Valve Final Amp

35 watts from a no-frills design

Design by Bob Stuurman

This valve power amplifier is a push-pull design using two EL34s (or their 6CA7 US equivalents). It has been kept fairly simple to avoid problems with DIY construction. The output power is well above 35 watts, with low distortion and a wide frequency range. This amplifier provides excellent sound reproduction when used with a pair of reasonably efficient, good-quality loudspeakers — and it shows that even a simple design with quite conventional specifications can sometimes make you tremble with excitement when listening to certain musical passages.



This final amplifier is based on a Philips design dating from the late 1950s, with a few modifications suggested by Claus Byrith.

These modifications consist of a separate supply for the negative grid voltage for the EL34s, an AC balance adjustment for the output stage, an EF86 pentode wired as a triode in the preamplifier stage and a reduction in the amount of overall negative feedback (20 dB). Two documents on this subject have been published on the Internet. They describe the design in detail, and they are certainly worth reading if you are interested in this topic (see 'References').

Since the actual circuit is well documented, we limit ourselves to a brief description in this article. However, we do have a bit more to say about some of the less well-known aspects of the design, because they provide good insight into the

problems associated with push-pull valve final amplifiers and the available solutions. In the first part of the article, we address the theoretical aspects of the design, and in the second part we turn our attention to its construction. Since this is a DIY project, rather than a kit, certain parts of the construction are described in fairly extensive detail.

Schematic diagram

Figure 1 shows the complete schematic diagram of a single channel of the Valve Final Amplifier. There are three supply voltages: a positive high voltage of +440 V, a negative grid voltage of -55 V and a filament voltage of 6.3 V. Separate filament circuits are used for the preamplifier/phase splitter (Fil1 & Fil2) and the output valves (Fil3 & Fil4). The filaments are symmetrically connected to circuit ground via R28 and R29.

The output valves are operated in the 'ultralinear' mode by connecting their screen grids to taps on the anode windings of the output transformer via $1k\Omega$ resistors. Due to the internal negative feedback via the screen grids, the pentodes exhibit characteristics lying between those of a triode and those of a normal pentode. Their internal impedance is reduced to practically the same level as that of a triode, and distortion is reduced to the triode level. However, the output power also drops to around 65 percent of that provided by a pure pentode output stage.

Instead of obtaining the negative grid voltage for the output valves from a voltage drop across the cathode resistors, we use a separate grid voltage supply. This prevents the operating point of the valves from shifting during operation. The magnitude of the negative grid voltage for the output valves can be adjusted using P2 ('DC current'), while the DC balance can be adjusted using P3.

The output stage operates in Class A for small signals, but it shifts increasingly towards Class B as the signal level increases. The current consumption also increases with larger signals. The operating point can be set within certain limits by adjusting the magnitude of the negative grid voltage. Since a separate supply is used for the negative grid voltage, the full anode supply voltage is present across the output valves.

The cathodes are connected to signal ground via 10 Ω resistors (R24 and R25). The voltages across these resistors are proportional to the currents through the valves (10 mV/mA).

Three test points are provided for aligning the circuit. TP0 is circuit ground, while TPV3 and TPV4 are the alignment test points for valves V3 and V4, respectively.

The EL34s provide maximum output power when the voltage on the control grid is



Figure 1. Schematic diagram of the Valve Final Amp.

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Amplifier Specifications

Input impedance: Input sensitivity: Nominal loudspeaker impedance: Maximum output power: Bandwidth at I W: THD+ noise (I W/8 Ω, I kHz): Signal to noise ratio:

$\label{eq:2.1} \begin{array}{l} I \ M\Omega \\ 600 \ mV \\ 8 \ \Omega \ (4 \ \Omega \ optional) \\ 39 \ W \ into \ 8 \ \Omega \\ 5 \ Hz \ - \ >40 \ kHz \\ 0.06\% \ (B \ = \ 80 \ kHz) \\ 62 \ dB \ (B \ = \ 22 \ kHz) \\ 88 \ dB \ (A-weighted) \end{array}$

Performance

A few measured results are shown here. Plot A shows harmonic distortion versus frequency. The lower curve was measured at an output power level of I W, and the upper curve at 27 W. The I W curve in particular is very nice, and this is a typical power level for listening to music. Plot B, which is rather more irregular, shows an FFT analysis of a 1 kHz signal at an output power of I W. The I kHz sine wave has been suppressed by the measuring equipment, and the remaining peaks represent the distortion residuals of the amplifier. You shouldn't be overly alarmed by this plot, since the very wide dynamic range of the analyser (150 dB) gives an exaggerated impression of the actual situation. The most important components are the distortion peaks at 2 kHz and 3 kHz, which lie at -77 dB and -90 dB, respectively. For a relatively simple design using valves and transformer output, this is a very good result. The bulge at 50 Hz results from residual hum in the supply voltage and has nothing to do with the distortion spectrum.



approximately 26 V. This drive level can be easily provided by the phase splitter. The phase splitter is a type having the cathodes connected together and the grid of the second triode (V2b) grounded for AC signals by C6. Since V3a is driven by the grid and V2b is driven by the cathode, there is a small imbalance in the magnitudes of the AC voltages on the anodes. The voltages can be adjusted to be exactly the same using P1 ('AC balance').

The phase splitter exhibits a gain of approximately 26 times, so a signal level of 1 V on the grid of V2a is needed to fully drive the output stage. The high resistance of the cathode resistor (R13) yields low distortion and a high cathode voltage (around 87 V), thus allowing the grid of V2a to be driven directly from the anode of the EF86 preamplifier valve without using a coupling capacitor.

The preamplifier is wired as a triode by connecting the screen grid to the anode, since the high gain that can be obtained with a pentode is not needed. This reduces the noise factor to that of a triode, while retaining the good internal screening and freedom from microphonics characteristic of this value.

A signal level of 60 $\,\,\mathrm{mV}$ on the grid of the

EF86 is needed to fully drive the output stage. Due to the 20 dB of negative feedback provided by R7 and R6, the input level needed to fully drive the output stage is 600 mV. At this level, the output power is 39 W. The amplifier starts to clip at an input level of 0.7 V, which corresponds to an output power of around 46 W.

The resonant frequency of the output transformer due to its leakage inductance is approximately 80 kHz. At this frequency, the open-loop gain must be small enough to ensure that the amplifier remains stable. The necessary gain roll-off is provided by C4 and R8, with a bit of help from C5. The values of these components were determined experimentally using square-wave signals.

When the amplifier is switched on, the high voltage and negative grid voltage are present almost immediately. However, the filaments must warm up before any current can flow through the valves. Diode D1 is thus included to prevent an excessively high voltage from appearing on the anode and screen grid of the EF86. The circuit reaches its normal operating state after a few tens of seconds, with a voltage of approximately 185 V across D1.

RF-suppression ('stopper') resistors are used for the control grids of all the valves. They were present in the original design, so we have kept them here as well.

In the original design, the screen coupling capacitors for the output valves (C9 & C10) had a value of 470 nF. The current through the output valves proved to have rather large fluctuations at a very low frequency (0.2-0.5 Hz), which were also present at the loudspeaker output. This was probably due to small variations in the negative grid voltage. Since these fluctuations have a small amplitude and the output transformer has a large self-inductance, they are not blocked by the output transformer, and they find their way to the amplifier input via the negative-feedback network. This phenomenon was reduced to an accept-





Figure 2. Schematic diagram of the power supply.



able level by decreasing the value of C9 and C10 to 100 nF. This does not have any audible effect on the reproduction of low frequencies.

Power supply

The good characteristics of the Valve Final Amp are in part due to its robust power supply. The Amplimo type 7N607 toroidal transformer, which weighs around 3.5 kg, can provide 340 V at a healthy 700 mA. After rectification and filtering, more than 400 mA at 440 V are available to the amplifier. The winding for the negative screen voltage provides 40 V at 100 mA, which yields an adequate voltage (55 V) after rectification and filtering. The total filament current of the valves is about 7 A. The 6.3-V winding is rated at 6.8 A, but since the load on the high voltage winding is fairly small and practically no power is drawn from the screen-voltage winding, this does not present a problem.

Figure 2 shows the schematic

diagram of the power supply. The high voltage is rectified by four diodes wired in a bridge configuration. The diodes have a surge current rating of 60 A. Interference suppression ('anti-rattle') capacitors are connected across the diodes. Since it is practically impossible to buy high-voltage electrolytic filter capacitors with large capacitance, a pair of 470 µF/400 V electrolytic capacitors are connected in series to achieve an effective capacitance of 235 μ F. Diodes D9 and D10 prevent the capacitors from being reversebiased when the amplifier is switched off. Resistors R1 and R2 divide the voltage evenly across the capacitors and discharge them within several minutes after the amplifier is switched off. C12 provides RF decoupling. Protection is provided by a 315-mA, fast-acting (F) fuse, and it can be a lifesaver for the output valves if the negative grid voltage becomes too small (less negative).

Output transformer

The most important, most critical and invariably most difficult to obtain component of a push-pull valve amplifier is the output transformer. The original Philips design used an output transformer having ten primary windings connected in series, with eight secondary windings interleaved between the primary windings. The secondary windings could be connected in a series/parallel arrangement to obtain the desired input and output impedances. This must have been a real whopper of a transformer, and we estimate that it surely must have weighed more than 5 kgs.

You may be wondering why it was necessary to use a transformer wound in such a complicated manner. The reason is that the ability of a transformer to pass a sine-wave signal decreases as the frequency of the signal increases. Even with very good transformers, the drop-off at 25 kHz is already around 0.5 dB.

Figure 3 shows the equivalent circuit of a transformer driven by an electronic valve. Part (a) shows the situation at very low frequencies. Here the self-inductance of the primary must be high in order to limit the current and allow sufficient magnetic flux to be generated without going into saturation. Part (b) shows the situation at mid-range frequencies, where is a high impedance. Part (c) shows the situation at high frequencies, where the signal is attenuated by the leakage inductance (L_s) and the interwinding capacitance (C_w) . The leakage inductance arises from the 'leakage' of magnetic flux as a result of incomplete coupling between the windings.





Figure 3. Output transformer equivalent circuit at various frequencies.

It takes time for a signal to pass through a transformer, since the low-pass filter formed by the leakage inductance and load impedance creates a time delay. The resulting phase difference between the input and output signals increases with increasing frequency. The output signal thus lags further and further behind the input signal as the frequency increases At 20 kHz, the phase difference can already be 14 degrees. Needless to say, this can have serious consequences for the reproduction of rectangular signals. Fortunately, there is a technique that can be used to deal with the problems of attenuation of high-frequency signals and increasing phase difference at higher frequencies. This technique is negative feedback.

Returning to the output transformer (see Figure 3), we see that L_s and the C_w also form a resonant circuit, so a rapid increase in the phase angle occurs when the signal frequency passes through the resonant frequency of this circuit. This can make the amplifier unstable. Consequently, the openloop gain of an amplifier with negative feedback must be attenuated such that the gain-feedback product (A $\times \beta$) is less than 1 at this frequency. If the amplifier is to have a wide bandwidth, it is thus essential for the output transformer to have a sufficiently high resonant frequency. This requires the leakage inductance and winding capacitance to be small, which can only be achieved using complicated winding methods such as the method used in the previously mentioned Philips output transformer. Naturally, such a transformer cannot be inexpensive.

After some searching, we found a valve output transformer that appears to be eminently suitable for the modified Philips amplifier design. This is the type LL1620PP transformer from the Swedish company Lundahl. This transformer has a 'C' core made from a special type of iron, with two primary windings and four secondary windings on each leg. The two halves of the core are held tightly together on the transformer frame by a welded ribbon. The push-pull version of this transformer (versions for use in single-ended amplifiers are also available) has a small (25 μ m) air gap, so a slight imbalance in the DC currents through the primary windings can be tolerated without causing a large reduction in the primary self-inductance. The four primary windings are connected symmetrically in series, yielding taps at the 50-percent points the of windings that can be connected to the screen grids of the pentode output valves for operation in the 'ultralinear' mode. The eight secondary windings can be connected in series and/or parallel in various manners in order provide an

output impedance of 4 Ω or 8 Ω . At 13 mH, the leakage inductance of the LL1620PP is somewhat on the high side, but this is inevitable with such a large primary self-inductance (no less than 300 H). Since the open-loop gain and negative feedback have both been reduced in the modified version of the amplifier, it remains stable despite the relatively leakage inductance.

The most important specifications of the transformer are listed in the 'Basic LL1620PP Specifications' box. The transformer dimensions are shown in Figure 4a. Paxolin boards with leads numbered as shown in the figure are fitted on both sides of the windings. The winding diagram of the transformer is shown in Figure 4b. Each primary winding is sandwiched between two secondary windings.

To make it easier to use the transformer and reduce the chance of wiring errors, the author has designed three small printed circuit boards for making connections to the transformer. They are not available from Readers Services, but if you want to make them yourself, you can download the lavouts from the Elektor Electronics website (Free Downloads, reference number 020071-1, month of publication). However, it is certainly not difficult to wire the transformer into the circuit by hand. The necessary connections are shown next to each of the circuit board layouts.

For each of these circuit boards, the transformer is located on the 'component side' of the board. The numbers on the boards (1, 8 and 11) correspond to the lead numbers of the

Basic LLI620PP Specifications

Primary/secondary turns ratio:	4 x 9.2 / 8 x
Primary winding DC resistance: *	308 Ω (4 x 77 Ω)
Secondary winding DC resistance:	0.4 Ω
(average per winding)	
Primary winding self-inductance:	300 H
Primary winding leakage inductance: *	13 mH
Primary impedance in this design:	6 kΩ
Secondary impedance in this design:	4 Ω or 8 Ω
Air gap:	25 μm
Transformer loss at 62 W:	0.2 dB
Weight:	2.5 kgs

* all windings connected in series

transformer as shown in Figure 4a.

The connections and circuit board layout for the primary are shown in **Figure 4c**. Simply slip the circuit board over the leads and solder it in place. The connections are marked as follows: supply voltage = Tr+, anodes = A / A^{*}, screen grids = G / G^{*}. Here '*' indicates the start of the winding.

In the original Philips design, the taps for he screen grids were at the 40-percent points of the windings, as measured from the centre tap. Here the proportion is 50 percent, which shifts the output stage more toward triode operation and causes the output power to be somewhat lower. In order to keep the coupling between the anode winding and the screen grid part the winding as tight as possible, windings on the same leg of the transformer have been matched together.

The transformer has eight secondary windings, which can be connected together in series or parallel in various ways in order to obtain the desired secondary impedance for the loudspeaker (4 Ω or 8 Ω) and the required primary impedance (6.0 k Ω). In the 4 Ω configuration, two sets of secondary windings are connected in series, while three sets are connected in series in the 8 Ω version.

The circuit board layout and connections for a 4 Ω loudspeaker connection are shown in Figure 4d (note the two wire links on the bottom side of the board, marked with short lines). Figure 4e shows the circuit board layout and connections for an 8 Ω loudspeaker impedance. In this case, there is only one wire link. Both configurations include a 1 $k\Omega$ shunt resistor at the output (R30). This resistor provides a certain amount of protection for the output transformer if no loudspeaker is connected. It also improves the stability of the amplifier with a capacitive load, such as may be present with a long speaker cable.

The leads for the secondary of the transformer are formed by bringing the tinned ends of the windings out to the terminal board. If you use one of the illustrated printed circuit boards for the 4 Ω or 8 Ω connections, bend the secondary leads flat against the wide tracks on the board and solder them in place.

The construction of the amplifier will be described in the next instalment. Since this involves a fair number of illustrations, a few plots of the measured performance of the amplifier are included in this instalment (see 'Performance').

References www.lundahl.se

(020071-1)





Figure 4. LL1620PP transformer: (a) dimensions and leads, (b) winding diagram, (c) primary winding connections and layout of the optional printed circuit, (d &, e) secondary winding connections and layouts of the optional printed circuit boards for 4- Ω (d) and 8- Ω (e) loudspeakers

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Valve Final Amp (2)

Part 2: printed circuit boards and construction

Design by Bob Stuurman

This final amp is easy to build. The stereo version essentially consists of two amplifier boards, a power supply board for the high voltage and negative grid voltage, two output transformers and a power transformer. We have designed two printed circuit boards for building the final amp, but it can also be constructed in the 'old-fashioned' manner using solder turrets.



The chassis is made of aluminium and consists of two parts: an open-ended U-shaped channel section and a flat plate resting on top of the channel section. The channel section is fitted upside down, with the output transformers on top and the power transformer underneath. The combined weight of the transformers alone is more than eight kilos, and using a channel section gives the chassis adequate stiffness.

The rear wall of the channel section is aligned with the rear edge of the plate. All of the connectors are mounted on the rear wall, along with the master volume control. An IEC appliance socket with integrated filter, switch and fuse holder is used to keep the 230-VAC wiring to a minimum. There is no need for a pilot lamp, since the valves glow nicely when the amplifier is on.

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Figure 1. Copper layout and component layout of the printed circuit board for one amplifier channel.

