Construction project:

New high performance stereo preamplifier - 1

As the next in our line of Pro Series of audio projects, we're presenting the Pro Series Two control preamplifier. It offers quite exceptional performance, an effective range of features and a simple construction technique.

by ROB EVANS

In the December '89 and January '90 issues of *Electronics Australia* we described the Pro Series One power amplifier; a very high performance MOSFET design capable of delivering more than 140W per channel. Not surprisingly, this has proven to be an extremely popular project, with many readers expressing more than satisfaction with the amp's performance and enquiring as to the availability of a matching preamplifier.

Well, as promised in the January issue, we can finally unveil the Playmaster Pro Series Two preamp – a unit designed to complement the quality of the Pro Series One power amp, and to provide the best possible driving signal for any other power amp with a rated input level of around 1 volt RMS.

We should point out that the extended development time for this project was certainly not due to any hiccups in the basic design, or a mid-stream change of direction on our part. Rather, the months were largely spent *refining* the preamp's circuitry and printed circuit board (PCB) layout, and performing exhaustive practical tests after each change.

That's not to say that the design is at all fussy – it's simply arranged to produce the maximum audio quality from readily available components. You can rest assured that the impressive results are quite repeatable – we've spent the time to make sure.

The final performance is indeed quite exceptional. As shown in the associated specs' table, the signal to noise ratio is better than –105dB under *all* conditions, while the channel separation is a healthy -90dB (at 1kHz) and the distortion rates at a low 0.0025%. This performance level rivals most contemporary CD players, and as was our aim, allows the preamp to behave as the proverbial 'straight piece of wire with gain'.

We should also state from the outset that the concept of this design was to produce a preamp with uncompromised listening performance, rather than a control centre with every conceivable feature. Quite frankly, the more domestic hifi environment is nicely catered for by integrated amplifiers such as our Playmaster 60/60 – which as it happens, is no slouch in the performance stakes.

The additional goals in the development involved the more practical aspects of reliability and ease of construction. Just as in the Pro Series One poweramp, this criteria has been satisfied by a straightforward construction technique with a minimum of interwiring, and a very simple mechanical layout.

By the way, we gratefully acknowledge the assistance provided by Jaycar Electronics, who supplied crucial parts for both the Pro Series Two preamp, and the Pro series One power amp.

Features

In a nutshell, the main features of our new preamp are its performance capabilities. In line with our design philosophy, the unit's audio quality has priority over any extra controls or facilities – as you may have noticed, the bells and whistles have been kept to a minimum. Nevertheless, as spartan as the preamp may appear, the features that *have* been included are really quite useful – and don't incur any performance penalties.

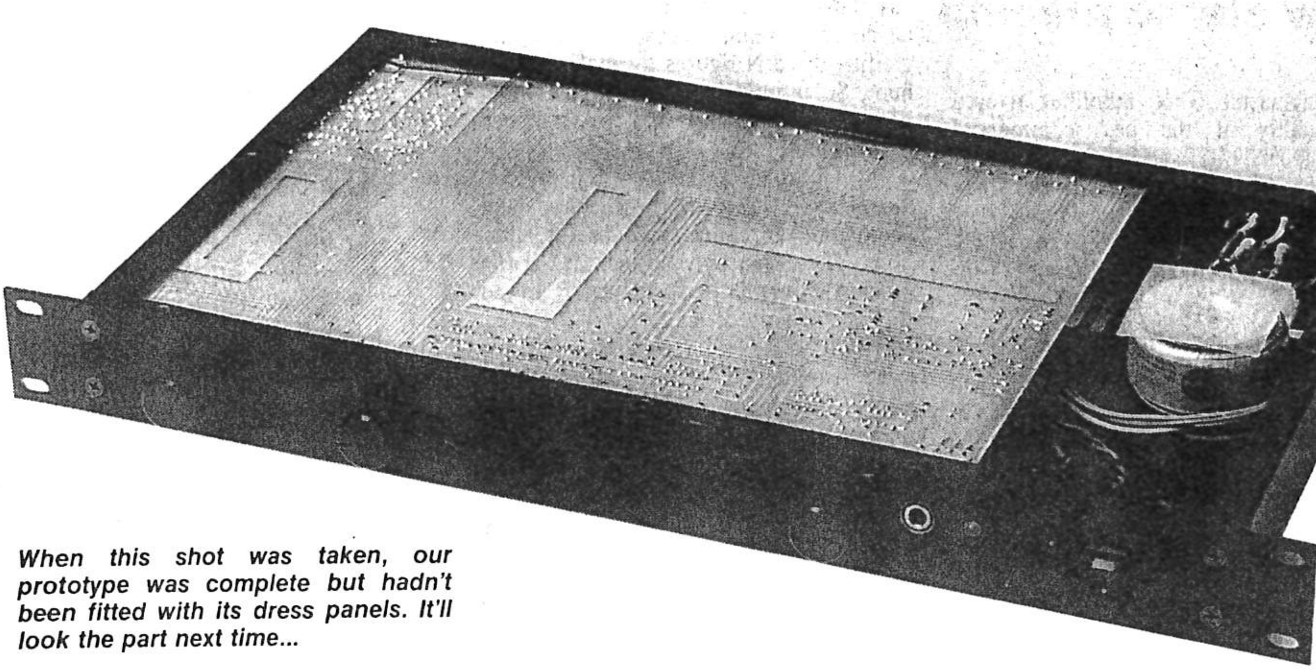
The facilities that have not been included on the other hand, were dropped because of both their inherent performance limitations and questionable usefulness. The most obvious of these are the familiar bass and treble controls.

Since a tone control stage works on active filter techniques employing phase cancellation, we can expect a number of significant anomalies to be introduced as the controls are moved from their centre position. In this condition, the attenuation of the feedback loop is modified in a specific band of frequencies, so as to provide a level of bass/treble boost or cut. We can therefore expect a certain level of increased noise and phase shifts from the circuit, as its frequency response curve is altered from the mean.

This in fact proved to be the case when a test circuit was put through its paces. As well as introducing the above errors, it contributed small levels of both static and dynamic distortion components to the output signal. While these problems were much less evident with the tone controls at their centre position, this is of course when the stage is providing no useful purpose.

A 'tone defeat' switch could bypass these controls when not required, but the fact remains that if such a stage was installed and used, the preamp's normally clean output signal would be considerably corrupted.

From a more practical viewpoint, it's likely that the tone controls would only be used to compensate for loudspeaker deficiencies or troublesome room acoustics – these days, the various signal sources are quite accurate, and don't require tonal corrections. But really, by far the best solution is to cure these



When this shot was taken, our prototype was complete but hadn't been fitted with its dress panels. It'll look the part next time...

problems at their source by using loudspeakers of sufficient quality to match the rest of the system, and experimenting with their positions within the room (we're not really asking you to rebuild your house)...

As a final point, standard tone controls are not very effective at compensating for the complex peaks and dips of loudspeaker/room problems anyway. For a positive effect, a much more sophisticated equaliser is required.

Other than the tone controls, you may have noticed that the usual balance control has also been 'given the bullet' (so to speak). A balance control is rarely used, except for the loudspeaker/-room inconsistencies given above, where the same basic solutions still apply. In the electronic sense, such a control will not affect the preamp's performance while it's in the centre position – providing a special type of potentiometer (M/N) is used. However, as soon as its action attenuates one channel, an appreciable resistance is placed in series with the signal path, and noise is introduced (as discussed later). So a balance control is fine, as long as you don't use it... Out it goes!

On a more positive note, we have elected to include a recording output selector switch (record), as well as the normal source selecting switch (listen). This allows any of the five signals sources to be channeled to the tape output connections, independently of the signal selected for listening. Normally these two functions are combined in one rotary switch, tying the audio output to the source that is being recorded. With a recording switch, you can listen to (say) a CD while taping a radio program – without repatching the signal leads.

We have also added a headphone outlet for private listening. This is driven by a dedicated high performance amplifier, so as to take advantage of the excellent listening quality offered by many of the currently available headphones. The actual signal quality at this outlet is high enough to rival that of the main preamp outputs.

The remaining sections of the front panel feature the volume control (its arrangement is indeed a feature, but more of this later), a mains power switch and two indicator LEDs. The 'on' LED indicates the presence of the internal DC supply, while the 'mute' LED displays the status of the output muting circuit.

Design considerations

To achieve the performance goals of our new preamp, we had to pay particular attention to the type of gain stages used, and less obviously, the manner in which the stages are interconnected.

The earthing technique for example can have a major effect on the channel separation, distortion, power supply interference rejection and general stability of the circuit. To attain the best possible performance in this regard, we elected to use three separate earth paths throughout the preamp.

These are a left signal earth, right signal earth and a general (or 'noisy') earth return. The PCB is arranged so that the left and right signal earths are fully independent until they join at the preamp's main output terminals – where they ultimately connect to a single earth point at the unit's power supply. This main earth point also connects to the general earth path, which (mainly) acts as the return for the various filter/bypassing capacitors scattered around the PCB.

In order to match the earthing requirements of the Pro Series One, and most other power amplifiers, the preamp is fully floating with only the metal case tied to the mains earth. This ensures maximum stability of the combined pair, and avoids the possibility of hum-inducing earth loops.

As a further guarantee of minimum interference between channels, we elected to use single op-amps (NE5534) for the gain stages, rather than the more convenient dual packaged devices such as the NE5532 and LM833. While these offer quite respectable figures for the channel separation between the internal op-amps, two individual devices will inevitably provide better performance in this respect.

Our search for a suitable phono stage took us through a number of designs, ranging from simple op-amp configurations to relatively complex transistor/op-amp hybrids. As it happens, we really didn't need to look any further than the back issues of *Electronics Australia*.

