100Hz to 10kHz with ±0.5dB accuracy (30Hz + 1dB, 20kHz + 3dB).
8. With the gain at 1kHz being equal to 40dB (the second stage gain increase is 6dB) s/n ratio referred to 2mV/kHz is 67dB(A). In this case the harmonic and intermodulation distortion increase for times (compared with 3 & 4); the difference tone distortion remains of the same order as in 5.
9. The upper 3dB frequency limit of amplification exceeds 125kHz (internal resistance of the sinsoidal oscillator was made 15kΩ, the approximate equivalent of a pickup and C_L at h.f., the correction circuit under RIAA stipulation changed to 240-ohm resistor; and capacitor C_L eliminated).

Measurement data at input levels of about 200μV

1. Gain at 1kHz frequency: 54dB.
2. Signal-to-noise ratio referred to input level 200μV at 1kHz with an equivalent cartridge resistance 5 ohm: 61dB(A) and in linear band 22.4Hz to 22.4kHz: 54dB.
3. Harmonic distortion at input level 800μV at 1kHz frequency: 0.15%.
4. Intermodulation distortion at input levels 100μV/60Hz and 800μV/7kHz: 0.25%.
5. Difference tone with level of basic frequencies 10kHz and 11kHz being 800μV and at amplitude ratio 1/2: 0.2%.
6. Signal-to-noise ratio referred to input level 110μV at 1kHz at preamp. gain 60dB (first stage gain nearly doubled): 59dB(A), and referred to input level 9 × 110μV: 78dB(A). Difference tone doubled (compared with 5), though harmonic and intermodulation distortion remain as in 3 & 4.
7. Other technical data are not worse than in preamp. version for input levels of 2mV per 3.54cm/s.

*maximum recording velocity at 60Hz around 2cm/s.

Audio preamplifier

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Measurements were conducted with the help of Brüel and Kjaer equipment. Listening tests confirm that the circuits do not introduce sound coloration.

Modifications

Some increase in signal-to-noise ratio is available by the following alterations:

- gain increase in the first and second stages mainly at the expense of decrease of emitter resistor values
- use of transistors giving even lower noise
- collector current decrease in the first stage (this leads to some increase of s/n ratio over the band)

More close conformity of the frequency response to the RIAA recommendation (approximately 1dB increase), can be realized merely by changing the correction circuit time constant.

The quest for an increase in s/n ratio should agree with known turntable rumble, disc noise and sometimes with other factors, for example tape-recorder noise, acoustic noises of rooms, etc.

Of course in doing so one must comply with linearity requirements, especially in the first stage. In the case of a different gain distribution among stage it is possible to decrease difference tone distortion to some extent at the cost of an increase in harmonic and intermodulation distortion and vice-versa. Harmonic and intermodulation distortion level is decreased by increasing the collector current in the second stage.

Difference tone distortion level is drastically decreased by increasing collector current in the first stage, which may demand some increase in power-supply voltage, giving only an insignificant decrease in s/n ratio.

When connecting a pickup giving higher output than 200μV or 2mV at 3.54cm/s set the output voltage level according to the required output level 100mV by increasing the values of emitter resistors, chiefly in the first stage.

The application of transistors having better linearity will give even better results as to distortion, but . . . at the present time as a rule designers take little interest in problems of the linearity of amplification elements.

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Additional reading

Hamm Russell O. Tubes versus transistors – is there an – audible difference?; *JAES*, 1972, vol. 21, no. 4.

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Wireless World binding — VAT change

As a result of the Government's recent increase of Value Added Tax we regret that the price of binding a volume of *Wireless World* has had to be increased. Instead of £6.65 the price is now £6.90 (which includes the index as before).

Apart from this the arrangements for the index and binding remain as noted in the May 1979 issue (p. 70). The index for Volume 84 (1978) is now available, price 50p including postage, from the General Sales Department, IPC Electrical-Electronic Press Ltd, Dorset House, Stamford Street, London SE1 9LU.

If you wish to use the binding service send your copies to Press Binders Ltd, 4-4a Iliffe Yard, Crampton Street, Walworth, London, SE17 with your name and address enclosed. Confirm your order to the General Sales Department (address above) and with this letter to Dorset House send a remittance of £6.90 for each volume. Please allow up to ten weeks for delivery.

In both cases cheques should be made payable to IPC Business Press Ltd

Passive notch filters

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Hence

$$a \cdot b = \frac{b}{1/a} = \frac{R_b + pL_b + 1/pC_b}{(1/R) + pC_a + 1/pL_a} \qquad \text{B.3}$$

Assume the relation

$$\boxed{R^2 = R_a \cdot R_b = \frac{pL_a}{pC_b} = \frac{pL_b}{pC_a}} \qquad \text{B.4}$$

then $R_b = R^2/R_a$, $pL_b = R^2 \cdot pC_a$, and $1/pC_b = R^2/pL_a$.