Safety precautions

Hazardous voltages are present in this amplifier. The electrolytic capacitors in the power supply have a large capacity, so it takes quite a while for the high voltage to drop to a safe level after the amplifier is switched off. For this reason, you should connect two 230-V 15-watt incandescent lamp bulbs in series across the high voltage while the amplifier is being tested. As soon as the mains voltage is switched off, they will discharge the electrolytic capacitors in a few seconds, and they will have practically no effect on the operation of the amplifier.

Amplifier construction

The copper track and component layouts of the amplifier printed cir-

cuit board are shown in **Figure 1**. The only component that is not included on the board is the output transformer. The circuit board is single-sided, and using the artwork shown here (and available from our website) some of you will be able to make it themselves. However, the board it is also available ready-made from Readers Services (order number **020071-1**). Two of these boards will be needed for a stereo version of the amplifier.

All connections to the circuit boards are

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COMPONENTS LIST

Amplifier (one channel)

Resistors:

All fixed resistors: metal film, Beyschlag type MBE0414 or BC Components type PR-02, dim. 4x12 mm.

 $RI,R2,RII = IM\Omega$ $R3 = 4k\Omega7$ $R4,R17,R18 = 47k\Omega$ $R5 = 390\Omega$ $R6,R22,R28,R29 = 100\Omega$ R7 (LS = 8Ω) = $3k\Omega$ 3 R7 (LS = 4Ω) = $2k\Omega 2$ $R8 = 27k\Omega$ $R9 = 100k\Omega$ $R10,R26,R27,R30 = 1k\Omega$ $R12,R14 = 150k\Omega$ RI3 = 82k $RI5 = I5k\Omega$ $RI6,RI9 = 390k\Omega$ $R20, R21 = 2k\Omega2$ $R23 = 10k\Omega$ $R24, R25 = 10\Omega$ $PI = 50k\Omega$ preset P2 = 10 k presetP3 = 20 k preset (All presets: Bourns type 3386P)

Capacitors:

All film capacitors: Wima type MKS4, unless indicated otherwise.

CI = 470nF 100V, lead pitch 15mm C2 = 100nF 400 V, lead pitch 15mm $C3 = 10 \mu F 350 V \text{ or } 450 V$, axial, dim. 12x25 mm C4 = 100 pF 630V, polypropylene, dim. 5x11 mm C5 (for LS 8Ω) = 680pF 630V, polypropylene, dim. 5.5x15 mm C5 (for LS 4Ω) = 1000pF 630V, polypropylene, dim. 5.5x15 mm C6,C12,C13 = 220nF 250V, lead pitch 15mm C7,C14 = 470nF 630V, lead pitch 27.5 mm $C8 = 10\mu F 450V$, axial, dim. 15x30 mm C9,C10 = 100nF 630V, lead pitch 22.5 mm $CII = 470 \mu F 63V$, radial, dim. I2.5x25 mm

Semiconductors: DI = 200 V I.3 W zener diode

Valves:

VI = EF86 (US: 6267) V2 = ECC83 (US: 12AX7) V3,V4 = EL34 (US: 6CA7), matched

Miscellaneous:

2 noval (9-way) valve sockets, ceramic 2 octal (8-way) valve sockets, ceramic Tr1 = output transformer, Lundahl type LL1620 P-P PCB, order code **020071-1** (see Readers

Services page)



Figure 2. Copper layout and component layout of the printed circuit board for the power supply.

made using solder posts with a diameter of 1.3 mm and matching connectors. Noval valve sockets are used for V1 and V2. These sockets are available in plastic and ceramic versions; the circuit board has been designed for the ceramic version.

Ceramic octal sockets are used for V3 and V4, the EL34s. They have solder tabs with a width of 2 mm and a thickness of 0.5 mm. In order to allow the sockets to be fitted flat against the circuit board, the drilled holes for the solder tabs must be

widened somewhat by (mis)using a circuit board drill as a routing bit.

The circuit board has six mounting holes, which allow it to be firmly attached to the base plate. This provides extra support for the portion holding the output valves.

If you stick to the parts shown in the components list, building the printed circuit board is a breeze; everything fits perfectly. The PR-02 resistors from BC Components (formerly Philips) are 1% types and have four colour-coding bands.





Since it can be difficult to read their values from these bands, it's a good idea to always check them with an ohmmeter.

The valve sockets are soldered to the copper side of the circuit board. In order to align the individual contacts properly while soldering them in place, you should insert the valves in the sockets. When fitting the octal sockets, be careful to orient the notches properly. The sockets will 'fit' in all possible orientations, and it's next to impossible to remove a socket once it's been soldered in place.

The single-sided printed circuit board for the power supply (**Figure 2**) is available from Readers Services under order number **020071-2**. Here again, 1.3-mm solder posts with matching connectors are used. Building the power supply board is so simple that we don't need to say anything about it, except to remind you to watch the polarity of the diodes and electrolytic capacitors.

Building the amplifier

The dimensions of the chassis plate and channel section are shown at the lower left of the wiring diagram (Figure 3). The channel section is made from a piece of aluminium sheet 370 mm long and 290 mm wide, with its long edges folded to form a U-shaped channel with 80mm walls.

The convenient feature of this chassis is that the channel section and plate can be prepared separately. However, some of the holes must be made in both the channel section and the plate, which requires the two parts to be temporarily bolted together. For this purpose, you can drill holes for 2-mm screws inside the outlines of the transformer covers.

For the next stage, you will need paper templates, preferably made from tracing paper. The templates for the amplifier boards and the power supply board can be made by simply copying the component layouts, since they show the dimensions of the circuit boards and the locations of the mounting holes. For the output transformers and their covers, you will have to make a drawing showing the outside dimensions (of the cover) and the locations of the drilled holes. The template for the power transformer consists of a circle and its centre point. Make templates for

COMPONENTS LIST

Power supply

Resistors:

 $\label{eq:R1R2} \begin{array}{l} \text{R1,R2} = 47 k\Omega \text{ (Beyschlag type MBE0414 or BC Components type PR-02, dim. 4x12 mm)} \end{array}$

Capacitors:

 $\begin{array}{l} {\sf C1-C4} = 100nF~400V, {\sf lead~pitch~15~mm} \\ {\sf C5-C8} = 100~n/1000~V, {\sf lead~pitch~22.5~mm} \\ {\sf C9} = 470\mu F~63V, {\sf radial, lead~pitch~5~mm,} \\ {\sf dim.~12.5x25~mm} \\ {\sf C10,C11} = 470\mu F~400V, {\sf radial, lead~pitch~10} \\ {\sf mm}~({\sf e.g., Roederstein~series~EYS}) \\ {\sf C12} = 100nF~630V, {\sf lead~pitch~22.5~mm} \end{array}$

Semiconductors:

DI-D4,D9,D10 = IN4007 D5-D8 = BYW96E

Miscellaneous:

Fuse, 315 mA (fast) with PCB mount holder Mains transformer, secondaries 340V at 0.7A, 6.3V at 6.8A and 40V at 0.1A (Amplimo # 7N607) PCB, order code **020071-2**

Miscellaneous parts

- IEC mains appliance socket with integral filter, switch and fuse holder, fuse I.5A(T) (time lag)
- 2 NTC-resistors, 5 Ω 5 W (Amplimo or Conrad Electronics)
- Audio potentiometer, $100k\Omega$ stereo, logarithmic law (e.g., Alps type RK-27112) with knob
- 2 cinch sockets, chassis mount (isolated)
- 2 binding posts, red (isolated)
- 2 binding posts, black (isolated)
- Terminal block strip
- Covers for output transformers

SUGGESTED SUPPLIERS

Lundahl transformers

Lundahl Transformers AB, Tibeliusgatan 7, SE-761 50 Norrtälje, SWEDEN. Tel. +46 176 139 30, Fax +46 176 139 35. Distributor overview at <u>www.lundahl.se</u>

Valves and valve sockets

Chelmer Valve Co. (www.chelmervalve.com), Conrad Electronics (www.int.conradcom.de), Amplimo (www.amplimo.nl)

PR-02 resistors

Farnell (<u>www.farnell.co.uk</u>), C-I Electronics (<u>www.dil.nl</u>)

MKS capacitors

Farnell (www.farnell.co.uk), C-I Electronics (www.dil.nl), Conrad Electronics (www.int.conradcom.de)

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Figure 3. Sample wiring diagram and mechanical layout (bottom view) for a stereo amplifier.

Alignment

An 8- Ω or 4- Ω load, as appropriate, must be connected to the loudspeaker output of the amplifier during alignment and whenever measurements are being made on the amplifier. Several power resistors attached to a heat sink can be used for this purpose. If the amplifier is not loaded, arcing can occur in the output transformer, possibly resulting in a defective transformer.

The output valves are not self-biasing, since a negative grid voltage is used instead of cathode bias resistors. Consequently, they should preferably be purchased as matched pairs.

The following items must be aligned in the order listed: DC current, DC balance and AC balance. The characteristics of the valves change as they age, so it is advisable to check the settings every two weeks at first, and after that every two months. The current through the output valves fluctuates somewhat, which makes it difficult to use a digital voltmeter to make the adjustments. An analogue moving-coil meter is much easier to use for this purpose. Since the adjustments need to be



made repeatedly, an alignment aid is a handy accessory. For this purpose, a pair of three-way female headers (one for each amplifier board) can be fitted in convenient locations using double-sided adhesive strips. The middle contact is connected to Tp0, and the outer contacts are connected to TpV3 and TpV4, respectively. The alignment aid can then be connected using a length of cable with a 3-way circuit-board header.

The current flowing through each EL34 should be 50 mA (combined anode and screen-grid currents). This yields a power dissipation of around 22 W for each valve. At this level of current, the voltage across the cathode resistor of each valve will be 0.5 V.

The circuit diagram of the alignment aid is shown next to this box. It also has to be aligned before it can be used. To do so, connect a DC voltage of 0.5 V to terminals Tp0 and TpV3 of the alignment aid and set S1 to the topmost position ('DC current'). Then adjust P1 until the meter shows a value of 50 (read mA for μ A).

When SI is in the 'DC balance' position, the circuit measures the voltage between TpV3 and TpV4. If the currents through the two valves are equal, the meter reading will be 0. The nice feature of this circuit is that it has higher sensitivity for this adjustment, since the only series resistance is provided by RI.

When SI is in the 'AC balance' position, TpV3 and TpV4 are tied together and connected to a headphone plugged into K1. The alignment signal can be heard using the headphone.

Adjusting the DC current and DC balance

On each amplifier board, first set PI and P3 to their midrange positions and rotate P2 fully counter-clockwise, so that the negative grid voltage has its maximum negative value. Connect the alignment aid with its switch set to 'DC current,' and then switch on the power. Wait a few minutes, and then adjust P2 for a meter reading of 40 mA. Next, change SI to the middle position ('DC bal-ance') and adjust P3 to obtain a meter reading as close as possible to 0. After the amplifier has warmed up for ten minutes, you can increase the DC current to 50 mA and tweak the DC balance as necessary.

Adjusting the AC balance

The AC balance of an amplifier is usually adjusted using a distortion meter. Mr Byrith has devised a method to allow this be done using an audible signal. Set the switch to the 'AC balance' position and connect a sine-wave signal to the input of the amplifier (1 kHz / 100 mVrms). While listening to this signal with the headphones, rotate P1 until the 1-kHz tone is as weak as possible. You will also hear mains hum and harmonics of the sine-wave signal, and the loudness of the signal will fluctuate, but it is certainly possible to find a setting where the 1-kHz tone is at a minimum. The signals on the cathodes have opposite phases, and when they are in balance they have equal amplitudes. Clever thinking!

Square-wave alignment

Capacitor C5 in the feedback loop corrects the phase lag. If its value is a bit too small, the corners of a square-wave signal will be rounded off, and if its value is a bit too large, the corners will have overshoots. You need to have access to a square-wave generator and an oscilloscope if you want to check and/or adjust the square-wave response.

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Figure 4. Bottom view of the fully assembled amplifier.

the IEC appliance socket and the Alps volume control as well.

Tape the templates to the chassis plate such that the amplifier boards are spaced 13 mm from the front and side edges (this clearance is required for the supporting strips in the case). The fixing nuts for the transformer covers must fit inside the channel section. Align the C cores with each other, and position the power transformer in the middle of the channel section.

Now you can centre-punch and drill all of the holes. For each output transformer, two holes are needed to allow the wiring to pass through the chassis. If they are drilled within the outline of the cover, they will be hidden when the assembly is finished.

Drill six holes with a diameter of 8 mm around the openings for the output valves to allow cooling air to flow past the EL34s, since they become rather hot.

Run the wiring for the filament supply in a length of small cable duct stuck to the inside of the front wall of the channel section ('pos 1' in the detail at the lower right of the wiring diagram). Make feedthrough openings at the positions of the filament connections on the circuit boards.

Run the wiring for 0 V, -55 V and +440 V in a second small cable duct located at 'pos 2'.

Attach the amplifier boards to the chassis plate using 10-mm standoffs. Adjust the separation between the boards and the plate using shim washers so that the sockets for the output valves are firmly pressed against the top plate. Fit the power supply board using standoffs as well.

Fit an aluminium screening plate between the amplifier boards, and use a sheet-metal enclosure to screen the Alps volume control.

Testing

As long as the amplifier boards are not yet fitted, everything is easily accessible. In order to test the amplifier boards, it's convenient to first assemble the power supply portion. Fit the power transformer and the power supply circuit board to the channel section, along with the IEC appliance socket. Install a 1.5-A slow-blow fuse. In our amplifier, we fitted two four-way connector strips to a piece of epoxy board using countersunk 3-mm screws, and then secured this board to the fitting screw for the power transformer using an extra nut. The lower set of terminals (as shown in **Figure 3**) is for the filament wiring.

Practically all of the wiring, except the heavy leads for the loud-speaker terminals, consists of 0.5-mm² flexible hookup wire with various colours of insulation. Three such wires can be easily fitted into a connector strip terminal.

The four upper connector-strip terminals are used to connect the primary leads of the power transformer to the IEC appliance socket. An NTC resistor is placed in series with each lead, in order to reduce the switch-on surge. They are not absolutely necessary, but they are a simple and effective way to achieve a 'soft' switch-on.

Once the wiring interconnecting the IEC appliance socket, mains transformer and power supply board is finished, you can begin testing by checking the power supply by itself. First connect the two 230-V/15-W incandescent lamps in series between the +440 V and 0 V terminals, and then switch on the power. If the lamps light up brightly, you can then (carefully!) check the high voltage and negative grid voltage.

After switching off the power, connect the loose amplifier boards to the power supply and output transformers. Before applying the high voltage, first check that the filaments of the valves light up. With the EF86, you can see this by looking in from the top, although it is a bit tricky. Next, switch off the power and remove the output valves, and then connect the high voltage leads. Switch on the power and allow the EF86s and ECC83s to warm up, and then check the voltages on these valves. Small variations from the nominal values are possible, but a major deviation means that there is probably an incorrect resistor value somewhere.

If everything is OK, switch off the power and plug in the output valves. Now you can perform a preliminary alignment of the amplifier (see the 'Alignment' box). After this, you can fit the amplifier circuit boards in the enclosure and route the rest of the wiring.

Finishing

Such a nice amplifier naturally deserves an attractive wooden case. We made our case from lengths of 9mm multiplex board, after finishing off the openings in the chassis plate and channel section. There are two rectangular openings in the back of the case for the connectors and volume control. Our case is finished with veneer, but it would naturally also be possible to build a case using solid wood. Self-adhesive feet are fitted below the channel section.

Even without a case, the amplifier sits quite stably on the walls of the aluminium channel section. If strips of wood are taped onto the transformer covers, the completed amplifier, with the valves installed, can also be placed upside down on top of a table. This makes it easy to access all of the circuitry, and it is also convenient for fitting a wooden case.

The bottom of the case can be closed with an aluminium plate if desired. If you use such a plate, be sure to earth it, and drill openings for cooling airflow.

It's a good idea to switch on all the other equipment in your audio system before switching on the final amp, in order to avoid a switch-on 'thump'.

(020071-2)

Advertisement

Valve Preamplifier (I)

Classic technology revisited

Design by B. Stuurman



This stereo preamplifier is primarily intended to be used as a companion to the Valve Final Amplifier described in the April and May 2003 issues, but it can easily be used with other power amplifiers as well. The circuit has all the usual controls and input options, and it is relatively easy to build, even if you're not a valve guru.

Like most preamplifiers, this one has controls for volume, tone (treble and bass) and balance. The input options also correspond to the usual scheme, the only exception being that this preamplifier has a phono input — which can no longer be taken for granted these days. The complete stereo version consists of five printed circuit boards: an amplifier board for each channel, an I/O board holding all the Cinch connectors and relays, a high-voltage sup-

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Figure 1. The active components of the preamplifier consist of an EF86 and an ECC82. FET T1 is a 'bonus'.

ply board and a low-voltage supply board. The last of these provides 12.6 V DC for the valve filaments, among other things.

Two valves

As can be seen from the schematic diagram of the amplifier (**Figure 1**), no overall feedback is used, but only local feedback in the phono preamplifier. This is because valves are manufactured to such tight tolerances that excellent results can be achieved even without using feedback.