Over the years, a reasonably simple transistor/op-amp circuit has been used to great affect in most Playmaster amplifiers. This arrangement has been progressively developed as higher performance devices became available, to a point where the version appearing in the Playmaster 60/60 amplifier rivaled the quality of the best commercial phono stages.

So this circuit, with its proven track record, has been included in our new design to complement the extremely high performance of the line level inputs.

Practical considerations

Having satisfied the essential performance requirements, we need to ensure that the preamp provides our chosen features *without* any compromise to the audio quality – and of course, is straightforward to build.

In this regard, a great deal of effort has been taken with the design of the printed circuit board (PCB). As well as including the previously mentioned earthing arrangement, the PCB layout totally avoids the use of shielded hookup wire, which has proven to be both unreliable and difficult to terminate. All connectors and gain stages are directly linked via a main PCB, and one small supplementary PCB for the selector switches. Unlike some previous preamp designs, this allows for a very simple construction technique and less chance of performance anomalies.

We also feel that the most practical way to power a preamp is with its own internal mains transformer and associated circuitry. While an external power supply (be it a free standing unit or that of the main power amp) removes the possibility of a nearby transformer inducing hum in the preamp's more sensitive circuits, there are a couple of practical penalties.

Deriving the preamp's power from the poweramp for example means that it's not really an independent unit – it must always be used with a suitably modified poweramp, or some kind of external transformer. On the other hand, a preamp equipped with its own physically separate power supply shouldn't suffer from hum problems, and will be easy to mate to other poweramps. However, the additional expense of another box, connectors and cables makes this avenue difficult to justify – besides, it's messy.

The answer in our case was to follow the path of the Pro Series One poweramp, and use a high quality toroidal mains transformer. These transformers are very efficient (hence cool running), and have a tightly controlled radiation pattern when compared to the more conventional E-core units. This in turn has allowed us to install the transformer inside the preamp's case, without degrading the S/N figures through induced hum. So ultimately, we have the independence, simplicity and performance that's required.

As mentioned earlier, we felt that an independent recording selector and headphone amplifier were useful additions. These can only be included if the preamp's overall performance is in no way compromised.

Fortunately, a recording selector switch simply redirects the input source signals, and doesn't directly affect the preamp's audio path. Similarly, the headphone amp may be arranged as an independent unit (which only 'monitors' the preamp's main output), rather than included as an integral part of the main signal path.

The muting circuitry can also be included as somewhat of a feature in itself, although the reasons for this are not all that obvious. While most designs offer a muting circuit to disconnect the audio output for a couple of seconds during power-up, they disengage in a relatively slow fashion. We've taken steps to correct this shortcoming with a relatively sophisticated muting circuit.

To understand the usual muting problem, imagine that the preamp has been on for some time, and then the power is turned off and on in rapid succession – if you accidentally turn the unit off for example, the instinct is to quickly switch it on again. This invariably produces a loud thump from the speakers as the preamp circuits begin to shut down, and are quickly re-energised. The average typical muting circuit (due to its simple design) will not disconnect the audio output quickly enough to stop any transient signals being passed to the poweramp.

To ensure a failsafe muting cycle, the control circuit in our new preamp senses the AC supply voltage quite directly, and shuts off the audio output within a few milliseconds of the supply being removed. The actual muting is performed by the contacts of a standard relay rather than an active switching device (such as a transistor or FET), which can introduce subtle forms of distortion as their parameters vary with the applied signal voltage.

Noise and op-amps

During the development of almost any op-amp stage, its final signal to noise ratio can be calculated with quite reasonable accuracy. This is possible by use of a little basic circuit analysis, and the equation for thermal noise in resistors:

$$En = \sqrt{4kTR\triangle F}$$

where k = Boltmann's constant (1.38×10^{-23})

T = the temperature in Kelvins (usually 298)

R = the resistor value

$\triangle F$ = the noise bandwidth (usually 20kHz)

While this allows us to calculate the noise contributed by the resistors in the signal path, we must also consider the op-amp itself, which will of course generate a significant levels of internal noise. The op-amp's noise performance is usually expressed in terms of an equivalent input noise voltage (in nV/$\sqrt{Hz}$), and the equivalent input noise current (in pA/$\sqrt{Hz}$) – the $\sqrt{Hz}$ takes the bandwidth into account.

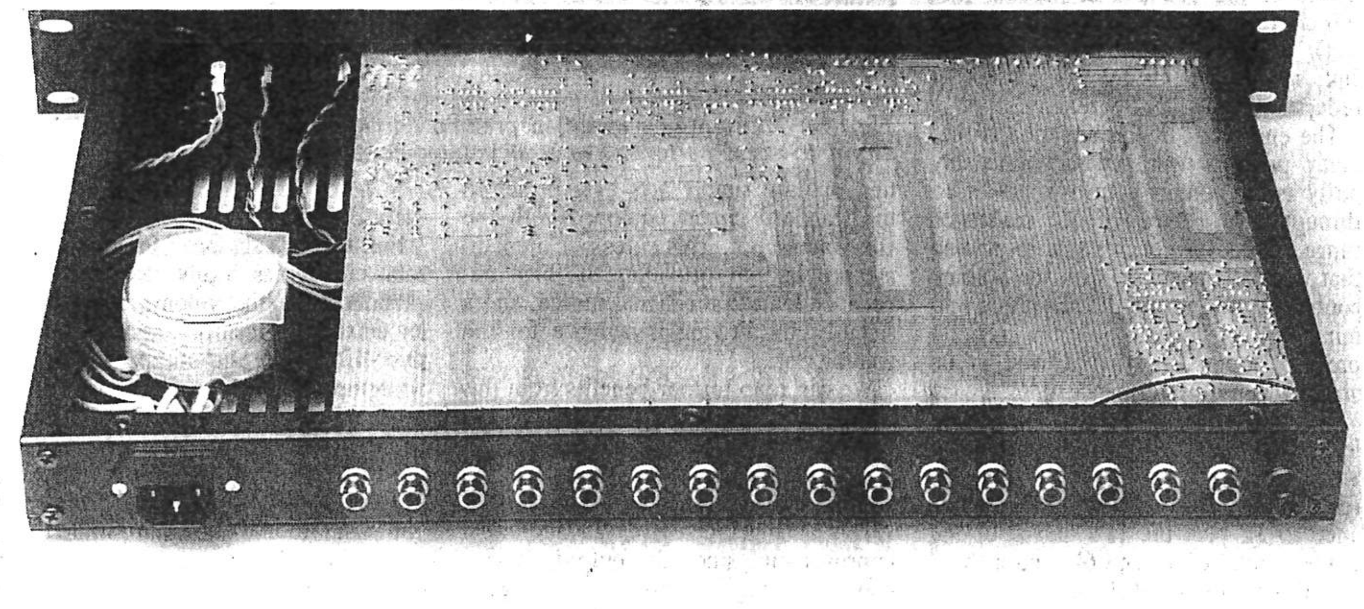
So to find the total noise level at the op-amp's input, we calculate the noise generated by any resistors in the signal path, and (vectorially) add this to the op-amp's overall equivalent input noise. The figure is then multiplied by the gain of the stage for a final estimation of the output noise. For a more detailed description, see 'Estimating noise in op-amp stages' which appeared in the April 1987 issue of *EA*.

In general then, we can achieve best overall noise performance by using an op-amp with a very low input noise figure, and the smallest possible resistors in the signal path (including the feedback network). However, there are of course a number of practical limitations when choosing these components.

The resistors that will directly affect the noise performance of a stage can be broadly divided into two groups – those which lie in the direct signal path (the op-amp's actual input resistor, and the internal resistance of the signal source), and the resistors in the feedback loop (used to set the op-amp's overall gain).

Unfortunately, the first group is often out of our control, since the input resistor will often define the input impedance of the stage and may need to be quite high. Also, the components which set the internal impedance of the signal source (in fact, its output impedance) are usually inaccessible. However most hifi signal sources (such as CD players) are typically less than 1k ohm, which is not too much of a problem.

The second group of resistors on the other hand, may be selected for the lowest possible value with only one major constraint. Since the feedback network offers a direct load to the out-



Another shot of our prototype before the dress panels were added. Note the very compact toroidal power transformer, and the lack of complex hand wiring.

put of the op-amp, the minimum total resistance of the network is limited by the drive capabilities of the device itself. Most op-amps will only deliver full performance into loads of at least 1k ohm – this of course includes both the feedback network *and* any load outside the actual stage.

So the options for minimising noise in an audio gain stage are really quite limited, due to the conditions around the circuit and the op-amp itself. However, a number of the above effects will be significantly reduced if we choose a device such as the NE5534AN.

This op-amp offers an equivalent input noise voltage of only $3.5nV/\sqrt{Hz}$, and the capability to deliver its full output swing into a 600 ohm load. And as a bonus, the NE5534AN has excellent supply rejection and distortion figures (typically less than 0.003% THD). However as mentioned above, its noise performance in particular will depend upon the actual circuit configuration.

Hidden noise

The main circuit used in most audio preamplifiers is really quite a simple arrangement. After all, the basic requirements are to boost the input signal level to that of the power amp's sensitivity, and to offer a relatively high input resistance and low output impedance. These conditions are quite easily satisfied by a simple non-inverting op-amp stage with a gain of 5 or 6, and an input terminating resistor of 47k. So for a working preamplifier, we simply replace this resistor with a volume pot (of a similar value) and include an input switching network to select the various signal sources.

This appears in a simplified form in Fig.1a, which represents the basic arrangement of most preamps. In this circuit, we've shown the input terminated (externally) in a 600 ohm resistor for the purposes of noise measurement, and the usual 'stopper' resistor (1k ohm) in series with the op-amp's non-inverting input. Assuming that the device is a 5534, the feedback resistors can be set to the lowest practical value for minimum noise (in this case, a total of 620 ohms) while maintaining a gain of 5.6. Finally, the 50k pot acts as a volume control and defines the circuit's input impedance.