Replace the numerator of. equation B.3 with the above identities

$$a.b = \frac{b}{1/a}$$

$$= \frac{(R^2/R_a) + R^2 \cdot pC_a + R^2/pL_a}{(1/R_a) + pC_a + (1/pL_a)} = R^2$$

Thus, arm 'a' is the inverse of arm 'b' in the lattice of Fig. 1.1, i.e., the product $a \cdot b$ equals R^2. the input impedance of the lattice, with components relation of equation B.4, equals R at all frequencies.

From Fig. B.1 and reference 5,

$$\begin{pmatrix} V_1 \\ I_1 \end{pmatrix} = \begin{pmatrix} \left(\frac{b+a}{b-a}\right) & \left(\frac{2ba}{b-a}\right) \\ \left(\frac{2}{b-a}\right) & \left(\frac{b+a}{b-a}\right) \end{pmatrix} \cdot \begin{pmatrix} V_2 \\ I_2 \end{pmatrix} \qquad \text{B.5}$$

Thus

$$V_1 = \frac{1}{b-a}[(b+a)V_2 + 2ba \cdot I_2]$$

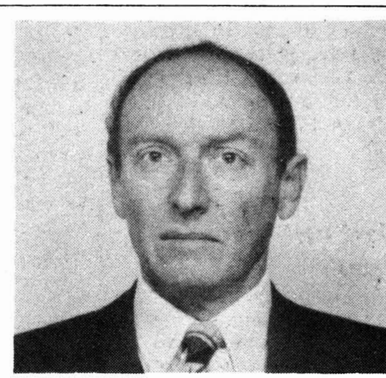
From Fig. B.1 $I_2 = V_2/R$

and from equation B.2 $a = R^2/b$

V_1 becomes

$$V_2\frac{b+R}{b-R} \text{ or } \frac{V_2}{V_1} = \frac{b-R}{b+R}.$$

To be continued



With an honours degree in electrical engineering from the University College of North Wales, Bangor, Gideon Kalanit joined Rediffusion in 1956, and was involved with the development of cable television systems, including experimental pay-tv systems. This work led to the interest in notch filters, the subject of this article.

In 1965 he joined Marconi Instruments to design equipment for television signal measurement, and rejoined Rediffusion Engineering in 1969. Presently with the advanced systems department, he is concerned with instrument design and the development of Dial-a-Program cable tv systems.

necessary to transmit a distress call on 500 kHz. Under these conditions it may not be possible to do so, or, if possible, it would be at much reduced power output. Should a vessel in these circumstances be any distance from another station it could result in the call going unheard on m.f. Perhaps this explains why vessels have disappeared in heavy weather without a distress call being heard.

Does not this raise the question is 500 kHz a suitable frequency for distress traffic working under these conditions?

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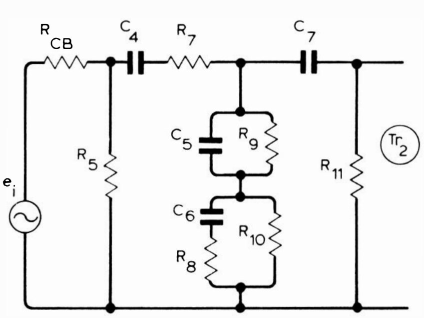
South Australia

PRE-AMPLIFIER WITH NO T.I.D.

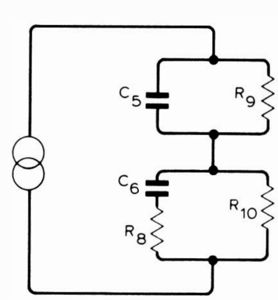
We all read very attentively the June 1979 *Journal of the Audio Engineering Society* in which Mr Lipshitz has given so many examples of the errors in commercial pre-amplifiers (even in "very expensive models") and his letter in your January 1980 issue is one more reminder. In 1978 we could not have known, unfortunately, about his article of 1979. Further, the specification of the equalization network will be considered according to the circumstances. Unfortunately, the question of the equalization network is not the main point of my article "Audio pre-amplifier with no t.i.d." in the August 1979 issue.

Firstly, the term "grossly in error" should be put in context. Let's take into account the fact that the pre-amplifier is always followed by volume and tone controls, filters, loudspeakers and a listening room. As far as is known, these units distort the signals to a greater extent (in amplitude and phase). By the way, in my August 1979 article I pointed out a discrepancy of the frequency response at the edges of the audio band, and I mentioned the possibility of modifying or completely replacing the equalization network. And for sure there is nothing in the article, using Stravinsky's words, that has "finally arrived". Taking all this into account it doesn't seem reasonable to complain of the RIAA network being "grossly in error".