The circuit is quite simple. The actual control amplifier is built around V2a and V2b, an ECC82

(12AU7). Volume control P1 is placed directly at the input of the control amplifier, and a relay is used to select one of two possible signal sources: the output of the phono stage or the line input (pins 3 & 4 of K2). The tone control is placed between V2a and V2b. This is a passive control using logarithmic potentiometers. P5 is included at the output of V2B to adjust the balance.

Amplifier details

The phono preamplifier is built according to a modified Philips design, using an EF86 or its equivalents the 6267 or CV2901 (V1). This is an outstanding valve for this purpose, due to its low noise and minimal microphonics. The necessary RIAA compensation is obtained by using frequency-dependent feedback. At the average output level of an MD cartridge, the output voltage of the phono stage is approximately 54 mV, which is also the input sensitivity of the following control amplifier.

K2 is the Line input. A CD player, tuner or the like can be connected here via the I/O board (which is described below). Relay Re1 is normally connected to the Line input, and the CD player, tape, tuner or auxiliary input is selected by a relay on the I/O board.

Despite our original intention, it was ultimately decided to add a Line output (K2) to allow recordings to be made. This output should be regarded as a sort of extra, since the FET amplifier stage used here (T1) is a



Figure 2. The I/O circuit uses relays to dramatically reduce the amount of wiring.

'foreign' element in a valve amplifier.

The tone control, in which P2 adjusts the treble and P4 adjusts the bass, is a standard circuit. At the 'straight-through' setting, the control has an attenuation factor of 14, which means that quite a bit of gain is necessary to achieve the input level required by the valve final amplifier described in the April and May issues. This gain is provided by an ECC82 (12AU7) dual triode (V2). Each triode section provides a gain of approximately 11, so the total gain is more than adequate.

The balance control (P5) of this amplifier is rather unusual. Although a 'normal' balance potentiometer could be used here, two logtaper potentiometers were used in the prototype (one for each channel). When a log-taper potentiometer is at the centre of its mechanical range, it is set to approximately 16 percent of its total electrical resistance. With a 50-k Ω potentiometer, the resistance between the wiper and the start of the track is then 8 k Ω , while the resistance between the wiper and the end of the track is 42 k Ω . If the potentiometer is wired in reverse, the additional series resistance (when the potentiometer is set to the midrange position) can be limited to 8 k Ω , while still allowing the signal to be reduced to zero. The two (single) potentiometers must be mechanically coupled such that they rotate in opposite directions. This can be easily done using a pair of gears.

The power supply provides a filament voltage of 12.6 V. An ECC82 can be directly connected to this voltage, but an EF86 can only be connected to 6.3 V, so two valves (one on each of the two amplifier boards) must be connected in series when a 12.6-V supply is used. This can be easily done using connector K2. For one channel, pins 7, 8, 9 and 10 are connected together using jumpers, while for the other channel, pins 11, 12, 13 and 14 are connected together. After this, pins 11 and 12 for the first channel are connected to pins 9 and 10 of the circuit board for the second channel. For a mono amplifier, a 330- Ω , 2-W resistor can be inserted in the circuit.

The final detail is that in our version of the amplifier, a wire link was used in place of R10. If the gain of the ECC82 proves to be too large, a resistor can be fitted to convert R1 and R11 into a voltage divider.

I/O circuit

The I/O circuit shown in Figure 2 contains all of the cinch connectors for the preamplifier, which are fourteen in total. Attenuators are included in series with the CD, Tape, Tuner and Aux inputs. They consist of resistors R1-R4 (or R6-R9 for the second channel) and the common resistor R5 (or R10). The amount of attenuation can be adjusted as necessary by modifying the values of the series resistors. This may be necessary with the CD input in particular, since some CD players provide an output voltage that is several times greater than the line input sensitivity of 250 mV used here. By the way, if you use a series resistor with sufficiently high resistance for the Aux input (e.g., 220 k Ω), you can connect a phonograph with an oldfashioned ceramic hi-fi cartridge directly to the Aux input. This type of cartridge provides a sufficiently high output voltage and does not require RIAA compensation, as do MD cartridges.

Relays are used to keep the wiring of the inputs simple. A switchover relay is located on each of the amplifier boards. When it is energised, the output of the phono stage is connected to the input of the control amplifier, while otherwise the output of the I/O board is connected to the input of the control amplifier. The CD, Tape, Tuner and Aux inputs on the I/O board are connected to the Line In input via relays Re1–Re4 (or Re5–Re8 for the second channel). The desired input is selected by energising the associated relay.

One final remark regarding the I/O circuit: if you are not absolutely certain that the outputs of all of your signal sources are free of DC voltages, it is recommended to place 2.2- μ F MKT (metallised plastic) capacitors in series with Line In, between the I/O board and the amplifier boards (Cx and Cy in Figure 2).

Power supply

The preamplifier requires two supply voltages: a high voltage of 260 V and a filament voltage of 12.6 V. In order to avoid availability problems, two standard toroidal-core transformers

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are used here. The first transformer (Tr1) converts the mains voltage to 15 V. After rectification and stabilisation, this voltage is used to power the filaments. The secondary voltage is also converted back to approximately 230 V by the second transformer (Tr2). The toroidal-core transformers have practically no external magnetic fields, so they are humfree. They can be mounted one on top of the other in order to save space in the enclosure.

The complete schematic diagram of the power supply is shown in Figure 3, with the low-voltage portion at the bottom and the high-voltage portion at the top. Tr1 is a 30-VA type, while Tr2 is a 15-VA type (the smallest in the series). The transformers have two secondary windings, which are connected in parallel. Rectification is provided by D1-D4 and D5-D8. A noise-suppression capacitor is connected in parallel with each diode, in order to slow down the switching of the diodes and thus prevent them from generating RF interference. C9 and C11 are the buffer capacitors for the low voltage and high voltage, respectively. The ground terminal (0 V) of the low-voltage supply is labelled '0V''.

The low voltage is stabilised by an adjustable voltage regulator using the well-known LM317. Trimpot P1 can be used to adjust the output voltage to exactly 12.6 V. If jumper J1 is fitted, the voltage divider is changed such that the output voltage can be set to 6.3 V. This allows the power supply to also be used for simpler valve circuits, although the amount of current that can be drawn in this configuration is much less (approximately 400 mA). C10 improves the transient response, and LED1 indicates that the power supply is operating. In addition, the LED and R4 provide a minimum load in the absence of an external load, to ensure that the required minimum operating current is drawn from the IC.

The job of the IRF740 MOSFET (T1) in the high-voltage portion of the supply is to largely eliminate 100-Hz ripple. For a normal load current of 50 mA (each control amplifier needs approximately 15 mA), the ripple voltage across C11 is



Figure 3. The power supply consists of separate high-voltage and low-voltage sections.

 $U_{\rm r} = 1.7 (I / C) = 1.7 (50 / 220) = 0.4 \, {\rm V}$

where:

Ur = peak-to-peak value of the ripple voltage I = load current in mA C = value of C11 in μ F

 $C = value of C11 in \mu F$

If we provide a ripple-free voltage at the gate of T1, such that this voltage is much lower than the ripple voltage and provides sufficient leeway for variations in the mains voltage, the voltage on the output (source of T1) will also be free of ripple. Here R5, R6 and C12 provide the necessary voltage.

The circuit around T1 also provides a 'soft-start' characteristic for the high voltage supply. After the supply is switched on, the high voltage increases gradually and attains its final value after approximately 3 seconds. It also decreases gradually when the supply is switched off, due to the large capacitance of C11. These soft-start and soft-stop characteristics prevent switch-on and switch-off pops. D9 discharges C12 after the supply is switched off, R7 ensures that no voltage is present at the output after a few minutes and C13 provides HF decoupling. Fuse F1 protects T1 in the event of a short circuit on the output. Incidentally, it is not necessary to use an external heat sink for the FET, even if it has to supply 50 mA.

Construction

Amplifier circuit board

The printed circuit board for the control amplifier is shown in **Figure 4**. Although it is certainly not difficult to build, a few remarks are in order.

Sockets with plastic bodies are used for the valves. These sockets are fitted to the component side of the board, as usual. The SIL connectors for K1, K2, P1, P2, P4 and P5 are supplied in long strips, and individual lengths can be sawn off using a coping saw with a metal-cutting blade. Each connector has at least two contacts, so they cannot twist around. To minimise the chances of incorrect connections, male contacts are used on the amplifier board and female contacts are used on the I/O board.

For some of the capacitors, the board is





Figure 4. The printed circuit board for a single-channel (mono) amplifier board. Two such boards are needed for a stereo preamplifier.



Figure 5. You can check your fully assembled amplifier board against this photo.



COMPONENTS LIST

Amplifier board

Resistors: $RI = 680k\Omega$ $R2.R12 = 150k\Omega$ $R3 = 68k\Omega$ $R4, R9, R17 = 2k\Omega 2$ $R5,R11 = 100k\Omega$ $R6 = 390k\Omega$ $R7 = 10M\Omega$ $R8,R19 = 10k\Omega$ $RI0 = 0\Omega$ (wire link, see text) $RI3 = I5k\Omega$ $R|4.R|6.R20 = 47k\Omega$ $RI5 = Ik\Omega2$ $RI8 = Ik\Omega5$ $R2I = Ik\Omega$ All resistors: metal film 0.5-1W, e.g., Beyschlag MBE-0414

P1,P2,P4 = 500kΩ logarithmic, stereo

laid out for two different sizes, so it should always be possible to find a suitable type. PCB pins are used for the high voltage and filament voltage connections.

The voltages shown on the schematic diagrams are valid when one amplifier board is connected to the power supply. Variations of up to 10 percent are possible. The voltage on the drain of T1 may range from 6 to 9 V. If this is not the case, replace the FET, since there can be a large

COMPONENTS LIST I/O board

Resistors: RI-R4, R6-R9 = 100kΩ R5,R10 = 20kΩ

Capacitors:

 $Cx,Cy = 2\mu F2 MKT$ (metallised plastic) (see text)

Semiconductors: DI-D8 = IN4148

Miscellaneous:

Re1-Re8 = SI relay, 1 make contact, 12V coil, e.g., Conrad Electronics # 504602.
14 cinch chassis sockets.
SIL PCB header, 5-way.
6 solder pins with mating sockets.
PCB, order code 020383-3 (see Readers Services) $\begin{array}{l} \mathsf{P3}=20 k\Omega \text{ preset, horizontal, lead pitch} \\ 5.08/2.54 \text{mm} (\text{Conrad Electronics }\# \\ 424420) \text{ or } 5/10 \text{mm} \\ \mathsf{P5}=50 k\Omega \text{ logarithmic} \end{array}$

Capacitors:

CI = 100pF 250V polypropylene, e.g. Conrad Electronics # 458686 C2 = 330pF 250V polypropylene, e.g., Conrad Electronics # 458740 $C3 = 10\mu F$ 385V, axial, 12x30 mm C4,C10 = 15nF 630V, lead pitch 10mm C5,C14,C17 = 220µF 35V, radial, lead pitch 5mm $C6 = 2\mu F2 250V$, lead pitch 27.5mm or 22.5mm C7,C12 = 100nF 400V, lead pitch 15mm C8 = 4nF7 630V, lead pitch 10mm or 7.5mm C9 = 270pF 250V polypropylene, e.g., Conrad Electronics # 458732 CII = InF 630V, lead pitch 10mm or 7.5mm

C13 = 22 μ F 350V, axial, 16x36 mm C15 = 1 μ F 250V, lead pitch 22.5mm C16 = 10 μ F 35V tantalum, lead pitch 2.5mm

Valves:

VI = EF86, 6267 or CV2901 V2 = ECC82, E82CC or 12AU7

Semiconductors:

TI = BF245A

DI = IN4I48

Miscellaneous:

- ReI = DIL relay, 1x changeover, 12V coil.
 2 Noval (9-pin) valve sockets, PCB mount, pin circle 18mm, e.g., Conrad Electronics # 120529.
- IC socket, 14-way, turned pins (for Re1).
- 36-way pinheader, straight.

36-way socket, straight.

- 4 PCB solder pins with mating sockets. PCB, order code 020383-1 (see Readers
- Services).





Figure 6. The I/O circuit board can be used directly as a connector panel. 'L' and 'R' naturally stand for 'left' and 'right'.

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Figure 7. This photo shows how the components should be mounted on the I/O circuit board.

amount of individual variation in the FET characteristics.

A photograph of the fully assembled amplifier board is shown in Figure 5.

I/O circuit board

The printed circuit board for the I/O circuit is shown in **Figure 6**.

Tin the ground surfaces before the fastening the cinch connectors to the board. The inputs of the connectors are located on the copper side of the board. Place the supplied spring washers under the nuts and firmly tighten the nuts. Fit the other components only after the Cinch connectors have been fitted.

Each stereo channel has two ground points, labelled 'Gnd Out' and 'Gnd L/R In', which must be connected to the corresponding points on the control amplifier



Figure 8. The two parts of the power supply circuit board must be sawn apart.

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Figure 9. The fully assembled power supply boards. Pay attention to the polarity of the diodes!

boards. Part 2 of this article will include a wiring diagram showing how the screens of all the screened wiring should be connected in order to avoid ground loops. The rest of the wiring will also be described in detail in this diagram.

COMPONENTS LIST Power supply boards

Resistors:

- $RI = Ik\Omega 2$
- $R2 = 270\Omega$
- $R3 = 2k\Omega 2$
- $R4 = Ik\Omega$
- $R5 = 47k\Omega$
- $R6 = IM\Omega5$ $R7 = I00k\Omega$
- $PI = 500\Omega$ preset, horizontal, lead
- pitch 5/2.5mm or 5/10mm All resistors: metal film 0.5-1W, e.g.,
- Beyschlag MBE-0414

Capacitors:

- CI-C4 = 100nF 100V, lead pitch 10mm
- C5-C8 = 100nF 630V, lead pitch 22.5mm
- $C9 = 4700 \mu F 40V, radial, lead pitch 10mm or 12.5mm$
- $C10 = 100\mu F 63V$, radial, lead pitch 5mm
- $CII = 220 \mu F$ 400V, radial, lead pitch I0mm
- $C12=22\mu F$ 350 V, axial, 16x36 mm

Power supply boards

Two printed circuit boards have been designed for the power supply. One is for the low-voltage portion, and the other is for the high-voltage portion. These two boards are supplied as a single board, so they must

C13 = 100nF 630V, lead pitch 22.5mm

Semiconductors:

 $\begin{array}{l} \text{D1-D4} = 1\text{N5404} \\ \text{D5-D9} = 1\text{N4007} \\ \text{IC1} = \text{LM317} \\ \text{T1} = \text{IRF740}, \text{BUZ61 or} \\ \text{BUK455-400B} (\text{V}_{\text{DS}} \geq 400\text{V}) \\ \text{LED1} = \text{LED}, \text{5mm dia., red} \end{array}$

Miscellaneous:

TrI = toroidal transformer230V/2x15V, IA, Amplimo type 11013 (www.amplimo.nl) Tr2 = toroidal transformer 230 V/2 x15 V, 0,5 A, Amplimo type 1013 (www.amplimo.nl) JI = 2-way PCB pinheader Heat sink, R_{th} 5 K/W, e.g., Fischer type SK129/50,8/STS PCB mount fuse holder with cap. 5x20mm, lead pitch 22.5mm FI = 100 mAF (fast acting) fuse 8 solder pins with mating sockets (sleeves) PCB, order code 020383-2 (see Readers Services)

first be sawn apart, since they will be fitted on opposite sides of the 'transformer block' (the two transformers stacked together).

Assembling the circuit boards is a simple job. PCB pins and push-on mating receptacles (sleeves) are used here as well for the connections. The electrolytic capacitor shown in the components list for position C9 has three terminals, but the circuit board can also accommodate a type with two terminals. Use heat conducting paste when fitting IC1 to the heat sink.

After inspecting the assembled circuit boards, it's a good idea to temporarily mount them on a wooden board along with the transformers, so you can quickly check out the power supply boards. For safety, a test load consisting of a 230-V, 8-W lamp can be connected to high-voltage output, Without the lamp, the voltage on the output should be approximately 260 V, while with the lamp it will be a good deal lower. Be careful when measuring this voltage!

While you are checking out the power supply, you can also adjust the filament supply voltage to exactly 12.6 V.

Figure 9 shows what the fully assembled power supply boards should look like.

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Next month, we will continue with the assembly and wiring of the preamplifier, as well as an enclosure. We'll also have a close look at its specifications.

9/2003

Valve Preamplifier

Part 2: construction

Design by Bob Stuurman

In the first part of this article, we described all of the circuit boards. Now it's time to look at how they fit together. We first describe the mechanical construction, after which we turn our attention to wiring the entire assembly. We conclude with a brief look at the specifications and performance.



An important factor with this type of equipment is the enclosure. For the prototype, we selected a standard model from the Conrad line. It consists of black top and bottom covers made from robust 1.5-mm sheet steel and two 1-mm aluminium face panels. The top and bottom covers have ventilation openings. The edges of the face panels are bent at right angles to form lips, to which the top and bottom covers are fastened using four self-tapping screws. The face panels have a brushed matte finish and protective cover foils. For the sake of simplicity, we call the face panel at the front the 'front panel' and the other face panel the 'rear panel'.