Now as it stands, this arrangement is capable of very high performance. Thanks to the 5534, the signal to noise (S/N) ratio weighs-in at a theoretical –106dB (relative to a 1V RMS output level) with the volume control at its maximum position. On test, this figure was confirmed, and the distortion measured at around 0.002%.

With the volume control at its *minimum* position on the other hand, the S/N ratio is shown to be around –107dB. The slight difference between the figures for the minimum and maximum volume positions are due to the change of input resistance for each position. That is, when the pot's wiper is at the bottom of its travel the input resistance is set by the 1k resistor, whereas at the top position it becomes the 1k resistor plus the 600 ohm resistor in parallel with the 50k pot – a total of about 1.6k.

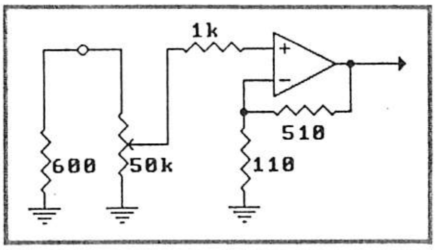


Fig.1(a): The circuit of most conventional hifi preamps can be reduced to this basic arrangement, for 'line' inputs.

All in all, the above noise figures are quite encouraging, and compare well to the specifications for most CD players. However, if we take the analysis one step further and calculate the S/N ratio with the wiper at its *middle* position, we arrive at a figure of only –94dB – some 12dB short of the figures for the extreme wiper positions. As can be seen from Fig.1a, the input path is now formed by the 1k resistor plus the two

New stereo preamp

halves of the 50k pot in parallel (the 600 ohm resistor is small enough to neglect). This produces a total of around 26k, which accounts for the substantial reduction in the S/N figure.

The end result of this situation is that many preamps will become noticeably noisy as the volume control is advanced through the centre of its resistance range. Like it or not, it's safe to assume that we listen to music with the volume control set away from the maximum and minimum positions! So this general preamp design is rather flawed in its *practical* noise performance.

The immediate answer to this problem might be to simply drop the pot value to say 10k, which would in turn reduce the middle position input resistance to 1k plus the two halves of the pot – a total of around 6k, and a theoretical S/N ratio of –101dB.

Unfortunately, the pot value also sets the preamp's input impedance in this design, which would consequently drop from the standard value of 50k to only 10k. But would this be a problem?

Since most signal sources have a reasonably low output impedance (say less than 2k), the above change should only drop the incoming signal level by less than 2dB – since we are dealing with a 2k/10k voltage divider. However due to the increased load, there is no guarantee that the source's low-frequency roll-off point is maintained at a satisfactory level.

The concern is as follows. Most signal sources (including CD players) use coupling capacitors in their output stage to ensure that there is no DC voltage present at the output terminals. This capacitor will form a high-pass filter in conjunction with the load resistance, whatever its value. Now if this load resistance is dropped by a factor of five (as in our case), the low-frequency roll-off point *must* increase by the same factor.

In practical terms, this means that if the source has a respectable –3dB point of say 15Hz for the standard load resistance (50k), then the reduced load (10k) will raise this point to 75Hz – not really an acceptable compromise. On the other hand, the output capacitor may be large enough to cope with the abnormal load, causing the cut-off point to rise from (say) 2Hz to 10Hz. Nevertheless, we don't feel it's worth the risk.

Fig.1b shows a circuit developed to correct the problem. In essence, the

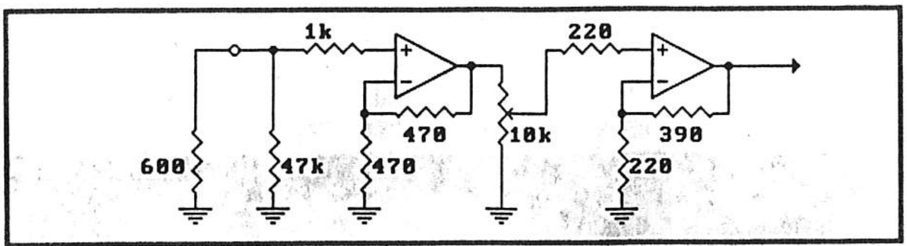


Fig.1(b): The author's modified preamp configuration, which achieves better signal-to-noise performance at all volume control settings.

first op-amp simply acts as a buffer to the standard circuit equipped with the 10k volume pot. This cures our immediate problem by providing the high impedance load to the signal source, and a suitably low driving impedance for the volume pot.

We can reap further benefits from this configuration by setting the gain of the first stage at 2, and the second stage at 2.77 – yielding an overall gain factor of around 5.6 (as before).

This arrangement gives an additional improvement, since the output stage is now operating at half of the original gain and will consequently amplify its total input noise by only 2.77, rather than 5.6. This in turn offers a significant benefit when the volume control is at its minimum position – that is, when the noise from the first stage is not passed to the second. In fact, the theoretical S/N ratio for this pot position is now –116dB.

Further calculations show that the total circuit has a S/N ratio of around –106dB when the volume control is at its maximum position, and an impressive –107dB for the middle pot setting. So compared to the noise figures of the original circuit (Fig.1a), we have a healthy 9dB improvement at the minimum volume setting, and a whopping 13dB reduction in noise at the middle pot position. At maximum volume the two circuits deliver the same noise performance, which is more than adequate.

Note however that since the volume pot has a logarithmic taper, the knob pointer will be positioned at about 3 o'clock for our half resistance value. This is in fact where the noise will peak in the standard circuit.

So if we assume that most listening is

SPECIFICATIONS

Signal to noise ratio (20kHz bandwidth, unweighted, ref: output level of 1V RMS)			
	Line level inputs:	Minimum volume	–115dB
		Mid volume	–106dB
		Maximum volume	–105dB
		Typically	–109dB (–112dB ref: 1.5V RMS)
	Phono input:	1kHz, 10mV input	–89dB (1k source)
cf34**Channel separation** (20kHz bandwidth, with respect to 1VRMS output level)		1kHz signal	–90dB (1k source)
		10kHz signal	–71dB (1k source)
Harmonic distortion (20kHz bandwidth, typical load)			
	Main output:	At 1VRMS	0.002%
		at 5VRMS	0.002%
	Headphone output:	Below clipping	0.002% (8 ohm load)
Input and output levels		Maximum output	8V RMS
		Rated output	1V RMS
		Line inputs	180mV (for 1V output)
		Phono input	3mV (1kHz, 1V output)
		Phono overload	150mV (1kHz)
Frequency response		Line inputs	20Hz to 20kHz –0.5dB; 10Hz to 80kHz –3dB
		Phono input	20Hz to 20KHz +/–0.5dB
Muting		Turn-on delay	approx 5 seconds
		Turn-off delay	approx 0.004 seconds

done with the volume control positioned between say minimum and 12 o'clock, it may be relevant to recalculate our figures for that range. The original circuit yields a noise figure of about –98dB for the control positioned at 30% of its range (12 o'clock), while the modified circuit returns a theoretical reading of –109dB – still a substantial 11dB improvement in the S/N ratio at this volume setting.

Well, after that barrage of noise figures you could be forgiven for thinking that this design was completed on theory alone – not so. At a very early stage in the development of this project the above figures were confirmed with both objective, and subjective testing. The only real problem that we encountered was that some of the very low noise figures embarrassed our test gear somewhat, and we had to resort to connecting the preamp channels in *series* to bring the resulting noise level above that of the instruments' own noise floor.

As an interesting aside, we briefly contemplated a 'straight through' mode for the preamp, where the signal source is routed directly to the output via the volume control – without passing through any active circuitry. While it seems that this system should provide the highest quality of all, it unfortunately falls foul of the dreaded volume control and its series resistance.

It follows that to maintain the correct input impedance (and load for the signal source) in this case, the combination of the volume pot and the power amp's input impedance has to be around 50k ohms. This in turn means that when the volume control is set to anything less than maximum, the power amp is operating with a relatively high resistance in series with its input.

Now, just as with a high quality op-amp, the power amp's performance will be severely compromised by the above conditions. In short, the basic preamp requirements of a high input and a low output impedance are not satisfied. Needless to say, we dropped *that* idea like a hot potato...

That's about it for the overall features and design. We can now take a look at the actual circuit, which is really quite simple – despite the numerous demands we've placed on its performance.

That's all we have space for in this issue. Next month we'll look at the final circuit and move on to the construction and trouble shooting of the pre-amp plus a few installation hints.

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Construction project:

New high performance stereo preamplifier - 2

Last month we discussed the performance goals and design of our new *Pro Series Two* preamplifier. In this issue, we present the actual circuit, construction details and troubleshooting hints.

by ROB EVANS

As you can see from the shots of the new preamp, its front panel has quite a spartan appearance and incorporates very few controls. These are the input selector switches, a volume control and a power switch – hardly a knob twiddler's delight. As discussed in the last instalment, this simple approach allows the circuit to deliver its full noise and distortion performance, without the corrupting influence of the usual tone and balance control stages.

All of the preamp's circuitry is contained on one large printed circuit board (PCB), with a small supplementary PCB acting as a connector for the two selector switches. This main board occupies most of the interior space of the standard one-unit rack mount case, while the remaining area holds the toroidal power transformer and its associated mains wiring. All in all, it's a very simple arrangement, which should make the construction process very straightforward.