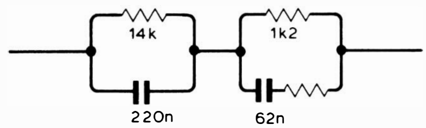
Postscript: Employing the classical equivalent network of the output circuit Tr_1 we have:



After the usual simplifications we have the equivalent network with a current generator:



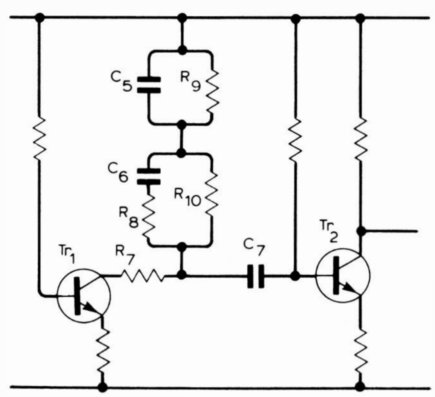
For a long time we have had the original and accurate method of calculation for such an equalization network; here, for example, is one of many possible versions:



It is clear from Lipshitz's letter and article that de-emphasis is passive, and in this case R_7, R_8, R_9, R_{10}, C_5, C_6 are the components "grossly in error". (It is just R_8, R_9, R_{10}, C_5, C_6 that are replaced in measurements by the 240-ohm resistor.)

If we take into account that R_5 of the following stage has some effect on the equalization network and there is a possible reduction of high frequencies by the input filter of the pre-amplifier (moving coil), as well as attentuation of low frequencies by all other following isolating capacitors (without putting on additional stages, etc.) we inevitably have to come to some compromise. And that has been achieved.

The circuit may also be used this way:



The resistor R_7 is used only for "equalizing" the loading of Tr_1.

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"TRIVIAL" AMPLIFIER DESIGNS

I was slightly perturbed by Mr B. J. Duncan's letter in the January 1980 issue. Since he has radically altered the design of my pre-amplifier by removing the discrete semiconductors and introducing i.c. circuitry, I hardly think it is fair to carry on referring to it as my design, and do not feel impelled to take any responsibility for its performance or lack thereof.

I do agree that an unacceptable aura of mysticism still seems to surround the performance of audio equipment. A great deal of nonsense is still being talked about "musical" capacitors, metal oxide resistors, and so on, although as far as I can tell the field is still wide open for the first brave man to stand up and explain how ears can register differences that not only escape the best test gear, but are also unknown to electrical science. Presumably, given time and a complete lack of supporting evidence, such silliness will once more become unfashionable.

However, I do differ with Mr Duncan in his assessment of the worth of increasing amplifier refinement based on actual engineering principles. If someone finds a way to reduce distortion in a given case from 0.005% to 0.004%, then surely the design approach involved is worth reporting, even if the current state of art in analogue magnetic recording renders such an improvement largely academic. Also, I suggest that there is much satisfaction to be gathered in constructing a piece of equipment that will degrade the signals passing through it as little as humanly possible, even if the signals available are of variable quality.

I see no reason why amplifier designers should shut up shop just because other parts of the audio chain have a lot of catching up to do.

Douglas Self
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A CRYSTAL BALL FOR CLEARER AUDIO

Once again we see Mr Douglas Self whingeing within these pages about how others see or change his "designs".

Mr B. J. Duncan's letter although prompted by Mr Self's February 1979 article "High performance preamplifier" refers to his "Advanced preamplifier" published in the November 1976 issue which, apart from the H. P. Walker RIAA stage, would appear to lend itSelf nicely to i.c. simplification.

I wonder if Mr Self or Mr Duncan have considered the recent Signetics SA5534 operational amplifier or the dual SA5532? I am sure that in future this part will become ubiquitous in audio designs as most of the real audio design has already been done by the i.c. manufacturer

Mr Self will no doubt use the 5532 frequently once he discovers how good it is, probably to the extent of paralleling many together to improve the already excellent noise figure or even to make an "Advanced" or could we expect a "High performance" power amplifier!

I suspect that the predicted "paperless office" of the future and the connection of "home computers" via telephone lines to a central computer "knowledge area" will put an end to the wonderful arguments in the letters pages of magazines like this one.

Perhaps Mr Self and Mr Duncan will then publish their future articles in the online "knowledge area" where they can better express their opinion without editorial control to a larger world wide audience.

Mr Self could use this "world wide arena" to freely comment about some other articles published within these pages that cannot be spared further column inches, like the August 1979 "Preamplifier with no t.i.d.".

Having built both the "High performance preamplifier" and the "Preamplifier with no t.i.d." for a friend, specifically as an RIAA stage, it is clear to the two of us and others that the "no t.i.d." design is far superior in its reproduction of music from a vinyl disc which, after all, is its intended purpose.

Overload, noise, distortion and other such metrics are required for commercial reasons to guide or even goad buyers into choosing one product over another but if excess amounts of negative feedback and design complexity are required to achieve these figures at the expense of less clear reproduction then "Less is more".

If Mr Self were to build the no t.i.d. preamplifier using the same components as his own, perhaps with a higher supply voltage to suit his overload requirement, he may well find, as we did, that it sounds very clear and natural and is truly much quieter compared to "High performance preamplifier" which I am sure will measure much better by the tradition-