Mechanical construction

The construction chosen by the author is the peak of simplicity. It uses a U-shaped channel section formed from 1.5-mm sheet aluminium, with dimensions of $290 \times 155 \times 85 \text{ mm}$ (Figure 1). All of the preamplifier subassemblies are fitted on or inside this channel sec-

tion. Four 15-mm brass standoffs are fitted to the front, and another four 20-mm standoffs are fitted to the rear. The face panels of the enclosure are attached to the standoffs using M3×5 screws. The end result is that the aluminium channel section is 'suspended' between the front and rear panels.

There is a 15-mm space between the front face of the channel section and the front panel. This provides room for the fixing nuts of the potentiometers and switch and the gearwheels of the balance potentiometers. The LED indicator lamp and the actuating button for the mains switch are fitted directly to the front panel. At the rear, the space is 20 mm deep. This region holds the I/O board (on 15-mm standoffs), the mains filter and IEC appliance receptacle and a small fan.

The power supply is fitted against the back of the channel section, with the transformer stack in the middle, the low-voltage circuit board to the left and the high-voltage circuit board to the right. The two amplifier boards are fitted at the front of the channel section. All of these circuit boards are screwed to 10-mm standoffs, with plastic standoffs being used for the high-voltage board. An aluminium plate with a thickness of 1 mm and a height of 55 mm, with a turned-up lip along one edge, is placed between the power supply



Specifications

Nominal signal level, 'Out' Nominal signal level, 'Line Out'	450 mV (380 mV in) 220 mV (380 mV in)
Input impedance (CD/Tuner/)	l 20 k Ω
THD+N, 'Out' Signal-to-noise ratio, 'Out'	0.1 % (450 mV out) 80 dBA
Bandwidth (maximum volume)	< 10 Hz–35 kHz (–3
Crosstalk	< -65 dB (1 kHz) <-40 dB (20 kHz)
Channel separation	> 76 dB (1 kHz) > 60 dB (20 kHz)
Phono sensitivity (450 mV out) Signal-to-noise ratio, Phono	5 mV (1 kHz) > 53 dB
Bass adjustment (theoretical) Treble adjustment (theoretical)	+18/-9 dB (50 Hz) +9/-10 dB (10 kHz)



Measured response curves:

Chart A shows the frequency spectrum at maximum volume. The distortion primarily consists of the 2^{nd} harmonic at -60 dB, which explains the

value of 0.1 % for THD+N. The fundamental was suppressed for the measurement. The supply-voltage ripple and induced noise from the transformers lie below –90 dB and are negligible.

Chart B shows the frequency response with the tone controls in the neutral, minimum and maximum positions. The actual frequency response may vary due to component tolerances (potentiometers and capacitors).

and the amplifier boards. The lip of this plate is fitted underneath the standoffs for the amplifier boards in order to securely attach it to the channel section. In the prototype, a length of plastic cable duct (the type with a hinged snap cover) was fitted along the top edge of this aluminium plate on the side towards amplifier boards, in order to provide a cableway for several cables. This cable duct can be clearly seen on the photo in **Figure 2**, which also shows the placement of the circuit boards and other components.

Details

Cooling fan

The ventilation openings in the top and bottom covers are not sufficient for the amount of heat to be dissipated. We did not want to deface the enclosure by making large holes in it, so we fitted a small fan at the rear of the enclosure. A series resistor (82 Ω in the prototype) causes the fan to turn quite slowly, and since it is fitted using rubber bushings, it is practically inaudible. The air stream from the fan is directed toward the heat sink of the LM317. This forced airflow through the enclosure also removes the heat generated by the valves.

dB)

After the front and rear panels have been firmly screwed to the standoffs, the fastening holes for the top and bottom covers will quite likely not align exactly with the corresponding holes in the face panels. This is because it is nearly impossible to form the U-shaped channel section to a precise dimension. Consequently, the standoffs at the rear must be lengthened using shim washers until the holes are aligned. This also prevents the fan from being clamped between the two surfaces, so it can do its job without making any noise.

On/off switch

In order to keep the mains wiring (a potential source of interference) as short as possible,



Figure 1. A U-shaped aluminium channel section is suspended between the front and rear panels of the enclosure, using 15-mm standoffs at the front and 20-mm standoffs at the rear.

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Figure 2. A 'bird's-eye view' of the inside of the preamplifier. All of the wiring is in place, the front and rear panels have been fastened to the channel section and the knobs are fitted.

the on/off switch should preferably be fitted as close as possible to the appliance receptacle. A pushbutton switch with a suitable mechanical extension reaching to the front panel is thus recommended. In the prototype, a special construction was used for this purpose, consisting of a rocker switch with a homemade extension arm, but we won't bore you with further details.

Balance control

As already mentioned in the first part of this article, the balance control is constructed using two log-taper potentiometers that are mechanically coupled using gearwheels (see Figure 3). Drill and tap the gearwheels for M3 setscrews. As the material is quite soft, it's a good idea to file flats on the shafts of the potentiometers. This will allow the gearwheels to be securely anchored with only moderate tightening of the setscrews.

Next comes a tip: it's quite easy to equip the balance potentiometers with a tangible midrange position and 'click stops'. This can be done by fitting the shafts of the volume and tone control potentiometers with the same type of gearwheels as



Figure 3. The balance potentiometers are coupled by a pair of gearwheels. The centre-to-centre spacing is 25 mm.

used for the balance control. If you arrange a leaf spring such that it presses a steel ball (from bicycle bearing, for example) against the teeth of the gearwheel, you obtain click stops. A tangible midrange position can be produced by removing one tooth at the midrange position. All of this is clearly shown on the photo in **Figure 3**.

Front panel layout

You are naturally free to label the front panel with text and/or symbols according to your personal taste. For those who prefer a ready-made solution, a front panel layout is available on the *Elektor Electronics* website for download free of charge. It is also ideal for use as a drilling template for the front panel.

Wiring

The aluminium middle plate has four holes with feedthrough bushings, aligned with the circuit board connections for the filament voltage and high voltage, to allow the wiring to pass through the plate. These holes are located at half the height of the plate.

Figure 4 shows how the power wiring of the preamplifier should be fitted. Start by connecting the transformers to the power supply circuit boards, fitting the wiring for the cooling fan and making the mains voltage connections between the IEC appliance receptacle and the switch, fuse and mains filter. Connect the filter terminal to a solder lug screwed to the channel section. Next, make the connections to the filament and high-voltage terminals using twisted-pair wiring passing through the feedthrough bushings. Fit the 'filament' jumpers to the K2 connectors on the circuit boards and connect the filaments of the EF86s in series with a length of wire. This wire is routed through the cable duct. Finally, connect the LED indicator lamp and the input selector switch (S1). Terminal 1 of S1 connects directly to K2/1-2 on the left-hand amplifier board, and then to K2/1-2 on the right-hand amplifier board via a length of flat cable. This six-way flat cable (which is reduced to 5 leads before continuing to the I/O board) also runs through the cable duct.

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Figure 4. Connection diagram for the power wiring.

Figure 5 shows how the signal wiring must be fitted. The volume, treble and bass potentiometers are connected using short lengths of flat cable, while the rest of the connections are made using screened cable. It is convenient to first cut lengths of screened cable and connect them to the I/O board before it is screwed in place, since it is then still readily accessible.

Be particularly careful with the connections for the cable screens. At the Phono In connectors, the screen is connected to the ground point on the circuit board, and on the I/O board the same screen is connected to Gnd L or Gnd R, respectively, which is also the connection point for the screen of the Line In cable. For the output cables, the screens are jumpered to the circuit-board ground points at P5 on each amplifier board. At the balance poten-

tiometers, the screens are connected to the terminals on the left, and on the I/O board they are connected to Gnd Out, which is also the connection point for the screens of the Line Out cables. The three screened cables leading to the left-hand amplifier board run through the cable duct. The 'ground network' is connected to the aluminium channel section at one point only, which is Gnd L on the I/O board.

As can also be seen in Figure 5, the additional DC blocking capacitors Cx and Cy can best be soldered directly to the Line In sockets on the I/O board.

Alignment

Aligning the tone controls is best done by connecting an oscilloscope or AC millivoltmeter to the output. Turn P2 and P4 to the left and right

limits of travel in turn, and at each position, adjust P3 to make the minimum and maximum values for the left and right channels as nearly as possible the same (at 50 Hz and 10 kHz, respectively). Since log-taper potentiometers are used for the tone controls, the midrange position will not precisely coincide with a flat frequency response. In order to find the setting where the response is flat, apply a 1-kHz square wave signal to the input (not the Phono input) and observe the output signal on an oscilloscope. Turn the bass and treble controls until the square wave is as good as possible. With the potentiometers adjusted to achieve this condition, secure the knobs for the tone controls with their arrows pointing to '0'.

To align the balance control, start by setting the right-channel balance potentiometer (to which the knob will be attached) to its midrange position. Then rotate the other potentiometer (but not the gearwheel!) until the output signals of the two amplifiers have equal amplitudes. With the potentiometers in

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Figure 5. Connection diagram for the signal wiring.



this position, secure the gearwheel in place. If any small difference remains, it can be adjusted by slightly rotating the case of the potentiometer.

After the final (output) amplifier is connected, a slight amount of noise and some residual hum will be audible when the volume control is set to maximum and the Phono input is selected. When the volume control is rotated, you will probably hear a weak 'rasping' noise. This can be eliminated by using a small metal spring to electrically connect the shaft of the potentiometer to the enclosure. The hum will also disappear after the upper and lower metal covers of the enclosure have been fitted in place. The two halves of the enclosure can be connected to the central grounding point via lengths of flexible wire.

Connection to the final amplifier

Screened audio cable sometimes has rather high capacitance (as much as 200 pF per metre). If the distance between the preamplifier and the final amplifier is forced to be rather large, it is worthwhile to give some attention to the connecting cables. In this case, you should select cables with low capacitance, so the high frequencies will be affected as little as possible. The author even built cables using RG-59 $75-\Omega$ coaxial cable (diameter 6.2 mm) and 'high-end' Cinch connectors. This type of cable has a capacitance of only 69 pF/m, so it can easily cover distances of up to several metres.

(020383-2)

MECHANICAL COMPONENTS LIST

- Two-section steel enclosure, 300x200x110 mm (wxdxh), Conrad Electronics # 520489*
- IEC appliance socket with internal filter
- Ventilator 40 x 40 mm, 20mm thick, 12V_{DC}
- 2 ABS gearwheels 50.M0.5, Conrad Electronics # 237850*
- -SI = rotary switch, 6 positions, 2 poles, break before make,
- Conrad Electronics 709751* - S2 = mains switch
- -F2 = fuse 250 mAT (slow), with holder
- LED signal lamp, 12 V_{DC}
- 4 black buttons, 21mm
- 4 button caps, red, with line, 21mm
- I button, 28mm
- I button cap, red, 28mm

* www.int.conradcom.de

Valve Preamplifier (I)

Based on the ECL86

Design by G. Haas - Email: experience.electronics@t-online.de

The good old valve amplifier is experiencing a renaissance. Consequently, we present a valve preamplifier that unquestionably belongs to the high end.



In the high-end region, valve amplifiers enjoy uninterrupted popularity. Although a lot of things can be done with modern semiconductors, the old-fashioned valve is still very much in the news in elevated audio circles. In both domestic and studio

equipment, valves are being used increasingly often in equipment such as compressors, equalisers, simple amplifiers, filters and the like. The objective is to achieve a warmer, more attractive sound than what can be obtained using only sterile semiconductor technology. The digital era in particular gives many recordings an unpleasant sharpness, which can be moderated by the knowledgeable and consistent use of valve technology. The valve preamplifier described here usually makes CDs sound more pleasant than they other-



Figure 1. A standard valve preamplifier configuration using one ECC83 for the two channels.

wise would.

This preamplifier is a no-compromise design. Only valves are used in the signal path, while semiconductors are used for the auxiliary functions. In this way, the two technologies complement each other. To be consistent, we have also avoided using semiconductors for switching in the signal path.

The valve determines the design

The choice of the amplification elements, the valves, largely determines the topology of the circuit. Given the limited selection of suitable valves, a few basic designs have come to predominate. However, these all have certain significant disadvantages.

A typical preamplifier, for example, is fitted with ECC81, ECC82, ECC83, ECC88 or similar valves. The ECC83 has a high no-load gain, but only a small working current (1 to 1.5 mA). The ECC81 and ECC82 have lower gain, but they can be operated at currents up to 10 mA. The ECC88 (or the equivalent type PCC88) is most commonly used in television sets, but it is also popular for low-frequency applications, since it can work at currents of up to 15 mA, with an operating voltage of only 90 V.

Connecting two double triodes in series, however, provides far more gain than is needed nowadays, while still not providing an adequate amount of current. In order to reduce the output resistance, a cathode follower circuit is frequently used. This significantly diminishes the dynamic output resistance, although it does not completely eliminate it. A typical cathode follower circuit using an ECC83 is shown in **Figure 1**. The dynamic output resistance R_d is given by the formula:

$$\mathbf{R}_{\mathrm{d}} = \frac{\mathbf{R}_{\mathrm{i}} \cdot \mathbf{R}_{\mathrm{k}}}{\mathbf{R}_{\mathrm{i}} + \mathbf{R}_{\mathrm{k}} \cdot (\mu + 1)}$$

With the following typical values for the ECC83, this yields the following value for R_d :

no-load gain:	$\mu = 100$
internal resistance:	$R_i = 62.5 \text{ k}\Omega$
anode current:	l _a = 1 mA
cathode resistance:	$R_{k} = (47 + 1.5) k\Omega$

$$R_d = \frac{64.5 \cdot 48.5}{62.5 + 48.5 \cdot (100 + 1)}$$



Figure 2. Block diagram of the preamplifier, showing its modular design.





Figure 3. The external connections: the schematic diagram of the relay board with the inputs and outputs.



Figure 4. Here is where the volume and balance are adjusted.

This appears to yield a low output impedance. However, in actual fact the total cathode resistance (48.5 k Ω) or the internal resistance of the valve will determine the effective output resistance of this arrangement (strictly speaking, only if it is overdriven). If we assume that the net capacitance of the circuit and the output cable is only 500 pF, which is easily possible with cables that are a few metres long, the attenuation at 20 kHz is 14 dB. This is why you often see recommendations to limit the cable length to 1.5 metres and to use the lowest-capacitance cable that is available. This certainly helps to reduce the problem, but it does not eliminate its source. The sound of the system will be audibly different when different types of cable are used, and this is a natural consequence of the construction of the amplifier.

Using a single valve for both stereo channels is a mistake that has been inherited from the early days of stereo hi-fi technology, and which has proven to be almost impossible to eradicate. The channel separation suffers, due to capacitive crosstalk within the valve and in wiring of the valve socket, and this considerably impairs the spaciousness and detail resolution of the sound.

Valve amplifiers are often operated without negative feedback. This may not have caused any problems in the days of monophonic sound, but the only way to guarantee proper stereo reproduction is to use strong feedback to achieve equal performance in the two channels of a stereo system, regardless of tolerance variations in the characteristics of the valves. This also causes the distortion factor to remain very low, even when the amplifier is driven hard, and it flattens the frequency response. This also satisfies the demand for the lowest possible distortion in the preamplifier.

For all of these reasons, the preamplifier described here does not follow the conventional path. An ideal amplifier has a high input impedance, a high no-load gain and a low output impedance. These conditions are easily satisfied by operational amplifiers using semiconductor technology. With valves, the situation is far more difficult. In order to avoid the above-mentioned design shortcomings, an ECL86 double valve is used here. The triode section of this valve is exactly the same as that of an ECC83. The pentode section can be used as a power amplifier that can deliver 4 watts with an anode current of 36 mA and a distortion factor of 10 percent. If we properly combine the triode and pentode sections, we can obtain a sort of valve operational amplifier with good characteristics that are similar to those of a modern semiconductor opamp.

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Figure 5. The highlight of the design is the valve amplifier stage.

Modular design

The block diagram of one channel of the preamplifier is shown in **Figure 2**. A modular approach is used in the design, with each functional group located on a separate printed circuit board. The amplification chain consists of four elements, which are the input selec-



Figure 6. The headphone and line output connections.

tor circuit, the volume/balance adjustments, the actual preamplifier and a changeover switch for line or headphone outputs. A protective circuit with a time delay switches the outputs to earth when a fault is detected in the output signal. An external power supply is used. The modular construction not only allows the circuit to be modified relatively easily, it also promises very good values for channel separation and the signal-to-noise ratio. This makes up for the increased cost and effort for the wiring.

In the following descriptions of the individual modules, the component numbers correspond to the labels on the circuit boards. In contrast to the usual practice with Elector projects, the components are not numbered sequentially for the complete circuit, but instead separately for each module. Only one channel of the amplification chain is shown; the component numbers for the second channel are either shown in parentheses or are distinguished by a 'prime' mark (').