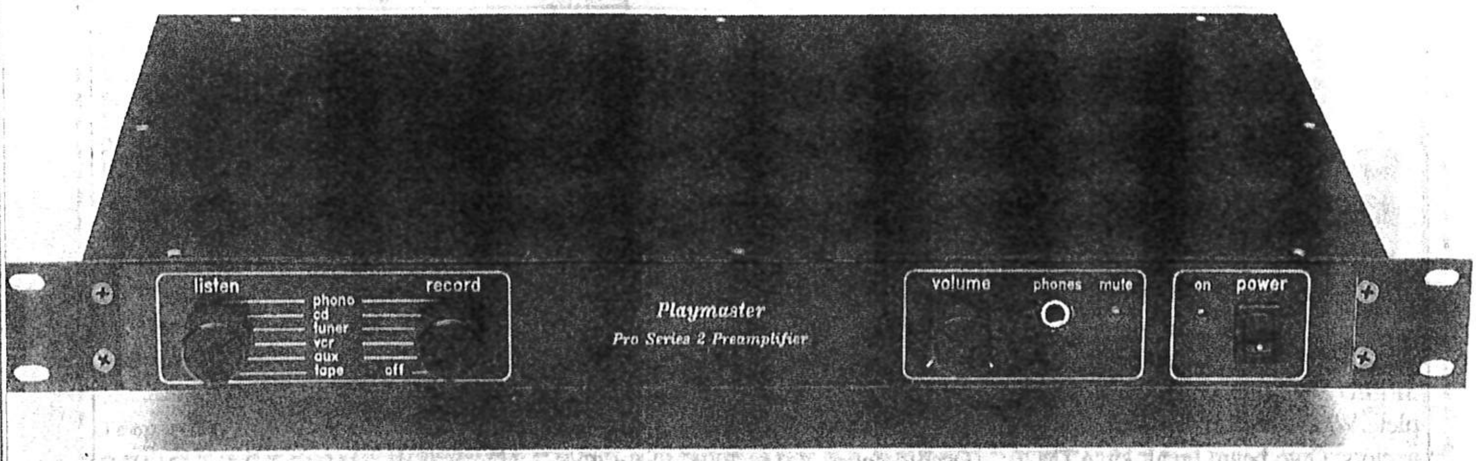
You might also notice that the actual printed circuit board looks quite complex from the copper side, with relatively small areas of unoccupied board. This is mainly because virtually all of the required connections are made by copper tracks on the PCB, rather than with hookup wire and shielded cable. In fact once the PCB is complete, you only need to wire up the input/output sockets and connect the power transformer to have a working preamp...

But an elaborate PCB design doesn't necessarily mean that the circuit itself is unduly complex. In the case of our new preamp, the action of most stages should be quite self-evident.

Circuit description

As shown in the overall circuit diagram, the basis of the preamp's circuitry is the amplifying stage formed around IC2 and IC3, which is a practical version of the low-noise arrangement shown in Fig.1b of last month's instalment. Additional components around this circuit include the coupling capacitors C13, C16, C17 and C20, and compensation capacitors C14, C15, C18 and C19; these define the preamp's final bandwidth at around 10Hz to 100kHz. The two stages are referenced to ground via resistors R17 (IC2) and R21 (IC3), while R16 and R26 terminate the input and output signals respectively.

The output of IC3 is both isolated from capacitive loads and protected from short circuits by R25, which ultimately drives the preamp's main output

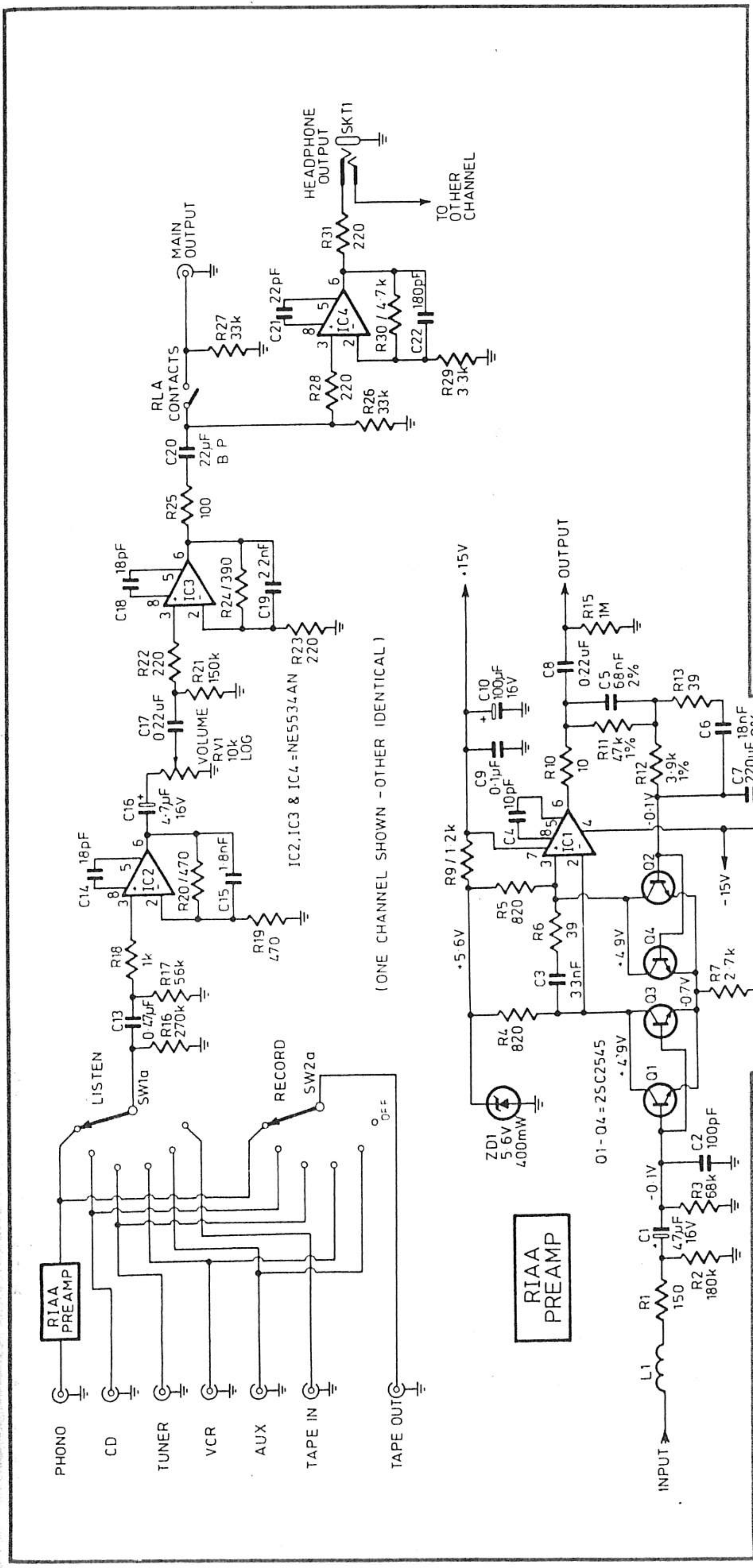


While it may look rather simple, the preamp offers exceptional audio performance.

socket via C20 and the muting relay contacts (RLA).

As well as providing the main output signal, IC3 also supplies signals for the headphone amplifier (IC4), via its input 'stopper' resistor R28. This stage is set to a gain of 2.4 by the feedback resistors R29 and R30, and stabilised by high frequency compensation capacitors C21 and C22. The op-amp's output drives the headphone socket via a 220-ohm isolating resistor (R31), which both protects the output against short circuits and compensates for the wide range of available headphone impedances (typically from 8-ohms to 600-ohms).

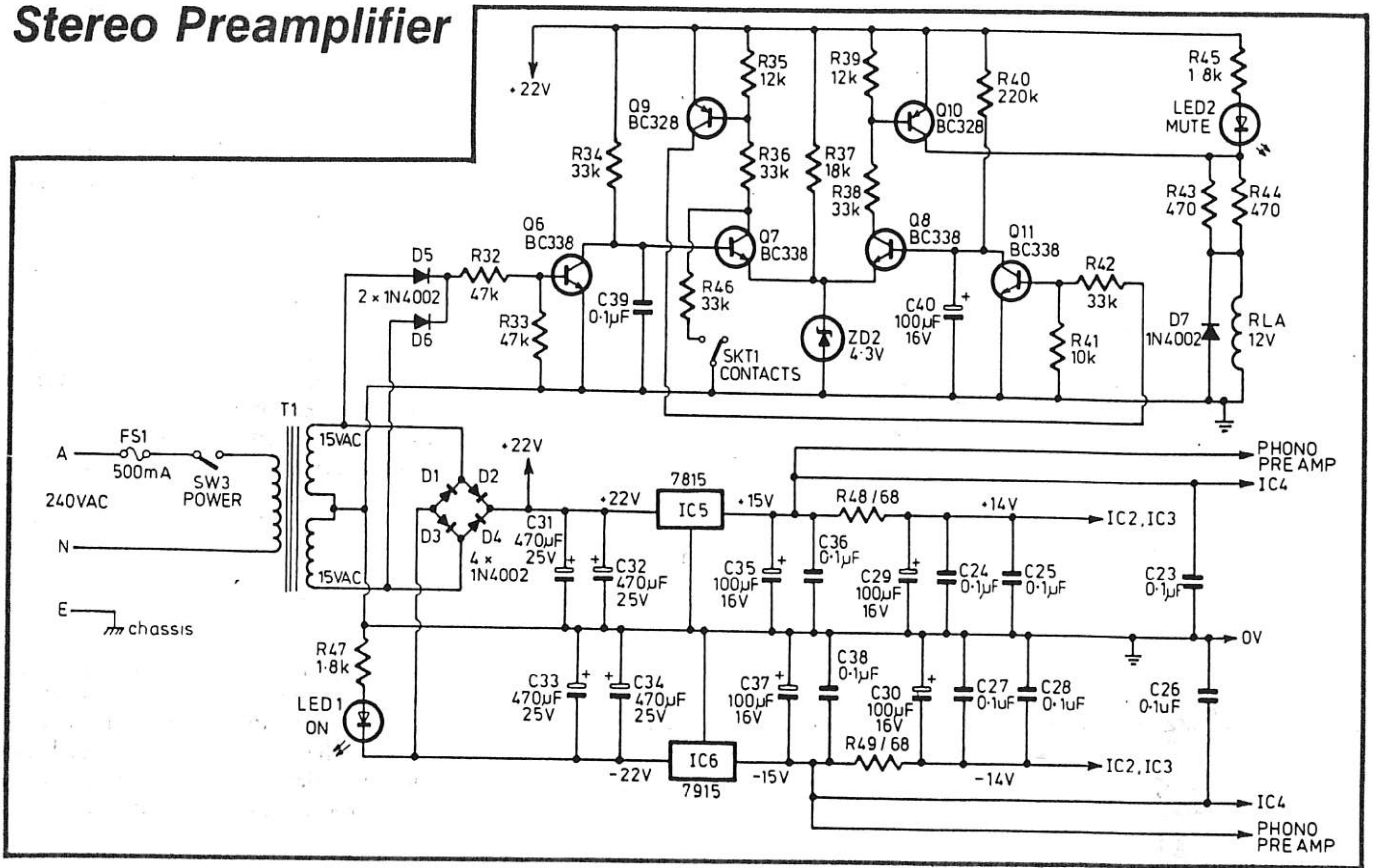
This very simple headphone amplifier circuit is possible due to the high output current capabilities of the NE5534, which as previously mentioned, can deliver its full output swing into a 600-ohm load. The load is less than this for 8-ohm phones, but in this case full output swing is not needed. The end result is a headphone signal with extremely low noise and distortion components – in fact, the specifications for the headphone outlet are virtually identical to those of the main preamp outputs.