Input selection with relays

The input selection module shown in **Figure 3** is laid out for four signal sources K1 through K4 (K6 through K9) that are connected to a single common line via the single-in-line reed relays Re1 through Re4 (Re5 through Re8). The consistent use of screening and the split layout of the printed circuit board provide a high degree of channel separation from the very start.

The 100-k Ω resistors R1 through R4 (R5 through R8) terminate the input sockets and conduct any static charges to earth. They prevent clicks when a different input source is selected or the input cables are reconnected. The resistors labelled Rx can be used with signal sources that have different output levels; their values can be chosen as necessary. However, you should make sure that the series resistance for K5 (K7) matches the source resistance for the tape recorder output. If the resistance is too large, high tones will be lost.

The 12-V relays, which are powered by the +12.6-V DC filament supply voltage, are switched by the

rotary switch S1. This is connected to the circuit board via solder posts.

The line out sockets K13 (K15) and the earth terminals K14 (K16) are located on the same circuit board, but they are fully electrically isolated from the rest of the board. The tape recorder output sockets must be mounted externally, and then connected to solder posts K5 and K7 (earth to K17).

Volume adjustment

From connector K5 (K7) of the relay board, the signal is conducted to the circuit shown in Figure 4 for adjusting the volume, balance and muting. The potentiometers that are used, with 41 detent positions, have good reproducibility. The balance adjustment also has a fixed midpoint setting. The mute switch S1 spreads the adjustment range, so that even low volume levels can be sensitively adjusted. If you find the 20 dB step unsuitable, you can modify the values of the resistors. The sum of R2 (3.9 k Ω) and Re (470 Ω) should remain approximately the same, but the individual values can be modified as desired. When the mute switch is in the middle position, the signal path is interrupted, and the output is muted.

The characteristic curve of the linear volume adjustment potentiometer P1 (which has good tracking due to its linearity) is slightly modified by inserting R1 between its lower terminal and earth. This compensates for the slight imbalance between the two channels in the initial region of the volume adjustment, which arises from the construction of the preamplifier. Of course, with such an arrangement it is not possible to set the volume level fully to zero.

The balance control P2, in combination with R4 and R5, has an adjustment range from around +3 dB to -4 dB. This is fully adequate for correcting the midpoint. With the indicated component values and the 100-k Ω resistors on the relay board, at the middle position of the balance control the input resistance is approximately 2.2 k Ω . This is a suitable value for all modern signal sources, and it is low enough to keep the disturbance sensitivity and the total noise at low levels.



Figure 7. The protective circuitry monitors the state of the preamplifier and switches the output relays.

The amplifier

The amplifier module shown in Fig**ure 5** makes a familiar impression. Valve V1a is the triode section of the ECL86, equivalent to half of an ECC83. The DC operating point is set to just below 1 mA by R2. The amplified signal reaches the pentode section V1b via capacitor C4. Neither section of the valve is wired in the conventional manner. The cathode and the grid bleeder resistor are not connected directly to earth, as is usual, but instead via resistors R12 and R3, respectively. This provides local negative feedback for each valve section. This constrains the valves to work at well-defined operating points, so it is not necessary to use selected valves

The load resistance of V1b consists of R15 and R16 in parallel (each 6.8 k Ω , 4.5 W). This divides the not insignificant power dissipation equally over two component packages. The signal is coupled out via two parallel 22- μ F capacitors (C7 and C8). Using two capacitors instead of a single 47- μ F capacitor

halves the ESR (effective series-resonant resistance) of the output capacitor, which benefits the high frequency transfer characteristic of the amplifier. Resistor R13 bleeds off static charges.

To ensure that the circuit has a welldefined behaviour, the output signal is coupled back via C5 and R4 in negative feedback. The basic amplification of V1a depends on the load resistance of R6, the operating voltage (the voltage across C2) and the cathode circuitry. Resistors R2 and R5 are connected in parallel for AC signals. The amplification is thus determined by the relationship of the parallel combination of R2 and R5 to the resistance of R4. If you want to modify the basic amplification, you can change the values of R5 or R4. The smaller the resistance of R4, or the larger the resistance of R5, the higher the gain. Capacitor C5 separates the AC and DC currents, so that the DC operating point of V1a does not shift. Resistor R2 may not be modified to change the gain, since this would shift the DC operating point in an unacceptable manner.

The supply voltage, U_B , is 330 V. It is decoupled and filtered by R14 and C6, in order to avoid channel crosstalk via the supply voltage. The supply voltage is further decou-



Figure 8. The power supply is a combination of a high voltage supply, a low voltage supply and an optional third supply.

pled and filtered for V1a by R8 and C2. Resistor R7 is a bleeder resistor that discharges the high-voltage electrolytic capacitors after the supply voltage has been switched off.

Options

The input resistor R can be omitted, since the volume and balance adjustments also shunt static charges. However, if the amplifier module is used in some other application, any desired value can be used as necessary. The indicated connection between R9 and the anode of R1b should only be used if you want

to operate the pentode in quasi-triode mode. In this case the combination of R15 and R16 is shorted out, and R9 is naturally redundant.

Line or headphones

Using the ECL86 as a preamplifier valve has the additional advantage that the pentode section can deliver 4 watts of output power. This capability can be used to advantage by connecting a switched phone socket (K1) between the amplifier output and the line output. When a headphones plug is inserted into this socket, it automatically disables the line outputs. The pentode section of the ECL86, which can deliver a relatively large current, can easily drive headphones with an impedance greater than 300 Ω , even over a relatively long screened cable, without degradation of the sound quality. Normal screened cables are fully adequate for this purpose.

The portion of the circuit shown in **Figure 6** also includes two relays, which are driven by the protective circuitry. The two resistors bleed static charges to earth.

Unconditional protection

The protective circuitry shown in **Figure 7** performs several tasks at the same time. Three conditions must be satisfied before the preamplifier is switched 'online'.

When the amplifier is first switched on, the outputs are initially shorted out via the relay outputs. The timer output of IC1 (pin 3) goes Low after approximately 100 s, at which time the valves should be properly warmed up and all charging processes completed. This delay prevents humming, burbling or other upsetting sounds from being generated by the speakers while the preamplifier is heating up. The output of inverter IC2b then sets input 2 of NAND gate IC2a High.

If a voltage is then present at pins AC1 and AC2, which are connected directly to the filament winding – diodes D1 and D3 monitor both half cycles of the AC waveform – then C1 is charged via R1 to 12 V (limited by D2). A High level is thus applied to input 1 of IC2a.

The DC components of the two output signals are monitored via connections DC1 and DC2. A DC voltage can for example be present if an output coupling capacitor breaks down. The AC components are shorted out by C2, which is discharged by R7. This combination also determines the time constant of the monitoring circuit. If a DC voltage higher than around 1.3 V is present, transistor T1 is driven via D5 and R4. This pulls input 8 of gate IC2a Low. A Schottky diode with a very low forward voltage drop is used to ensure that the monitor circuit will respond, even with very low values of the DC component. Transistor T1 is cut off only when no significant DC component is present, so that R9 can pull input 8 of gate IC2a high. Resistor R13 ensures that T1 is securely cut off in this situation. Diode D6 limits the maximum voltage applied to the base of T1, while R4 and C5 provide a short time delay, so that the circuit does not immediately respond to every tiny disturbance.

Only when all three of these conditions have been satisfied, so that all three inputs of the AND gate are High, will the protective circuit enable the output relays via inverter IC2c and the driver transistor T2.

When the mains transformer is switched off, the level at input 1 of the gate goes Low almost immediately. This causes the relays to be immediately disabled, so that no disturbance signals can be passed on to the following equipment while the capacitors in the preamplifier are discharging. The reset circuit consisting of R2, D4 and C1 also resets timer IC1 after a brief interruption of the mains power, so that the full delay time must expire each time. The delay time can be changed by modifying the values of R10 and C6. The time is equal to 1.1.RC. The protective circuitry is operated from the filament voltage power supply, with a voltage of 12.6 V.

One transformer, three power supplies

A good power supply is essential for the proper operation of a preamplifier. Since the amplifier draws relatively little current, a cleanly filtered and stabilised power supply can be built economically using modern semiconductors. **Figure 8** shows the schematic diagram of the entire power supply circuit. This is also built using several separate circuit boards, which are indicated by dashed outlines.

The preamplifier needs a high voltage for the valves, as well as a low DC voltage for the filaments, the relays and the protective circuitry. These voltages, as well as some others, are provided by a single mains transformer (NTR-10B) that is protected by a 0.4-A slow-blow fuse on

the primary side. This transformer is made using grain-oriented 0.35-mm steel laminations with especially low dispersion and losses, which are usually used for high-quality low-frequency coupling transformers. A very orderly winding technique and vacuum impregnation, which is not possible with toroidal transformers, provide long-term stability and corrosion resistance. The impregnating resin penetrates into the coils and permanently anchors each individual winding. Electrical safety is provided by a 4000-V breakdown test between the primary and secondary windings and a static screen that is connected to the protective earth lead. Obviously, you will not find this sort of high-end transformer in every electronics supermarket. It is only available from Experience Electronics, the author's firm.

At the top of the schematic diagram, you can see the high voltage section. This circuit filters the hum voltage to a level that is below the intrinsic noise level. Resistor R1 and diodes D1 through D3 generate a good reference voltage with a value of around 330 V. Series regulator transistor T1 is a high-power V-FET (type BUZ92) in a TO-220 package. This allows the high voltage power supply to be constructed in a compact manner, using a small heat sink. The heat sink is mounted on the printed circuit board, with makes for short leads in the high voltage path. Resistors R4 and R5 and transistor T2 provide current limiting. With the indicated component values, this is set to around 90 mA. Resistor R6 discharges the electrolytic capacitors after the power is switched off. Zener diode D4 limits the gate voltage of T1, which must not exceed 20 V.

In order to avoid hum voltages, the filaments are powered by a DC supply. The frequently heard assertion that using a DC current for the filaments damages the valves is simply nonsense. All that is necessary is to bring the filaments to a particular temperature, so that the cathode can emit enough electrons. Whether this temperature is achieved using an AC current or a DC current does not matter. The 'odd' value of 6.3 V for the filament voltage, by the way, comes from the early days of valve technology, when the filaments were powered by four carbon-zinc batteries. Since fresh batteries have a terminal voltage of more than 1.5 V, a margin of 0.3 V was chosen to prevent the filaments of the (at that time) very expensive valves from burning out prematurely.

Each of the two ECL86 valve filaments draws 0.66 A at 6.3 V. To minimise the losses in the filament circuit, the two filaments are connected in series. A 12-V voltage regulator connected to earth via a 1N4148 forms a simple but good 12.6-V power supply for the filaments. The relays for the input selector are also powered by this supply. The aluminium mounting plate is used as a heat sink for the voltage regulator, which must be mounted using an insulator.

At the bottom of the circuit diagram you can see a third power supply that provides symmetrical ± 15 V. This supply is not necessary, so it is optional. It can be used to power external devices, such as an equalisation preamplifier, from a non-earthed supply. If such a supply is already present in the preamplifier, you save the cost of a mains adapter and the associated cabling, and the external equipment can be switched on and off at the same time as the preamplifier. The aluminium mounting bracket is also used as a heat sink for this supply, which must be isolated. The supply voltage has additional protection in the form of two fuses in the transformer leads.

(000063-1)

The second part of this article will include the printed circuit board and component layouts, the components list, precise construction information and, of course, performance data.

Valve Preamplifier (2)

Construction, first part

Design by G. Haas

You should use only the best components to build a high-end amplifier, if you want to achieve and maintain the specified performance figures.



It is also important that you observe the suggestions given in the text and the components lists, and avoid using inferior-quality 'equivalent' types. All printed circuit boards sup-

Please Note, corrections to part I

The dashed connection between R9 and the anode of valve VIb is not an optional wire link. As indicated by the PCB component mounting plan, resistor R9 may be fitted in one of two positions marked R9 and R9*. In pentode mode (default), the resistor is fitted in position R9, i.e., connecting the suppressor grid (g2) to the +324 V anode voltage. This configuration is shown in the circuit diagram, Figure 5. The dashed connection has no meaning. The quasi-pentode configuration requires the resistor to be fitted in position R9*, i.e., between the suppressor grid and the anode.

Figure 6 does not show two backemf suppression diodes across the relay coils. These diodes are however present on the board, as well as included in the parts list.

In Figure 7, R10 should be 390 k Ω (not 100 k Ω), and C6 should be 220 μ F (not 1000 μ F).





COMPONENTS LIST Input Board

Resistors:

 $\label{eq:rescaled} \begin{array}{l} {\sf R1}{\sf -}{\sf R8} \,=\, 100 {\sf k}\Omega \\ {\sf Rx} \,=\, {\sf see \ text} \\ {\sf Re1}{\sf -}{\sf Re8} \,=\, {\sf SIL} \ {\sf reed \ relay, \ 12V, \ 1 \ make} \\ {\sf contact} \end{array}$

Miscellaneous:

K1-K9 = cinch sockets, chassis mount
Two cinch sockets, chassis mount, for recorder outputs
S1 = rotary switch, 1 pole, 4 contacts, break before make
Solder pins
Copper foil



Figure 1. Relay board: printed circuit board track layout, component layout and photographi.

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COMPONENTS LIST

Amplifier Board (one channel)

Resistors:

$RI = 680 k\Omega$
$R2 = Ik\Omega 8$
$R3 = 10k\Omega$
$R4 = 33k\Omega$
$R5 = 2k\Omega 2$
$R6 = 150k\Omega$, 2W
$R7 = 470k\Omega$, 2W
$R8 = 2k\Omega7$

$R9 = 8k\Omega 2$ $R10 = 680k\Omega$ $R11 = 150\Omega$ $R12 = 270\Omega$ $R13 = 27k\Omega$ $R14 = 150\Omega, 2W$ $R15,R16 = 6k\Omega 8, 4.5W$

Capacitors:

CI = $I\mu$ F 63V, 5mm raster C2 = $I0\mu$ F 400V, 5mm raster C3 = 47μ F 40V, 5mm raster C4 = 2μ F2 400V, 5mm raster plied by Experience Electronics for this project are made using fibreglass-reinforced epoxy with a 70- μ m copper lamination. All resistors in the components list are 1-percent tolerance, 0.7-W metal film types, unless otherwise indicated, with a lead spacing of 10 mm. The 2-W and 4.5-W resistors are metal oxide types with a tolerance of 5 percent, and have lead spacings of 15 mm and 25 mm respectively. The gold-plated Cinch sockets, as well as the potentiometers, should of course be very high quality.

Before we discuss the construction of the circuit boards, a few words about the enclosure are in order. Even the best electronics, no matter how clever the design may be, is of no use if it is not housed in a suitable enclosure. With valve

C5 = 220μ F 40V, 5mm raster C6 = 47μ F 450V, size 18.5x41mm C7,C8 = 22μ F 400V, 7.5mm raster C9 = see text

Miscellaneous:

DI = see text VI (RöI) = ECL86 I ceramic 'Noval' (9-pin) socket, PCB mount Copper foil Solder pins



amplifiers in particular, given their high working voltages, electrical safety is a primary consideration! A metal enclosure, connected to the protective earth lead, provides both safety and screening. If the enclosure also has an attractive appearance, there will be nothing to disturb your listening pleasure.

All circuit modules are housed in an aluminium enclosure. The advantage of aluminium is that it is nonmagnetic, so that it avoids magnetic distortions. In addition, it has very good design properties. The enclosure used here has rivitless joints, and the surface is polished and covered by bright nickel plating. Nickel has a warm tone, in contrast to the bluish tone of chrome, and this optically reinforces the 'flair' of a valve amplifier. In order to avoid having an

Elektor Electronics



excessive number of screws visible on the top surface of the enclosure, a mounting plate is used to fit all of the electronics. This plate is then screwed to the chassis using eight bright-nickel-plated Phillips screws. In this way, you can produce a visually attractive piece of equipment.

Board stuffing

Now you can start with stuffing the boards. Start with the relay board, as shown in **Fig-ure 1**. The Cinch sockets are screwed to the



Figure 2. The amplifier board is laid out symmetrically, with the two valves in the middle.

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board. You must first tin the tracks at the points of contact, to assure good connections. Apply a drop of solder to each fastening nut to secure it. This guarantees long-term, reliable earth connections. If there is contact resistance in the earth connection at this location, interference is an unavoidable consequence. The photograph shows the construction of the circuit board and the placement of the components. Strips of 0.15-mm thick copper foil are soldered between the sockets. These provide screening on both sides, and between the two channels. In this regard, we can remark that in many modern pieces of equipment, you will find pairs of Cinch sockets in plastic modules that can be screwed to the rear panel of the enclosure, with pins that can be soldered directly to the circuit board. These may be less costly for the manufacturer, and they are easy to mount, but using such modules here will impair the channel separation of the entire device, due to the close spacing of the sockets and the lack of screening.

Since soldering the nuts requires a lot of heat, this should be done first, before any other components are fitted. After this, you can insert the resistors, diodes, relays and solder posts.

The main circuit board with the amplifier, as shown in **Figure 2**, is laid out so that it can be used for other applications by adding or omitting components. Use a wire bridge in place of zener diode D1, which is marked with a star on the circuit board. This diode is only used as an option for other applications, in which the amplifier circuit is operated from a higher supply voltage. The good qualities of a stabilised power supply are retained by the low impedance of the zener diode. A highvalued series resistor would only degrade the quality of the supply voltage.

You should pay attention to certain details when inserting the components in the circuit board. In order to make the valves protrude nicely from the chassis, so that they are readily visible, the valve sockets are mounted on the solder side of the circuit board. This also ensures good heat dissipation. The socket pin assignments are shown in Figure 3. Power resistors R15 and R16, which become quite warm, are also mounted on the solder side, separated from the board by a certain amount, in order to improve their heat dissipation. It also doesn't hurt to make a series of ventilation holes in the circuit board in unused areas. This avoids the creation of heat pockets. All other components are mounted on the component side, as usual. The board is mounted to the chassis using suitable spacers.

There is not a lot to say about fitting the



Figure 3. Pin assignments of the ECL86, viewing the pins from the bottom.

- K_p pentode cathode
- s internal shield
- g_T triode grid
- g₂ pentode screen grid
- g₁ pentode signal grid

- pentode suppressor grid
- filament

g3

- k_T triode cathode
- a_p pentode anode
- a_T triode anode





Figure 4. Printed circuit boards for the protective circuitry.

components to the protection circuit board (**Figure 4**). Sockets may be used for the DIL ICs. If you pay attention (as always) to the correct locations and polarisations of all components, everything will be OK.

Note:

Kits, PCBs and parts for the Valve Preamplifier are available from Experience Electronics, Weststrasse I, D-8942 Herbrechtingen, Germany. Tel. (+49) 7324 5318, fax (+49) 7324 2553. Email: experience.electronics@t-online.de





COMPONENTS LIST Protection Circuit

Resistors:

 $\begin{array}{l} {\sf R1}, {\sf R2} = 10 {\sf k\Omega} \\ {\sf R3} = 33 {\sf k\Omega} \\ {\sf R4} = 1 {\sf k\Omega} \\ {\sf R5}, {\sf 6} = 33 {\sf k\Omega} \\ {\sf R7} = 100 {\sf k\Omega} \\ {\sf R8} = 10 {\sf \Omega} \\ {\sf R9} = 10 {\sf k\Omega} \\ {\sf R10} = 390 {\sf k\Omega} \\ {\sf R11} = 100 {\sf k\Omega} \\ {\sf R12} = 10 {\sf k\Omega} \end{array}$

$RI3 = I00k\Omega$

Capacitors:

 $C1 = 0.22\mu F MKT, raster 7.5mm$ $C2 = 0.33\mu F MKT, raster 7.5mm$ C3 = 10nF ceramic, raster 5mm $C4,C5 = 1\mu F 63V, raster 5mm$ $C6 = 220\mu F 40V, raster 5mm$ $C7 = 47\mu F 40V, raster 5mm$

Semiconductors:

D1,D3 = 1N4007 D2,D6 = zener diode 12 V, 1.3W D4,D7 = 1N4148 D5 = BAT43 TI = BC546B T2 = BD139-16 ICI = 555 IC2 = 4023

Miscellaneous:

- I off 8-way DIL IC socket, gold-plated contacts
- I off I4-way DIL IC socket, gold-plated contacts
- Solder pins

Valve Preamplifier (2a)

Part 2a: Construction (continued) and test data

By G. Haas

Once you've stuffed all the boards that go into the design, the Valve Preamplifier is ready for assembling and wiring. The results (see pictures and performance data) are pretty convincing.





Figure 2. The small printed circuit board for the volume and balance controls.



Next comes the volume control board (see **Figure 2**). The photograph shows how it is built. There is nothing difficult about mounting the components, and the board can be solidly attached to the front panel via the potentiometers, using suitable hardware.

The output relays and the headphone socket are mounted on the small circuit board shown in **Figure 5**. The socket is also used to fix the board to the chassis, so no additional fitting hardware is needed. You must pay careful attention to where the signal from the amplifier comes from and where is it goes to afterwards. The headphone socket is protected only if the wiring is done properly. In addition, make sure that

COMPONENTS LIST Volume Control

Resistors:

R1,R1' = 100 Ω R2,R2',R4,R4' = 3k Ω 9 R3,R3' = 470 Ω R5,R5' = 10k Ω P1 = 10k Ω stereo potentiometer, linear, tracking P2 = 10k Ω stereo potentiometer, linear, tracking 2 holders for potentiometers

Miscellaneous:

SI = rocker switch, 2 poles, 3 contacts Solder pins





COMPONENTS LIST

High Voltage Supply

Resistors:

 $\begin{array}{l} {\sf R1} = 10 k \Omega, 2 W \\ {\sf R2} = 100 \Omega, 4.5 W \\ {\sf R3}, {\sf R4} = 1 k \Omega \\ {\sf R5} = 6 \Omega 8, 2 W \\ {\sf R6} = 150 k \Omega, 2 W \end{array}$

Capacitors:

 $CI = 100\mu F$ 400V, raster 10mm $C2,C3 = 22\mu F$ 400V, raster 7.5mm

Semiconductors:

TI = BUZ92 T2 = BC546 DI,D2,D3 = zener diode 110V, 1.3W D4 = zener diode 18V, 1.3W GII = B500C1500, rectangular case(500V piv, 1.5A peak)

Miscellaneous:

Heatsink for T1: SK 68/50 (Fischer, Dau Components) Isolation and mounting material for T1 and heatsink Solder pins







Figure 5. The small printed circuit board for the output relays and headphone socket.

COMPONENTS LIST Headphones/Line Switch

 $\label{eq:K1} \begin{array}{l} \mathsf{K1} = \mathsf{stereo} \ \mathsf{headphones} \ \mathsf{socket}, \ \mathsf{PCB} \\ \mathsf{mount} \\ \mathsf{Re1}, \mathsf{Re2} = \mathsf{relay}, \ \mathsf{I} \ \mathsf{changeover} \ \mathsf{contact}, \ \mathsf{I2V} \\ \mathsf{coil} \ (\mathsf{Omron} \ \mathsf{G6E}) \\ \mathsf{R1}, \mathsf{R2} = \mathsf{680k} \Omega \end{array}$



Figure 7. the high-voltage power supply and ...







Figure 8. the low-voltage power supply.

COMPONENTS LIST Low Voltage Supply

Capacitors:

 $CI = 2200\mu F 40V, raster 7.5mm$ $C2 = 100\mu F 40V, raster 5mm$ C3,C4 = 100nF, ceramic, raster 5mm

Semiconductors:

GII = B80C1500, rectangular case (80V piv, I.5A peak) DI = IN4148 ICI = 7812

Miscellaneous:

Isolation material for IC1 Si1,Si2 = fuse, 0.2 A, slow, with PCB mount holder Solder pins

Figure 9. The wiring of the valve preamplifier is complicated.

the relays are correctly polarised, as shown on the component overlay. The selected relays are fully enclosed, which protects the contacts from contamination. In addition, the contacts are gold plated and rated for 250-V operation.

There is no circuit board available for the optional symmetric power supply. However, the rich store of *Elektor Electronics* projects should provide adequate possibilities.

Now that you have finished building all the circuit boards, they can be fitted into the enclosure. If you use the original enclosure, as described above, all of the necessary holes are pre-drilled, so all you have to do is to fit the hardware for the individual parts (including the mains cable input socket, the power switch, the potentiometers and the recorder sockets). If you prepare your own enclosure, take care to arrange the individual items in a favourable manner and ensure that the circuit boards are solidly attached. Don't fit the input module right next to the mains transformer, since otherwise you can be sure of introducing mains hum into the audio signals.

Internal wiring

Installing the wiring that interconnects the various circuit boards is probably the most complicated task of all. Consequently, we have prepared the wiring diagram shown in **Figure 9**. You should photocopy this,



and mark each connection on the copied diagram once you have made the connection. This manner of working has proven to be the most foolproof, including in the *Elektor Electronics* labs.

All leads that carry audio signals must be made using good-quality screened cable. Ordinary light-duly multi-strand wire can be used for the remaining leads, which consist of the signal source selector switch wiring, the protection circuit wiring and the power supply wiring. The wire should have a diameter of 1 mm and good insulation. Place the signal wiring and the other wiring so that the two types are separated from





each other as far as possible, and fix the cables to the chassis using clips. Connect the recorder output sockets to the common busses for the other circuit boards using wire bridges.

The moment of truth

When you have finished the wiring and everything has been checked and rechecked, you are ready to try the first functional test. First, break the connection between the mains transformer and the high-voltage circuit board (and insulate it against accidental contact). The filament supply must measure 12.6 V as soon as it is switched on, and it must be short-circuit proof. This voltage may vary by up to ± 5 percent. The valve filaments must glow visibly after around one to two minutes. Later on, the filaments will be the 'pilot light' for the preamplifier. However, if you want an additional, more distinct power-on indicator, you can simply connect an LED in series with a resistor and 1N4007 diode to the 15-V winding of the mains transformer. You can also right away check the operation of the relays.

After this initial test, you can activate the high voltage. It should reach its nominal value shortly after being switched on. If nothing smokes, check the expected values noted on

the schematic diagram. After this, use a sinewave generator and oscilloscope to check the audio paths and functions. Once this test has been successfully completed, close up the enclosure, connect the preamplifier to signal sources and a final amplifier, and switch everything on. A valve cathode must warm up for two to five minutes before it can emit enough electrons for the valve to be operational. However, you will have to wait around ten to twenty minutes for the valve to be thoroughly warmed up, before you will hear the 'right' sound. After this, all you have to do is relax and enjoy what you hear!

(000063-2a)

Measured performance

We made our own measurements of the performance of the valve preamplifier in the Elektor laboratories. Naturally, we would like to share the results with you. The 'raw' numbers are listed in the table. All measurements were made after a four-hour warm-up interval, with an effective input voltage of 1 V and an effective output voltage of 2 V. The outputs were terminated in 10 k Ω , and free inputs were terminated in 600 Ω . The balance potentiometer was in the middle position.

Signal to noise ratio	A-weighted	100 dBA
	linear, 22 Hz – 22 kHz	87 dB
THD plus noise	l kHz, BW = 80 kHz	<0.05 %
	l kHz, BW = 22 Hz – 80 kHz	<0.01 %
	l kHz, BW = 400 Hz – 22 kHz	<0.05 %
IMD	(50 Hz : 7 kHz = 4:1)	<0.02 %
DIM	(3.15 kHz square wave	
	+ 15 kHz sinusoid)	0.003 %
Channel separation	l kHz	84 dB
	20 kHz	63 dB
Crosstalk	l kHz	<-115 dB
	20 kHz	<-93 dB
Input impedance	Minimum volume	6.4 kΩ
	Maximum volume	2 .1 kΩ
Output impedance		<200 Ω
Amplification factor		2.54
Bandwidth		3.5 Hz - 500 kH
Balance adjustment range		+3 to - 4.7 dB
Attenuator		18 dB
THD	Output voltage = $50 V_{eff}$	0.1 %

It must be noted that the measurements were made without any screening of the valves, so RF disturbances from the measurement environment (Elektor lab with PC-based measurement equipment) may have influenced the measured results. If RF interference sources are present in the vicinity of the preamplifier, screens should be provided for the valves.

The five measured performance curves show the following:

A) Amplitude response

On the strongly enlarged scale up to 200 kHz, (the upper performance limit of our audio generator), a slight rise in the amplitude response can be seen. Within the 'interesting' part of the audio frequency band, the curve is dead straight!

B) Channel separation

The curves are self-explanatory; the two channels track each other very nicely. These curves start at 200 Hz, in order to eliminate the effect of power supply ripple on the measurements.

C) Frequency spectrum

The effect of power supply ripple can be seen in the frequency spectrum. The spectrum of this ripple reaches to around 800 Hz. This explains the relatively large difference between the A-weighted and linear measurements (100 dBA and 87 dB, respectively). With the I-kHz tone (I V_{eff}), essentially only the second harmonic is visible, reaching up to -90 dB. The influence of interference signals, which in this case come from some old monitors located near the amplifier, can be seen with the unscreened valves as peaks in the high-frequency region of the spectrum (30 kHz and 60 kHz).

D) THD plus noise

This curve was obtained with a bandwidth of 22 Hz to 80 kHz. The interference signal components come from the effects of power supply ripple, and probably also from induced signals radiated by the transformer.

E) Step response

A squarewave signal at the input (10 kHz, 1 V) produces a small overshoot (around 10 percent) at the output.





Valves at Low Plate Voltages

Interesting and surprising experiments with valves

by B. Kainka

Is it just nostalgia, or are valves really somehow better than transistors? Recently valves have been making something of a comeback in many areas. Using valves seems to involve a lot of effort, and in particular the high voltages frighten many people off. But there are dozens of valves lying about in many a cellar — so why not try something new with those old valves?

Valves are usually driven using an anode voltage of 250 V or more; practically never with an anode voltage below 100 V. For power amplifiers, in particular for radio transmitters, several kilovolts can be used. Such inconveniently high voltages naturally put many people off, as do the special transformers and high-voltage electrolytic capacitors that are needed. But things need not be like this. A series of experiments has shown that most of those valves nostalgically kept at the back of the cupboard will work at very low voltages. Of course, we are not talking about achieving the ultimate in power or amplification, but for simple applications — and a bit of fun — it fills the bill.

We will describe in this article how to build simple circuits using valves with a minimum of fuss. Operating at anode voltages of, for example, 12 V is not recommended by manufacturers and is not covered in any data sheet. So, if we are going to learn anything, we will need to experiment and make some measurements.

In order to forestall criticism from committed valve-lovers, we should say that the aim here is not to build the last word in hi-fi amplifiers or find the optimal operating point for some particular valve. We are more interested in gaining some experience with valves in a simple and safe way. There is a special satis-



faction in building a small circuit and making a simple working device. And it is not just about feeling the warm glow from the cathode: it is like going back in time to the early days of electronics, when the (relatively simple) technology was dominated by amateurs and everything was visible. Valves in their glass envelopes are certainly more 'transparent' than ICs in plastic packages. Of course, we could do things 'properly', and use anode voltages of 250 V; but that would not exactly make for simple and relaxing experimentation on the bench. A chassis would be required and everything would have to be built carefully into a case. And we would always have to watch out for those dangerous voltages. None of this is a problem if we stick to low voltages.



Types of valve

The question frequently arises: 'are valves still being made?' The answer is yes. Out of the enormous range of valves that used to be available, a few are still around, being made by several manufacturers. As well as valves still used in radio transmitters there are a few types specially designed for hi-fi amplifiers. Output stage valves such as the EL84 (6BQ5) and EL34 (6CA7), and ECC81 (12AT7), ECC82 (12AU7) and ECC83 (12AX7) double triodes are readily obtainable today, albeit at rather higher prices than a few decades ago.

Other sources of unused valves are mail-order suppliers such as Chelmer Valve Co., who offer a particularly good stock of European, US and Russian valves at reasonable prices. Especially interesting are the numerous types of miniature valves, so-called 'battery valves', which are designed to run on low voltages.

However, we are not limited to using new valves. Devices salvaged from the cellar will generally be old radio or television valves. In the good old days radios and televisions that were beyond repair were often cannibalised for their valves. In televisions we generally find P-series valves, designed to have their heaters wired in series, with a heater current of 300 mA, as well as the ECC81 and ECC82 types mentioned above, and any number of EF80s (6BX6). From radios we can obtain interesting devices such as EL84s and EL95s (6DL5) in output amplifiers. The E-series devices require a heater voltage of 6.3 V. All these devices can be brought back to life, and can even be used at low anode voltages.

Indeed, there are so many different types of valve that we do not have space in this article to cover the technical details and socket pinouts for all of them. All the necessary information can, however, be readily found on the Internet. There are also sites that give example circuits and hobby projects alongside such data. Even the use of low anode voltages is mentioned here and there, and it is becoming something of a hobby in itself. So, for example, the hobby corner of the author's website (<u>www.b-kainka.de</u>) includes example circuits from a 12 V headphone amplifier using two transmitter valves to an Audion-type receiver. There are also many links to similar projects.

The ECC81 (12AT7)

The ECC81 is readily available both new and second-hand. From the double triode series ECC81/ECC82/ECC83, which all have the same pinout, the ECC82 is also suitable, but the ECC83 has too low an anode current at low voltages. The ECC81 was originally intended for HF applications and sweep circuits in televisions and oscilloscopes. They are therefore capable of operation at high frequencies, and best performance is obtained with anode currents of between 5 mA and 10 mA. For HF applications the valve is often found in a cascode configuration, where the two triodes are in series and therefore share the available anode voltage. This is why the ECC81 has adequate anode current and transconductance at low voltages.