The schematics of the signal stages: IC1 and IC2 form a very low-noise gain stage for all settings of the volume control, as discussed in last month's installment. The RIAA (phono) stage uses ultra-low-noise transistors in a paralleled differential pair configuration.

The input to the main preamp stage is controlled by the rotary 'listen' switch (SW1), which selects signals from one of the five high level inputs or the output of the phono preamp stage. The 'record' switch (SW2) on the other hand, simply routes the required signal source to the 'tape out' connector – with the exception of the 'tape in' signal, as this would constitute a positive feedback loop when the tape machine is placed in its recording mode.

The phono stage uses yet another NE5534AN op-amp (IC1), but in this case in conjunction with four ultra-low-noise 2SC2545 transistors (Q1 to Q4). The transistors are configured as a paralleled differential pair, with a standing current of around 1mA for each side, as set by the constant current source



The preamp's power supply: The top section of the schematic shows the output muting control circuit, which uses AC-sensing and two timing circuits to provide a reliable muting action.

formed by Q5 (a FET), R7 and R8. The two op-amp inputs (pins 2 and 3) are driven directly from the differential pair collector loads R4 and R5, with C3 and R6 providing a low impedance load at very high frequencies.

The negative feedback (NFB) path arranged between the op-amp's output (pin 6) and the base of Q2 (and Q4) sets the RIAA phono equalisation curve, and the overall stage gain (around 35dB at 1kHz). This uses close-tolerance components at the critical points, and includes R11 to R14 and C5 to C7. While the input differential 'pair' tends to define the noise performance of this stage, the NE5534 allows us to enhance this figure by using a very low impedance NFB loop – and of course, its own internally generated noise level is extremely low.

To reduce the likelihood of stray RF signals reaching the input circuitry, inductor L1 (around 10uH) and C2 form a suitable low-pass filter, while R1 acts as the usual 'stopper' resistor. The input is AC coupled via C1, and referenced to earth by R2 and R3, which also set the input impedance to around 50k ohms.

The remainder of the preamp's circuitry involves a straightforward power supply, which produces the regulated +/–15V rails, and the timing/detector circuit to drive the muting relay.

A toroidal transformer (T1) with two 15V windings is arranged as a 30V centre-tapped source for the bridge rectifier, D1 to D4. The resulting +/–22V DC unregulated rails are filtered by C31 to C34, and supply both the muting circuit and the three-terminal regulators IC5 and IC6. The resulting +/–15V rails are immediately filtered and bypassed by C35 to C38, with further filtering performed at various locations around the PCB – that is, at each group of op-amps.

The relay muting circuit is based around the charging times of two RC combinations – R34 and C39, and R40 and C40. By sensing the state of the unregulated supply, these set the relay's drop-out time and turn-on delay respectively.

The relay coil (RLA) is energised via the current limiting resistors R43 and R44, and the (normally on) transistor Q10, which is in turn controlled by the action of Q8. The emitters of both Q8 and Q7 are held at around 4.3V by the zener diode ZD2 and its associated resistor R37, which forces each transistor to act as a simple comparator – that is, they will begin to conduct when the base voltage exceeds about 5V (4.3V + 0.7V).

When power is first applied to the circuit, C40 will slowly charge towards the 22V supply via R40 (ignore Q11 for the moment), until Q8's base potential reaches 5V, where its increasing collector current will bias Q10 hard on. The relay is therefore energised after this preset time (around five seconds), after power is first applied.

The remaining circuitry is used to retrigger this delay cycle if the 240V mains supply is interrupted – of course, turning the unit off constitutes an extended interruption.

D5 and D6 provide a full-wave rectified (but unfiltered) version of the transformer secondary voltage to the base of Q6, via the limiting resistors R32 and R33. Therefore Q6 is biased on for voltages greater than about 1.4V at the cathodes of D5 and D6.

Since our (rectified) AC input has a peak level of around 22V, Q6 will be generally biased on, except for the period between each cycle where the waveform momentarily drops below the 1.4V level. So for a couple of milliseconds Q6 will be off, allowing C39 to charge quite rapidly via R34 (see Fig.2).

Now the circuit values for C39 and

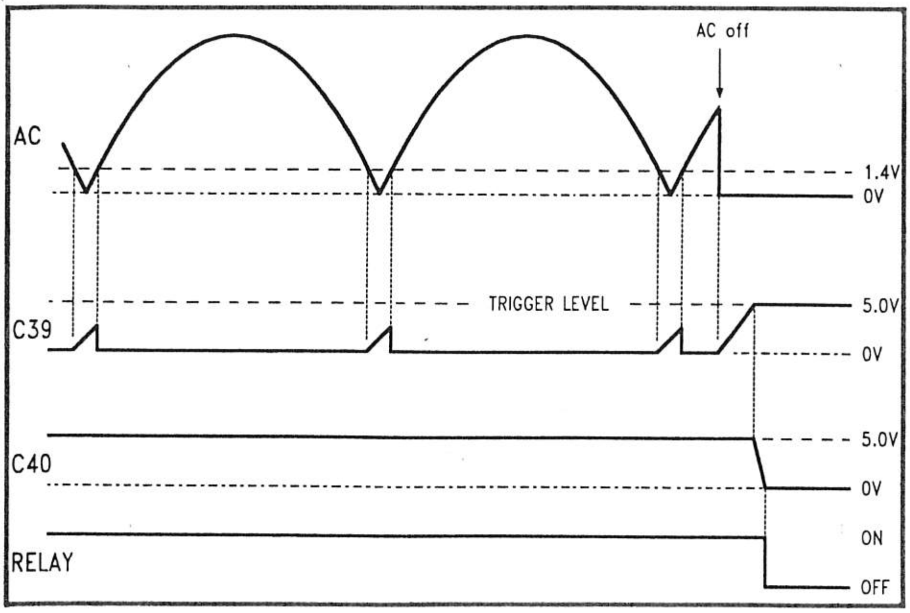


Fig.2: The operation of the AC-sensing section of the muting circuit. If the (rectified) AC input drops to 0V for more than a few milliseconds, the relay is de-energised.

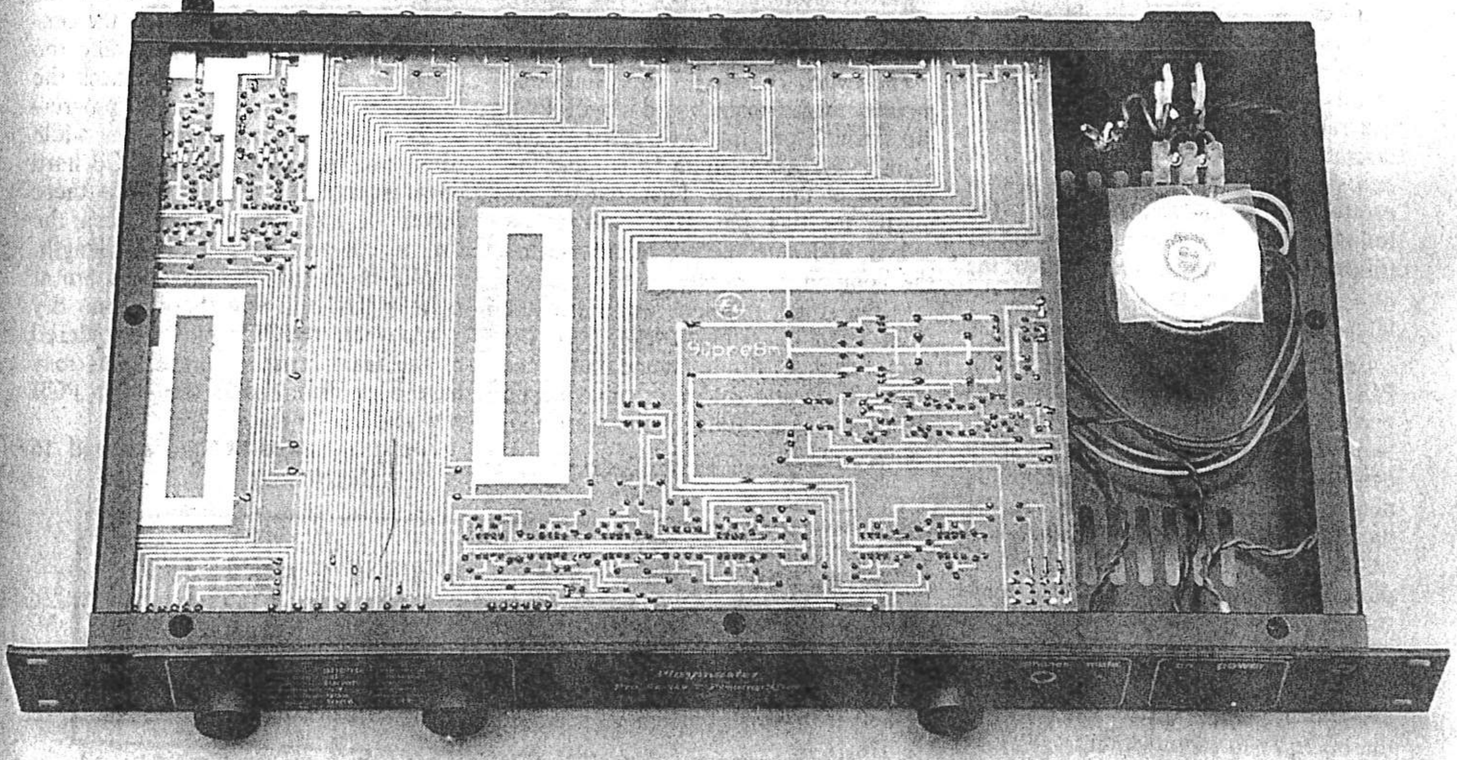
R34 have been chosen so that the capacitor will charge to a maximum level of around 2 volts, before it is discharged by Q6 at the end of the period. In fact the RC combination has a time constant of around 3ms. This means that in the normal course of events (a continuous AC supply), the voltage across the capacitor will not reach the trigger level at the base of Q7 – that is, the 5 volt level mentioned above.

On the other hand, if the AC supply is interrupted for more than about 4ms, C39 has enough time to charge to the trigger level of Q7, allowing it to conduct. This in turn forces Q9 and Q11 into their saturated states, via R36 and R42 respectively. The action of Q11 will quickly discharge the main timing capacitor, dropping the base of Q8 to 0V and ultimately de-energising the muting relay.

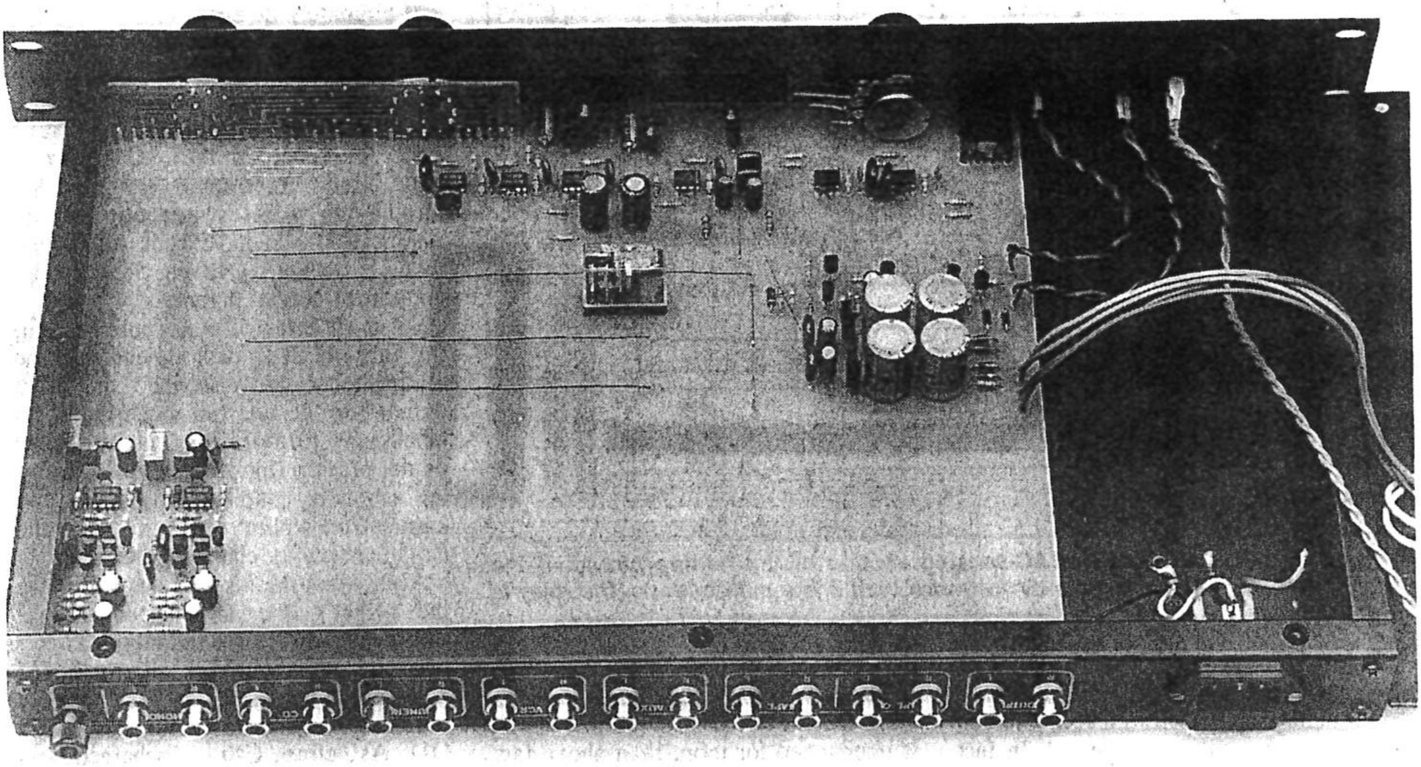
As the supply voltage falls, the circuit will not have enough power to re-energise the relay or complete the timing cycle. However, if the AC supply is only briefly interrupted (say, quickly turning the mains switch off then on again), C40 will have been discharged by Q11, forcing the timing circuit to operate in its normal manner.

So in all cases of mains interruption, the turn-on delay will de-energise the muting relay, and disconnect the audio output for the *full* timing period. This blocks any turn-on transients produced by the preamp (mostly due to the charging action of coupling capacitors), and ensures maximum protection for the amplifier/speaker chain.

Also, when the headphones are in use the socket's internal changeover contacts are used to force the muting circuit into its reset state. The collector of Q7 is pulled low via R16 and the socket's contacts, which in turn saturates Q9 and Q11. As before, C40 is discharged, and the relay is de-energised via Q8 and Q10. In this case however, the circuit will remain in this condition (with the output muted) until the headphones are disconnected.



The completed preamp with the top panel removed. Note the plastic insulator added to the toroidal mains transformer.



The component side of the PCB faces the unit's bottom panel. As you can see, it's quite a spacious layout.

Construction

While we've taken some pains to ensure that building the Pro Series Two preamp is a straightforward process with a minimum of wiring, the mechanical construction can be a little tricky if not completed in the right order. This is mainly due to the assembly method of the rack-mount case itself, which uses discrete nuts and bolts to hold the front and side panels together, rather than a captive nut arrangement. In short, the following assembly procedure should be followed quite closely, since some of the nuts are just not accessible during the later stages of construction.

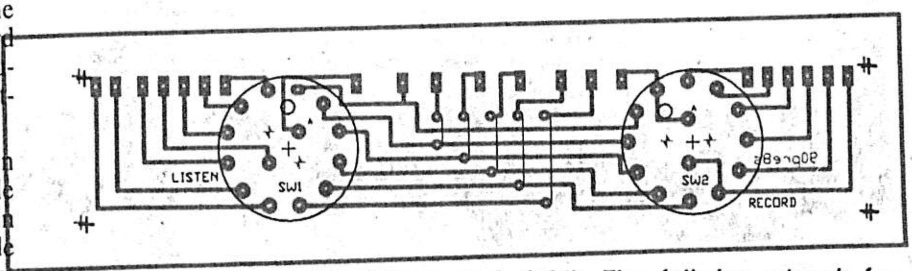
Begin by thoroughly checking both PCBs for any etching anomalies, such as bridged or open circuit tracks. Note that two corners of the main PCB have small cut-outs to clear the panel-mounting nuts at the 'listen' switch end of the unit. Also check that the main board fits neatly into the case, with just a couple of millimetres to spare in the front-to-rear panel dimension.

Mount the components on the main PCB, starting with the lowest profile components. There are 19 wire links in the board – the shorter links are made with tinned copper wire (or component leg offcuts), while the longer connections such as those between the power supply and phono stage should be formed with insulated wire. As usual, take particular care with the orientation of polarised components, such as the semiconductors and electrolytic capacitors – refer to the component overlay at all times.