Once one has obtained a used valve, the first question is naturally whether it works or not. This does not require a complete circuit to be built: a couple of simple experiments can be carried out on the bench with the aid of crocodile clips. First a heater voltage must be applied. Figure 1 shows the socket pinout of the ECC81. Almost all valves with a nine-pin 'noval' base have heater connections on pins 4 and 5. The ECC81, ECC82, and ECC83 are a bit special, however: the heater element has a centre tap on pin 9. This means that the valve can be used with a heater voltage of 12.6 V (with a current of 150 mA) or with a heater voltage of 6.3 V (at 300 mA). This is very convenient for our purposes, since we can use 12.6 V (12.0 V will also do!) for both the heater voltage and the anode voltage.

First connect the heater voltage of 12 V to pins 4 and 5. After about half a minute the cathode will start to glow. If no current flows, the valve is probably burnt out. This case is relatively rare. More frequently, the valve is not burnt out but rather badly aged and will have rather poor char-



Figure 1. Pinout of the ECC81.



Figure 2. Measuring the negative charge on the grid.

acteristics. For simple experimental purposes, however, it will probably be perfectly usable.

The second thing to test is whether the vacuum in the valve is still hard. Connect a voltmeter between cathode and grid (see Figure 2). If all is well, a voltage of approximately –0.5 V should appear on the grid (assuming the voltmeter has an input resistance of 1 M Ω). This is already showing the effect of free electrons. The hot cathode ejects electrons into the free space around it, and some land on the grid, giving it a negative charge. If, instead of measuring the open-circuit voltage, the short-circuit current is measured, a value of around 20 μ A will be found.

This effect is used in many circuits to automatically create a negative grid voltage, including in the headphone amplifier described below.

Whether the vacuum is still hard can often be determined by inspection. The ECC81 has a silver-coloured speck at its end, called the 'getter flash'. When the valve is manufactured the air is pumped out of it through a glass tube and then it is sealed. In the end of the valve there is a ring-shaped groove which is filled with a metal having a low melting point. This metal is heated through the glass using a powerful HF magnetic field: this evaporates the metal onto the inner surface of the glass below. The result of this whole process is to permanently trap the last



Figure 3. A simple headphone amplifier.



Figure 4. Using an output matching transformer.



Figure 5. Input and output signal voltages displayed on an oscilloscope.

remaining gas molecules in the envelope. If, after years or even decades, the getter flash is still bright, then all is well; if it is white or grey, it means that air has leaked in and the getter metal has oxidised.

Headphone amplifier

If the cathode glows, a grid current can be measured, and the getter metal is still bright, the valve is in good order and you can use it to build this simple headphone amplifier. The circuit is presented here in two variations. The first circuit, in **Figure 3**, requires just two components per channel in addition to the valve and headphones.

The anode current of 0.17 mA measured experimentally in a used valve is relatively low. In specially-designed low-voltage valves the current was somewhat higher: for comparison, the low-voltage ECC86 (6GM8 or CV5394)) has an anode current of 1 mA in this circuit. Note that if by chance you do have an ECC86 available, it cannot be fitted directly in the same socket: it has no centre tap on the heater, and so requires a voltage of 6.3 V between pins 4 and 5.

The first simple amplifier circuit works perfectly well with highimpedance (600 Ω or 2000 Ω) headphones. However, it is not good practice to maintain a DC current through headphones, although there is no danger of overloading the system, since the anode current is very low. The sound quality can suffer, however, at slightly increased currents and, furthermore, high-impedance headphones are relatively rare. If we want to use ordinary headphones from a personal stereo, having an impedance of 32 Ω , the output level will be very low. The problem is that the impedances are severely mismatched: the impedance of the valve output is of the order of kiloohms. An impedance converter will do the job: for example, we can use a small 230 V/24 V 1.8 VA mains transformer. This has a voltage ratio of about 10:1. The transformer we used in our experimental circuit (Figure 4) had a primary with a DC resistance of 2.5 k Ω and a secondary with a DC resistance of 100 Ω . Using a larger transformer has the advantages of lower loss and hence higher volume. With a voltage ratio of 10:1 the effective headphone impedance is increased by a factor of 100. If the headphones have an impedance of 32 Ω , the valve sees an impedance of 3.2 k Ω , which will give considerably

better results. The theoretically optimal operating resistance for the valve is in the region of U_a/I_a , which, at an anode current of only 0.17 mA is around 70 k Ω . The exact impedance is not critical in this application, and so even headphones with an impedance of 600 Ω can equally well be used in conjunction with the same transformer. The impedance seen by the valve would then be about 60 k Ω , practically ideal for the given anode current.

Characteristic curves

The oscilloscope traces in Figure 5 show that the valve does indeed work with a low anode voltage. The voltage gain of about 8 was obtained at high impedance using an impedance converter and 600 Ω headphones. In general, the following relation holds for voltage gain:

$$V = S \times R_a$$

where V is the voltage gain, S the transconductance and R_a the output resistance. Given that the output resistance is 60 k Ω . the transconductance of the valve must be 0.13 mA/V. This broadly fits in with the general observation that the transconductance of a valve at any operating point is approximately equal to the anode current divided by 1 V. The official data sheet for the ECC81 gives, for example, an anode current I_a of 3.0 mA at $U_a = 100 V$ and $U_q = -1$ V, with a transconductance of 3.75 mA/V. This comparison also shows a disproportionate fall in current and transconductance when operated at an anode voltage of only 12 V. This means that for serious applications the anode voltage should be as high as possible. An acceptable compromise between safety and gain might be around 24 V

In order to learn to understand the properties of the valve at low anode voltages, we need to study its characteristic curves. Manufacturers' data sheets are of no help here, since they do not cover operation at such low voltages (it was apparently at that time not of any interest). For the same reason ordinary simulation programs do not give realistic results.

To measure real data all that is

needed is to apply a variable grid voltage and measure the anode current (Figure 6). The measured characteristic curve shown in Figure ${\bf 7}$ indicates a rise in transconductance with anode current at negative grid voltages. When the grid voltage is positive, the transconductance stops rising, and in the region above +1 V, it starts to fall again. At the same time the grid current rises and, above about U_q = +0.5 V, can exceed the anode current. It is worth plotting the characteristic curve, in particular for used valves, in order to determine an optimal operating point.

Positive Grid

If it is desired to build the headphone amplifier as a permanent device, rather than merely experiment with it, it might be found that the output volume can be too low in some conditions, especially when operating from a 12 V supply. What is needed is to give the electrons a bit more energy by applying a slightly positive grid bias voltage. Old hands will now protest vehemently that this implies that a grid current will flow, resulting in severe distortion. That is true in principle, but it is not a problem if the grid drive has a relatively low impedance. Previously, in the golden era of valves, the grid had to be driven from a high impedance source, since the output of the previous stage necessarily had a high impedance. Today we can use the low-impedance headphone output of a small

CD player, and a little grid current no longer matters.

In the circuit in Figure 8 a grid current of the same order of magnitude as the anode current is set up. The anode current and the achievable output drive are now three times higher than with a grid voltage U_{α} of –0.1 V. This gives almost ten times the output power, which should be adequate for many uses. The grid voltage is set at +0.5 V and the anode current is 0.5 mA. We are therefore in the region of the characteristic curve where the transconductance is constant, and so distortion should be low. The sound of this simple amplifier is indeed very good, even though it might not be perfect from a purely technical point of view. The inevitable distortions introduced by a valve stage, especially when driven hard, are however generally not regarded as unpleasant.

Perhaps you have noticed that in this circuit the valve can be simply replaced by two NPN transistors. Instead of a grid current we have a base current, instead of an anode current. a collector current. Of course, the transistors no not need a heater: such is the nature of progress. Which circuit sounds better is a matter of taste: try it for yourself. Most people come to the conclusion that the valve sounds better. It is worth putting up with the fact that the power consumed by the heater is orders of magnitude higher than the output power of the amplifier: in return one can enjoy the cosy glow of the cathode and



Figure 8. Stereo amplifier with positive grid bias.



Figure 6. Plotting the characteristic curve.



Figure 7. Characteristic curve for an ECC81 at $\rm U_a=12$ V.

the opportunity to warm ones hands (carefully!) on the valve.

More power

Readers with a lust for power who are tempted to try to go a step further and drive the headphone amplifier to the edge of distortion might prefer to wait for the second article in this series. We will be looking at real power valves such as the EL84, EL95, ECL80 and ECL86, as well as a PL504, which we will be using in an amplifier with a loudspeaker output, and at an anode voltage of only 27 V. We will also describe several miniature Russian 'battery' valves, which not only work with low anode voltages, but which also dissipate much less heater power.

(030063-1)

Links

Hobby projects and experiments: http://www.b-kainka.de/ Chelmer Valve Company: http://www.chelmervalve.com

9/2003

Valves at Low Plate Voltages (2)

Part 2: more power!

by B. Kainka

Genuine power valves such as the EL84, EL95, ECL80 and ECL86, and in particular the PL504, do of course offer more power output at low anode voltages than the ECC81 and ECC82 types we discussed in the first part of this series. With the PL504 it is even possible to drive an ordinary low-impedance loudspeaker, using an anode voltage of only 27 V. In this second article we will also be looking at some miniature Russian 'battery valves', which not only operate at low anode voltages, but also demand considerably less heater power.

If, after our initial experiments with ECC81s and ECC82s (or their US equivalents 12AT7 and 12AU7), you have come to the conclusion that our valve headphone amplifier could do with a little more power, then it is time to take a look at some valves which are designed for higher power. Suitable candidates would be output-stage valves such as the EL84 (6BO5), EL95 (6DL5), ECL80 (6AB8), ECL86 and similar types.

The EL95

Here we tested a used EL95. The EL95 is a pentode, which means that it has two more grids than a triode. Grid 2 should be connected to the supply voltage, while grid 3 should be connected to ground. Pentodes have a higher gain and lower distortion than triodes. The EL95 is used here with an anode voltage of between 12 V and 24 V. **Figure 1** shows the circuit arrangement for one channel and the pinout of the valve.

The EL95 is relatively economical, drawing a heater current of only 200 mA. Despite $% \left({{{\rm{T}}_{{\rm{T}}}} \right)$

this, we obtain an anode current of 1.3 mA at 12 V and 3.5 mA at 24 V. A respectable power output can be achieved using a suitable transformer. A usable stereo headphone amplifier for low-impedance (32 Ω)

headphones can be constructed to run from a 12 V supply, for example from an ordinary mains adaptor. The heaters of the two valves should be wired in series to the same 12 V supply used for the anodes.





Figure 1. Amplifier using an EL95.

If good sound quality is important then the right transformer must be used. Very small mains transformers have a high winding resistance and saturate too easily. If larger transformers are used, the saturation problem disappears, but the inductance will be lower and so low frequency response will be impaired. A mains transformer can only replace a genuine audio transformer in limited circumstances: in general a custom transformer must be wound for each design of valve output stage, which is of course rather inconvenient.

And so we return to the simplest design and connect the headphones directly in the anode circuit. Experiment shows that this works well with 600 Ω headphones. The question is whether the headphones can tolerate a direct current of 3.5 mA, for example. This corresponds to a

static dissipation of just under 7.5 mW in the quiescent state. The thermal effect of this does not seem excessive, since headphones are typically rated up to 100 mW. A different question is whether the mechanical displacement of the diaphragm will lead to distortion. The direct current is effectively a mechanical pre-tensioning, or 'bias', of the diaphragm, which could, in principle, affect the sound. In practice, however, there is no perceptible disadvantage, and so it seems a sensible idea to construct a headphone amplifier without transformers.

If it is desired to avoid both the use of transformers and a direct current in the headphones, then capacitor coupling can be used (**Figure 2**). In this case, high-impedance (600 Ω) headphones must be used. The volume is lower than with an ideal out-



Figure 3. Loudspeaker amplifier using a PL504.

10/2003



Figure 2. Transformerless headphone amplifier using an EL95.

put transformer, but is entirely adequate for most applications.

Class A amplifier using a PL504

Can we do a little more? What about using a PL504 in a small class A loudspeaker amplifier? The PL504 is a physically large valve with a Magnoval base, which was used in the line output stages of television sets. Its heater requires 300 mA at 27 V. It therefore makes sense to use an anode voltage of 27 V also. In its intended application in the line output stage the valve had to operate at anode currents in excess of 500 mA, and of course this means that we can also expect good performance at lower voltages.

An experiment using 27 V produces an anode current of 33 mA. That is already more than (according to its data sheet) the EL95 can deliver at 250 V — we will certainly be able to build a very useful loudspeaker amplifier using a PL504 at 27 V. A reasonably large transformer will be needed for good results. The impedance connected to the valve should be around 800 Ω . The circuit was tested using a large mains transformer with a 230 V primary and a 24 V secondary, having a turns ratio of around 10:1 and hence an impedance ratio of around 100:1. An 8 Ω loudspeaker is then just right, since the valve will see 800 Ω . An oscilloscope can be used to check that the valve is suitable. If, when the amplifier is driven into limiting both positive and negative half-cycles are clipped to an approximately equal degree, then the circuit is optimally arranged.

In use, this amplifier (**Figure 3**) delivers plenty of volume and a pleasant sound. The anode power dissipation is around 1 W, as would be expected. The 27 V supply voltage is a little inconvenient: an experiment was tried using 24 V. The lower heater power does



Figure 4. Headphone output stage running on 24 V.

not appear to have any ill effect; the anode current falls to 25 mA and there is practically no effect on the sound.

PL504 headphone amplifier

When building a headphone amplifier, the PL504 has plenty of spare capacity to allow RC coupling to be used at the output. A cathode resistor of 100 Ω provides a grid bias voltage of -1.3 V. At the same time, distortion is reduced due to negative feedback. A 680 Ω resistor in the anode connection allows the amplified signal to be coupled out of the circuit. In this circuit (**Figure 4**) only about half of the amplified signal current flows through the headphones, the other half flowing through the anode resistor. With an anode current of 12 mA we can put up with this loss.

This headphone amplifier can be constructed without difficulty, since there is no need to find a suitable output transformer. The circuit has a pleasant sound and plenty of volume. If you can find two PL504s, consider yourself lucky; alternatively, there is the EL504 which has a heater voltage of 6.3 V at a current of 1.3 A. This is practically the same valve, but with a different heater. And of course there are many other power pentodes around which similar circuits can be designed.

Low-voltage valves from Russia

The 1SH24B, 1SH29B and 1P24B (Figure 5) are tiny battery valves from the former Soviet Union. These valves are still available in large quantities and can be obtained very cheaply. They have wire connections that can be soldered directly, and so special sockets are not required. A characteristic of these valves is the directly heated cathode: in other words,

the heater is also the cathode. This has an impact on the circuit design, since it makes it difficult to wire heaters in series.

Most valves have a concentric construction, but these are rather different. In the centre is a thin heater element which forms the directly-heated cathode; in the case of the 1SH19B and 1P24B there are two cathodes. All the other electrodes take the form of plates or wires arranged parallel to the cathode. This makes for a very robust and efficient valve.

The 1SH24B (high frequency pentode) has a heater current of just 13 mA at 1.2 V — a marvel of efficiency. The 1SH29B (universal pentode with $P_a = 1.2$ W) has a heater current of 64 mA at 1.2 V or 32 mA at 2.4 V.

Initial experiments with the 1SH29B show that it has very useful properties: at $U_a = U_{c2} = 40$ V the

anode current is 3 mA, grid 1 being held at the same voltage as the negative heater terminal. The transconductance is about 1 mA/V. The valve also works at lower voltages, but the anode current and transconductance fall off sharply. Even at an anode voltage of 12 V the characteristics are better than those of the ECC81.

Still better is the 1P24B (power pentode with $P_a = 4$ W). The 'P' might stand for 'power', and this tiny valve certainly has that. It needs a relatively high heater current of around 240 mA at 1.2 V, but in return we get very useful results even at an anode voltage of 12 V. With a grid voltage of zero we obtain an anode current of 2 mA and a transconductance of 1.5 mA/V. The valve is therefore particularly suitable for use in a small headphone amplifier. **Figure 6** shows a tested circuit for use with high-impedance headphones.

(030063-2)



Figure 5. The ISH24B, ISH29B and IP24B.



Figure 6. Audio amplifier using a 1P24B.

ECC83 (I2AX7) Microphone Preamplifier

studio quality with valves

Design by G. Haas

experience.electronics@t-online.de

In this semiconductor age, we find valves being used increasingly often for hi-fi and guitar amplifiers, top-end condenser microphones and studio equipment. This article presents an excellent microphone amplifier with a uniquely attractive sound.



Microphone amplifiers must amplify extremely small signals to much higher levels while introducing the least possible amount of additional noise. In principle, it does not matter whether a transistor, operational amplifier or valve us used as the gain element.

A signal can be amplified by any desired amount, but the limit is set by the signal-to-noise ratio. If the magnitude of the noise signal is equal to or greater than that of the desired signal, any amplification is pointless. Consequently, microphone amplifiers must be designed to have the lowest possible levels of hum, noise and distortion, since every corruption of the signal originating in the microphone amplifier will be magnified by the following amplifier. Particular attention must therefore be given to the design of the input stage.

A low-noise transistor or lownoise valve will not by itself automatically yield a low-noise amplifier.