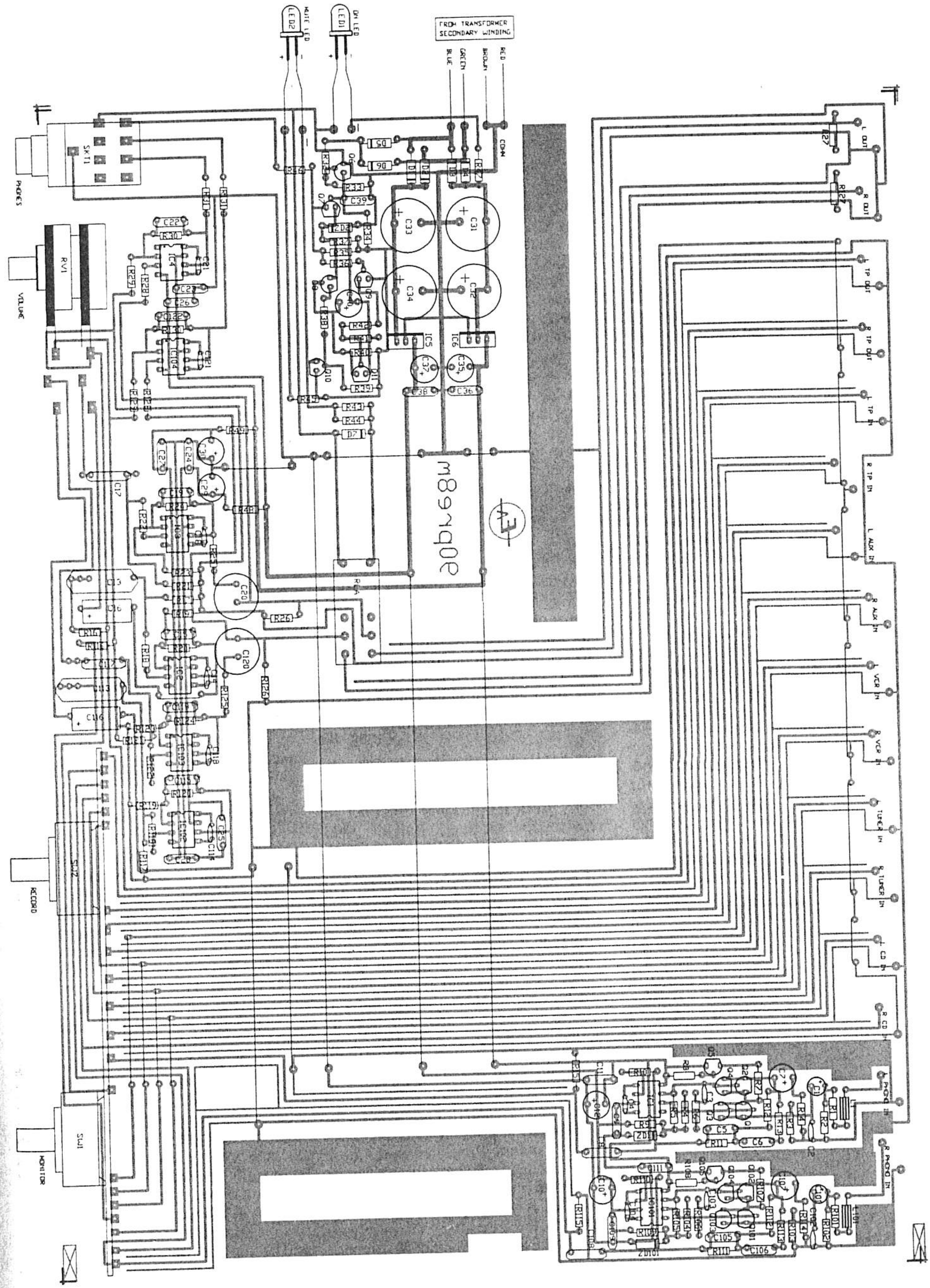
The switch PCB has 5 short links and mounts to the main board via 21 PCB pins, which are first installed into the main PCB. An accurate fit is quite important at this stage. First, cut off the plastic locating pin from each rotary switch, and note the pin orientation on the component overlay – this ensures that the switch positions will match the front panel markings. Next, push the two switches firmly home into the switch PCB and solder in place (again noting the orientation of the plastic locating pins).

To avoid difficulties during final assembly steps, the edge of the completed switch PCB must be mounted flush to the main PCB, over its *full* length. The best method here is to first attach the switch PCB by only a couple of the centre PCB-pins, while firmly holding the assembly accurately in place. Check the alignment, and once satisfied, progressively solder the remaining pins while continuing to hold the switch PCB hard down against the the main PCB – there should be almost no gap between the two boards over the full contact length. Finally check that the two PCBs are at right angles, and that there are no dry joints where the PCB pins are soldered to the main board – these joints may have been disturbed as the switch PCB was soldered in place.

The dual-ganged volume pot and 16



The component overlays (above and right): The full-size artwork for the preamp's front and rear panels, and both PCBs can be obtained from our office by sending a $10 service fee – it's too large to publish in actual size.


FROM TRANSFORMER
SECONDARY WINDING

RCA sockets are connected with lengths of tinned copper wire during the final assembly. For the moment, solder short lengths of this wire (about 30mm) to the appropriate pads on the main board, as indicated in the component overlay – a tedious job perhaps, but far simpler than terminating numerous lengths of shielded cable, as is often the case.

With the circuit boards now complete, it's worthwhile rechecking the accuracy of your work – in particular, check the orientation of the various polarised components. The next step is to assemble the case and install the PCB assembly.

As mentioned above, the chassis mounting nuts will be very difficult to reach unless this construction is completed in a particular order. Also note that the front, rear and side panel locating holes are oversize for their matching nuts and bolts, which allows a wide range of adjustment in the alignment between each panel. However the top and bottom panels are held in place by close-fitting countersunk screws, which mate to captive threaded lugs in the main chassis sections. Since the position of these panels is not adjustable, they will tend to determine the overall box alignment.

Start the final assembly stage by installing the IEC power connector (fuseholder at the top), phono ground binding post and the 16 RCA sockets in the rear panel. When looking at the inside of the panel with the power connector on the right-hand side, the RCA's ground tags should face towards the left. Tighten their lock nuts quite firmly, since the twisting action of inserting and removing RCA plugs can eventually loosen the sockets – don't overdo it though, or you may strip the thread or crush the insulator. Also make sure that the insulating washers are in their correct positions, so that there is no electrical connection between the socket and the case.

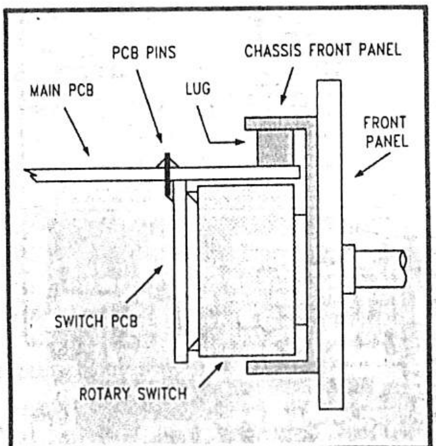


Fig.3: The completed PCB is a neat fit between the top panel mounting lug, and the lower edge of the chassis front panel.

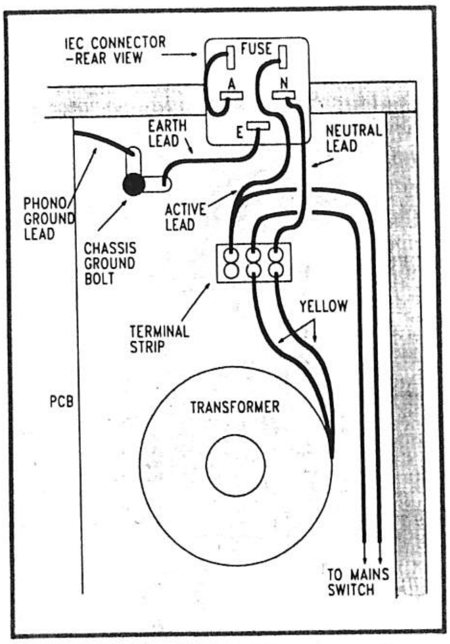
Next, bolt the two front panels to the left-hand chassis member (looking at the front panel), using the box lid or base as an alignment guide. Make sure that the various holes in the front dress panel and the matching chassis panel line up – temporarily install a nut and bolt in the other end of the front panels to hold this alignment.

The complete PCB assembly can now be gently worked into place. Note that the PCB and switch bodies are quite a close fit within the front chassis panel, with the board lying against the threaded lugs – see Fig.3. The rotary switches mount directly to the front chassis panel, without the need for spacing washers. However, don't forget to include the switch position selector ring, which should be set for six-position operation for both the listen and record switches. Also, make sure that the rings don't fall out as you are fitting the PCB.

That's the tricky bit out of the way. Next, fit the lock nuts for the pot and rotary switches, taking care not to scratch the front panel. Bolt the right-hand panel, then the rear panel into their positions, again using the chassis lid or base as an alignment guide. You will probably need a set of fine-nosed pliers to hold the nuts in place while securing the rear panel.

Solder a length of hookup wire (about 400mm) to the phono ground binding post lug, and connect two twisted wire pairs to the appropriate PCB pads for the indicator LEDs – each pair should be around 100mm long, and formed from (say) a red and black wire to indicate the LED polarity. The actual LEDs (red for ON, and yellow for MUTE) are a push fit into the front panel(s), and are connected as shown in the component overlay with heatshrink tubing fitted to each leg.

The RCA sockets can now be attached to the wire stalks already fitted to the PCB, which should be trimmed for a neat fit as you go. Make sure that the PCB is sitting on the rear panel's



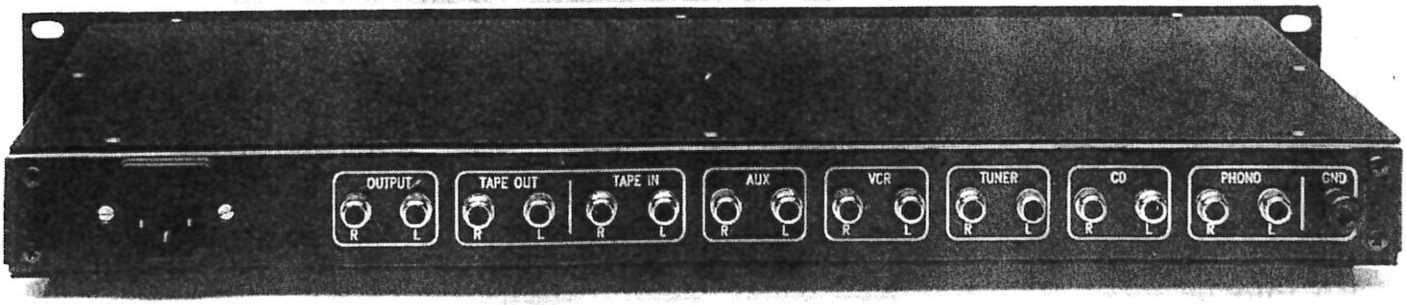
The mains wiring diagram: For a reliable (safe!) contact, scratch away the bottom panel's anodised finish where the earthing lugs are attached.

threaded lugs, and hold each wire in position with a set of pliers as you solder – these double as a heatsink so that the existing joints on the PCB won't be disturbed.

Finally, the bottom panel (with air vents) can be screwed in place, and the power transformer and mains wiring installed. Don't get confused at this point – when looking at the *top* of the completed unit, the *copper side* of the PCB should be showing. Before installing the bottom panel, make sure that the free end of the phono ground wire appears near the power socket. Both this wire, and the main earth lead from the mains socket should be securely terminated to solder lugs, and bolted to the chassis earth point. Scrape or file away the anodised finish at the earthing point, so as to ensure a reliable contact between the lugs and the chassis.

Complete the mains wiring as shown in the associated diagram, and fit the power switch. Both the IEC connector's terminals, and the power switch lugs should be covered by heatshrink tubing – take particular care with this task, as your life may depend on the quality of your work. Note that the IEC connector's internal fuse is wired in series with the mains active lead, which then passes through the power switch to one side of the transformer's primary winding – the neutral is connected directly to the remaining primary lead.

The toroidal transformer is mounted to the bottom panel by a single bolt and top plate, and cushioned by two neo-



The insulated RCA sockets that we used were a tight fit for most RCA plugs. Before using the preamp, you may need to flare the tip retaining part of each socket with a tapered tool.

prene rubber washers. The bolt only requires a moderate tension to hold the transformer firmly in place and shouldn't be overtightened, as this could stress the unit's internal insulation.

You may notice that we have attached a square of plastic to the top of the transformer. This serves to insulate the transformer's mounting plate and bolt from the top panel, and should be included. The reasoning follows that if the panel was forced into contact with these metal parts, a shorted turn would be formed by the chassis and the bolt, resulting in a heavy circulating current and transformer overload. If we assume that the top panel could be deformed, it may be worthwhile placing a piece of cardboard on top of the PCB, before it is finally installed. Nevertheless, the preamp's top panel is *not* the ideal place for a pot plant.

Final checks

In its completed form, the easiest way to ensure that the preamp is functioning correctly is to turn it on, and plug in a set of headphones. However, for the more cautious constructors, a few voltage checks may be prudent.

The first thing to check in this case is the power supply voltages, where the regulator output pins should read close to +15V and -15V. Although you will only have direct access to the copper side of the PCB, it's not too difficult to find the appropriate points by interpreting the component overlay. For a complete check, monitor the voltage at the output of each 5534 op-amp (pin 6) – it should be quite close to 0V.

When the unit is powered up, the ON and MUTE LEDs should immediately illuminate, with the MUTE turning off some 5 or so seconds later. This time period is not critical, so don't be too concerned if yours is a little different due to component tolerances – of course, a gross error would indicate that

Continued on page 112

PARTS LIST

1 1-unit rack-mounting case, black
1 PCB code 90pre8m (main board), 244 x 328mm
1 PCB code 90pre8s (switch board), 27 x 148mm
1 15VA toroidal transformer, 30V centre-tapped, with mounting hardware
1 mains rated SPST miniature rocker switch
1 IEC male chassis socket, with built-in fuse holder
1 IEC-terminated mains power lead
1 M205 500mA slow-blow fuse
2 2-pole 6-position sealed PCB-mount rotary switches
3 22mm diameter black anodised knobs
1 6.5mm PCB-mount stereo socket, with internal DPDT switch
1 PCB-mount DPDT relay, 12V 300 ohm coil
2 FX1115 (or equivalent) ferrite beads
16 panel-mount insulated RCA sockets
1 black binding post
2 solder lugs
1 3-way mains rated terminal strip

Resistors

All 1/4W, 5% unless noted: 2 x 10 ohms, 4 x 39 ohms, 2 x 68 ohms, 2 x 82 ohms, 2 x 100 ohms, 2 x 150 ohms, 8 x 220 ohms, 2 x 390 ohms, 4 x 470 ohms, 2 x 470 (0.5W), 2 x 560 ohms, 4 x 820 ohms, 2 x 1k, 2 x 1.2k, 2 x 1.8k (0.5W), 2 x 2.7k, 2 x 3.3k, 2 x 3.9k (1%), 2 x 4.7k, 1 x 10k, 2 x 12k, 1 x 18k, 9 x 33k, 2 x 47k, 2 x 47k (1%), 2 x 56k, 2 x 68k, 2 x 150k, 2 x 180k, 1 x 220k, 2 x 270k, 2 x 1M, 1 x dual-gang 10k log potentiometer

Capacitors

2 10pF ceramic
4 18pF ceramic
2 22pF ceramic
2 100pF ceramic
2 180pF ceramic
2 1.8nF metallised polyester
2 2.2nF metallised polyester
2 3.3nF metallised polyester
2 18nF (2%) metallised polyester
2 68nF (2%) metallised polyester
13 0.1uF metallised polyester
4 0.22uF metallised polyester
2 0.47uF metallised polyester
2 4.7uf 16VW axial electrolytics
2 22uF bipolar PC-mount electrolytics
2 47uF 16VW PC-mount electrolytics
7 100uF 16VW PC-mount electrolytics
2 220uF 16VW PC-mount electrolytics
4 470uF 25V PC-mount electrolytics

Semiconductors

8 NE5534AN op-amps
1 7815 3-terminal regulator
1 7915 3-terminal regulator
8 2SC2545 low noise NPN transistors
4 BC338 NPN transistors
2 BC328 PNP transistors
2 2N5485 FETs
7 1N4002 diodes
2 5.6V 400mW zener diodes
1 4.3V 400mW zener diode
1 5mm red LED
1 5mm yellow LED

Miscellaneous

Nuts and bolts to suit terminal strip and solder lugs; tinned copper wire; red and black hookup wire; enamelled copper wire (0.3mm or 28 B&S); 21 x PCB pins; heatshrink tubing
3V 400mW zener diode
1 5mm red LED
1 5mm yellow LED

Stereo Preamplifier

Continued from page 99

something is amiss. As another check, you should hear a noticeable click from the unit as the muting relay is energised at the end of this timing period.

The headphone checks may be somewhat of a anticlimax, since you shouldn't hear any noise from the line-level inputs, regardless of the volume control setting. On the other hand, selecting the phono input will produce significant levels of noise, unless the inputs are connected to a normal load – that is, a moving magnet phono cartridge.

In the event of a problem, a little logical deduction and a clear understanding of the circuit operation will quickly point you in the right direction. Fortunately, a preamp can easily be divided into discrete stages for faultfinding purposes, unlike a poweramp for example, which operates as one complicated stage due to the overall feedback loop.

If the preamp is complete and you have to gain access to the component side of the PCB, there should be enough length in the mains wiring to unbolt the bottom panel and rotate it out of the way. Bear in mind however, that the mains earth is connected to the bottom panel and will no longer be connected to the remaining chassis panels. If you need to apply power, temporarily install a clip-lead between the main earth bolt and the unearthed chassis – it's best to play safe.

Clicks, pops & hum

When it comes to using the preamp in the real world, things invariably become a little more complex due to the earthing anomalies and interference created by ancillary equipment. It would be quite frustrating to build a preamp with the exceptional noise figures of the Pro Series Two, only to have the final signal shrouded in noise caused by the rest of the system.

In this regard, we have arranged the preamp circuitry to be floating above the mains earth, avoiding the most likely earth loop problems with the associated power amp, which should have its circuitry referenced to earth. However, if any of the signal sources are also tied to the mains earth, an annoying low-level hum can develop. While conducting performance trials with the Pro Series One power amp, we found that inserting a 1 ohm 10W resistor in series with the link between each poweramp PCB and the chassis greatly reduced these problems.

On the click and pop side of things, the most likely form of interference is from the action of the power amp's power switch. The best cure in this case is to install a 10nF mains-rated capacitor across the offending switch, which helps to dampen the back EMF (and its associated radiation) from the transformer's primary winding. The combination of the Pro Series preamp and power amp has no problems in this regard, and will power up and down with very little audible effects – the capacitors are not required for either unit.

A secondary form of interference can often come from transients created by electric motors cycling on and off – the household refrigerator is a common culprit. While this type of problem can usually be cured at the source, we found the new preamp to be extremely resilient to both this and pure RF-based interference – presumably due to the uncommonly low impedances used throughout the circuitry.

Unlike many other power amp/preamp combinations, we also found no audible side effects (that is, hum) from stacking the preamp directly on top of the poweramp. This is a further advantage of toroidal transformers, as used in both units.

As a final point, if you find that the headphone outlet has insufficient gain for an unusual type of headphones, simply decrease the value of R29 until a comfortable level is found. However, there is a strict limit to the maximum gain you may use.

This is because if you are listening to the main output via a power amp and speakers, and the headphone amp is set to a high gain, it may be driven into hard clipping while the main output is still well within its limits. Now if IC4 is grossly overloaded, it begins to draw significant input current via R28 in sympathy with each clipped peak, which in turn creates a nasty interference signal at the main output. Nevertheless, you can modify headphone amp gain up to a figure of around five, and still maintain the output's very low distortion figures – provided the power amp has a nominal input level of 1V or less.

That's it. If you have built both the Pro Series One power amp and Pro Series Two preamp, you will find that their performance will rival many of the most expensive commercial preamp/poweramp combinations.