Figure 1. Basic noise measurement circuit

U _{ntot}	$=\sqrt{(U_V^2 + U_{Req}^2)}$
U_{V2}	$= U_{Req}^2$
U _{ntot}	$= \cup \vee \cdot \sqrt{2}$
U _{ntot}	= total noise voltage
UV	= valve noise voltage
U _{Req}	= noise voltage of resistor
R _{ea}	= equivalent noise resistance

Noise arises from the motion of electrons in any type of electrical conductor. The fundamental noise level of a given component is set by its construction and the materials used. The noise generated by an input stage is determined by the valve noise (or semiconductor noise) and the internal resistance of the signal source (resistance noise).

Specifications

Supply voltages			3	350 V at approx. 4	l mA	
	for ECC	2835		12.6 V/0.15 A		
	for ECC	808	(6.3 V/0.34 A		
Frequency response	$a_u = 40 \text{ dB}$		2	28 Hz - 24 kHz (—I dB)		
Input impedance	l kHz		â	approx. 900 Ω		
Unweighted noise voltage	20 Hz - 20 kHz		-	—72.5 dBm		
Noise voltage			-	–81.0 dBm(A)		
	CCIR-4	68	-	—67.8 dBm		
Input referenced noise voltage	CCIR-4	68, a _u = 50 dB	-	–117.8 dBm		
Harmonic distorsion	dtot	d2	d3	d4	d5	
-40 dBm, a _u $= 30$ dB	0.342%	0.020%	0.287%	0.018%	0.041%	at 80 Hz
	0.023%	0%	0.001%	0%	0%	at I kHz
-40 dBm, $a_{\rm H}$ = 40 dB	0.353%	0.030%	0.294%	0.018%	0.040%	at 80 Hz
	0.025%	0.006%	0.001%	0%	0%	at I kHz
-40 dBm, $a_{\rm U}$ = 50 dB	0.350%	0.023%	0.293%	0.018%	0.040%	at 80 Hz
	0.046%	0.036%	0.003%	0%	0%	at I kHz

Noise measurements

Figure 1 shows a measurement circuit that can be used to determine the equivalent noise resistance (R_{eq}) of the valve used here (ECC83). The values of Ra and Rk are typical for this type of valve, but they anyhow do not have any effect on the measurement. First, the noise voltage of the valve (U_V) is measured at the anode with switch S closed, using a millivolt meter. The switch is then opened, and the value of \boldsymbol{R}_{eq} is adjusted until the measured value is a factor of $\sqrt{2}$ greater. The value of $R_{\rm eq}$ is then recorded; this is the equivalent noise resistance of the valve. From the formulas, it can be concluded

that if R_{eq} is smaller than $R_V,$ valve noise predominates, while if R_{eq} is greater than R_V , resistance noise predominates.

If a pentode is used instead of a triode, there is an additional noise source in the form of partition noise. In a pentode, the number of electrons leaving the cathode is larger than the number arriving at the anode. As more electrons leave via the screen grid, the noise level increases. This is why we often see an EF86 pentode, which has low noise and microphonics, wired as a triode. The larger gain that can be achieved with the pentode configuration has been foregone in favour of better noise performance. A pentode in the triode configuration, or just a triode, is often used in such cases. Triodes also have a structural advantage over pentodes, in that they tend to produce second-harmonic distortion. This is



Figure 2. Microphone impedance matching using an input transformer.



Figure 3. An inverting operational amplifier using a valve.

more pleasant to the ear than the 'scratchy' third-harmonic distortion produced by pentodes due to variations in the division of the cathode current between two electrodes, the anode and the screen grid, which depends on the drive level.

Transformer matching

In traditional circuits, such as that shown in Figure 2, an input transformer is used to match the microphone impedance to that of the valve. This transformer typically has a turns ratio of 1:10 to 1:30. With an input transformer, it is possible to boost the input signal level with practically no noise. However, stray circuit capacitances in combination with transformer capacitances limit the upper corner frequency and linearity of this arrangement, especially at large turns ratios. This problem can only be mastered using an elaborate transformer construction and sophisticated circuit design. The valve in Figure 2 works without feedback, so the amplification factor depends only on the turns ratio of the input transformer and the transconductance (g_m) of the valve. If the valve is replaced, the gain may also change.

Opamp circuits

A valve can also be wired as an operational amplifier, as shown in **Figure 3**. The plus and minus signs next to the valve electrodes identify the corresponding inputs of the valve opamp. Capacitors C1–C3 serve only to separate dc and ac voltages; in principle, they have no further effect. The gridleak resistor R3 is needed by the valve, but its resistance is so large that it has no significant effect on the overall circuit. The cathode of the valve corresponds to the noninverting input of the opamp. Since Rk is needed to set the dc operating point of the valve, it must be bypassed for ac signals by Ck to connect this input to signal ground.

Now we have an inverting opamp whose gain is set by the resistance ratio R2:R1, independent of the amplifying component. Of course, the open-loop gain of this component must be significantly greater than the value of R2:R1. The input resistance of the circuit is equal to that of R1. As the value of R2 cannot be made arbitrarily large, since the value of grid-leak resistor R3 also cannot be made arbitrarily large, the value of R1 will be relatively small for large amplification factors. This imposes a significant load on the signal source. The internal resistance of the signal source forms a voltage divider in combination with R1. The control grid, just like the inverting input of an



Figure 4. A non-inverting operational amplifier using a valve.

opamp, represents a virtual ground.

If the opamp circuit is modified as shown in **Figure 4**, the gain is essentially determined by the ratio $R_B:R_A$. This gives us considerably more freedom in selecting the values of R1 and R2. If R1 and R2 are now replaced by an impedance-matching transformer, R1 becomes the source impedance of the signal source and R2 becomes $R1 \times n^2$. An equivalent circuit using a triode guarantees high gain with low noise. However, this arrangement has the disadvantage of having a limited amount of fundamental gain.

This situation can be improved using the circuit shown in **Figure 5**, which includes an additional valve. V2 acts as an impedance converter, since the feedback signal is taken from the cathode resistor. This yields

the same considerations for \boldsymbol{R}_{A} and R_B as in Figure 4, but since the cathode resistor of V1 is not bypassed, the fundamental gain is less. This has a beneficial effect on the distortion characteristic and long-term stability of the circuit, due to the use of negative feedback. The emissivity of valve cathodes decreases with age. If a lower level of system gain is used from the start, the useful life of the valves is extended. Valve V2 makes up for the missing gain. Here again, the cathode resistor is not bypassed with a capacitor, since the ac voltage on the cathode is needed for the negative feedback. Overall negative feedback is also provided via $R_{FB}\!,$ in order to constrain the characteristics of the overall system without requiring selected valves to be used.



Figure 5. Amplifier with impedance converter.

Microphone preamplifier

Figure 6 shows the complete schematic diagram of the preamplifier, with all component values. The input transformer (type E-11620), which is one of the most important components for this application, is wound with a turns ratio of 1:8+8. Here it is wired for a 1:16 ratio. This provides a good compromise between signal level boosting and the noise performance of the circuit. Furthermore, this transformer can also be used for other purposes, so its price can be kept within reasonable limits by virtue of a relatively large production volume.

The input transformer can be used with an input level of around 800 mV_{eff} at 40 Hz, but that does not mean that the amplifier circuit should be fed such a strong input signal. The maximum input level depends on the maximum output level of the complete installation. The transformer is fully encased in mu-metal, since otherwise even minute amounts of coupled-in noise would be amplified to high levels by subsequent amplifier stages.

The component values have been chosen to allow a gain of around 25 to 60 dB to be used with high sound quality. The gain is essentially determined by the values of R6 and R15. A gain of 25 dB is provided by the signal level boost of the input transformer alone. A fixed minimum gain can be thus set using R6. R15 can also be replaced by a wire bridge, a selector switch with fixed dB settings, or a trimpot. Of course, only premium-quality components should be used for this purpose. The selector switch must have goldplated contacts and make-beforebreak switching, since otherwise it will produce crackling noises and switching clicks.

Coupling capacitors C4 and C5 are specially marked in the schematic diagram. The marking indicates the lead connected to the outer foil of the capacitor, which should be connected to the non-critical side of the circuit. Many types of film capacitors are correspondingly marked. The result is that the capacitor screens itself, thereby reducing the susceptibility of the circuit to interference. The printed circuit board, whose layout is shown in **Figure 7**, allows the input transformer to be used at a ratio of either 1:16 or 1:8 by means of wire bridges. This allows other types of valves with the same basing to be used, such as the ECC81, ECC82 or similar dual triodes. However, if a different type of valve is used, the component values cannot simply be used as is. It is essential to modify them as necessary to match the dc operating point of type of valve used.

Components R3, C1 and C9 attenuate the resonance peak formed by the input transformer in combination with the amplifier circuit, in order to make the frequency response of the amplifier as flat as possible. The indicated component values can be adjusted as necessary according to circumstances. With the indicated values, the overall arrangement has a slight rise in the frequency response (around 0.8 dB) at 17.7 Hz. This could be suppressed even more, but only at the expense of a lower corner frequency at the high-frequency end.

Resistor R1 provides a finite load for the input transformer. The grid of the valve has such a high impedance that the transformer would otherwise operate with practically no load on the secondary. Since this can also result in a non-linear frequency response, a finite load impedance provides a definite benefit.

High-quality power supply

Both the enclosure for the circuit and the power supply must meet demanding require-



Figure 6. The final circuit of the microphone preamplifier, including the base diagrams for the two types of valves used.

f filament	al anode I	all anode 2
gl grid l	gll grid 2	kl cathode l
kll cathode 2	fM filament tap	s screen
(as seen from the bottom viev	ving the pins)	

COMPONENTS LIST

Resistors:

(metal film, 1% tolerance, 0.7 watts, unless otherwise noted)

 $RI = IM\Omega$ $R2 = 10k\Omega$ $R3 = 180k\Omega$ $R4 = Ik\Omega5$ R5 = 10k $R6 = 100\Omega$ $R7 = IM\Omega$ $R8 = Ik\Omega5$ $R9 = IM\Omega$ $RI0 = Ik\Omega5$ R11 = 470k Ω , metal oxide, 2% tolerance, 2W RI2,RI3 = 220k Ω , metal oxide, 2% tolerance, 2W $RI4 = 4k\Omega7$ RI5 = see text and Table 2

Capacitors:

CI = 680pF ceramic C2,C3 = only fitted when oscillation or RF noise is noted (approx. 10-47pF) C4,C5 = 0.22μ F 630V, MKS4, lead pitch 22.5mm C6,C7 = 10μ F 450V, lead pitch 5mm C8 = 100μ F 40V, lead pitch 5mm C9 = 100pF ceramic

Semiconductors: D1,D2,D3 = 110V zener diode, 1.3W

Miscellaneous:

RI (\ddot{U} I) = E-11620 VI (R \ddot{o} I) = ECC83S, E83CC, 12AX7, ECC808 (see text) I valve socket, ceramic, PCB mount

Kits, special parts and PCBs available from

Experience Electronics Weststrasse I D-89542 Herbrechtingen Germany

Internet: <u>www.experience-electronics.de</u> E-Mail: <u>experience.electronics@t-online.de</u>

Tel.: +49 7324 5318 Fax: +49 7324 2553

Maximum input voltage

(as a function of gain, for I percent total harmonic distortion)

a _u	u _i	R15
25 dB	375 mV	0Ω
30 dB	180 mV	llkΩ
40 dB	180 mV	62 kΩ
50 dB	85 mV	1 73 kΩ





Figure 7. Circuit board layout for ECC83 (board available from Experience Electronics).





Figure 8. Circuit board layout for ECC808 (board available from Experience Electronics).





Figure 9. –30 dB input attenuator and connections for a phantom supply.

ments, since the circuit will deliver good results only if it is fitted into a fully screened metallic enclosure. The valves are heated using a 12.6-V dc voltage. The high voltage must be well smoothed. Suitable circuits have already been presented for the Valve Preamplifier (*Elektor Electronics*, June through September 2000 issues). Zener diodes D1–D3 must be used if several preamplifiers are powered from a single supply, or if the power supply has passive RC smoothing. The power supply output voltage must be 350 V.

Using a stabilised supply voltage provides the valves with well-defined operating conditions, which is beneficial since the gain of a triode more or less depends on the value of the supply voltage. An important point is that the negative terminal of the filament voltage must be connected to the negative terminal of the high voltage.

When choosing a valve type, you should pay attention to certain details. The measured performance values were achieved using an ECC83S, which is a cross between the ECC83 and the E83CC (military version). The noise figures of ECC83S valves are significantly better than those of standard ECC83 valves, so the ECC83S is clearly preferable. The ECC83 is also available with a variety of American designations, such as 12AX7, which exactly corresponds to the standard ECC83. The 12AX7A and 12AX7WA are versions with tighter tolerances, lower noise and lower microphonics, while the 7025 is the long-life version. An E83CC, or one of the equivalent American military versions with type numbers such as 6681, 6057 and 5751, can also be used if desired. Although these types are significantly more expensive, they have the advantage of being less microphonic and having longer service lives than the standard type.

The term 'microphonic' refers to the fact that mechanical vibrations, particularly in the control grid, can modulate a valve and lead to unpleasant noises or howling in an amplifier installation. This is thus not the place to cut costs in a good-quality microphone preamplifier.

The preamplifier should not be fitted into the same enclosure as the power supply, since otherwise electromagnetic interference and mechanical humming from the mains transformer can manifest themselves in an unpleasant manner. In some cases, it may be necessary to mount the circuit board elastically, for instance using rubber bushings. The circuit is designed such that it is not necessary to use selected valves.

There is yet another interesting option. The ECC808 valve was developed in response to the shortcomings of the standard ECC83 or its direct equivalent the 12AX7. The ECC83 and ECC808 are identical electrically, but the noise characteristics of the ECC808 are better by a factor of three, it is less sensitive to hum and it is significantly less microphonic. Its noise characteristics roughly match those of the ECC83S. In addition. it has a screen between the two triodes, which is of secondary importance in this application. The base arrangement is also different, with the control grid pins being located well away from the anode and heater pins. Consequently, and ECC808 cannot be used as a direct replacement for an ECC83. For this reason, we have also developed a second circuit board layout, as shown in Figure 8. The component values remain exactly the same, with the only difference being that the ECC808 requires a filament supply of 6.3 V dc at 0.34 A, instead of the 12.6 V at 0.15 A used for the ECC83. Unfortunately, the ECC808 is not exactly cheap, since it has become scarce. However, it represents an interesting alternative, and its price can be justified in a high-quality microphone preamplifier stage.

Interpreting the measured values

The measured values for the amplifier, which are shown in **Table 1**, require a little bit of interpretation. The open-loop gain, which is the gain when R15 is not fitted, is around 68 dB. If we want to allow a maximum gain of 60 dB, this leaves only 8 dB for negative feedback, which is not very much. Valves do not have high open-loop gains, unlike modern opamps. Consequently, it is recommended to select a gain in the range of 30 to 50 dB, since the best results with regard to harmonic distortion and frequency response will be obtained in this range.

The harmonic distortion measurements were made at 1 kHz and 80 Hz. As can be seen, the harmonic distortion increases at low frequencies, particularly odd harmonics. The influence of the input transformer can be seen here, since matching transformers generate predominantly this type of harmonic distortion components. The even harmonics can be attributed to the valves. The second-harmonic component has a pleasant sound that is typical of a good 'valve sound'. An increase in harmonic distortion at low frequencies is not especially serious, since the ear is relatively insensitive in this range. Another thing that can be seen from the harmonic distortion values is that the total harmonic distortion at 1 kHz is greater than the average value of the individual harmonic distortion values. At this frequency, amplifier noise predominates. In this case, the measurement equipment cannot distinguish between harmonic distortion and noise, since it makes broadband measurements at frequencies above 1 kHz.

The noise values are to be understood as absolute voltage levels at the output of the amplifier. The input-referenced noise values are obtained by assuming a noise-free amplifier with a noise source at a certain level connected to its input. Three noise values are given: 20 Hz - 20 kHz, A-weighted and CCIR-486. The CCIR-486 filter is used with studio equipment. With this filter, instead of measuring the effective noise value, the rectified peak value is measured using a filter characteristic similar to that of an Aweighted filter, but with the noise components between 1 kHz and 12 kHz being significantly more heavily weighted. That is why it gives the worst noise value.

In order to correctly evaluate the amplifier, it is necessary to correctly interpret the measurements. If a 200-ohm metal-film resistor is connected to the inputs of the instrument, a level of around -118 dBm is measured using the CCIR-486 filter. A

dynamic microphone with a source impedance of 200 Ω generates a noise voltage of -118 dBm. If we assume our amplifier to be noise-free and subtract its gain from the noise level measured at its output, we arrive at a value of -117.9 dBm (weighted using the CCIR-468 filter). This means that the amplifier is only 0.2 dB away from what is physically achievable (0 dBm = 775 mV, the standard studio level).

There is another important point to consider, namely the maximum input voltage. It must be borne in mind that the input transformer boosts the input level by a factor of 16. Thus, if a level of 10 mV is present at the transformer input, the voltage on the grid of the first valve is already 160 mV. Since the grid voltage is only around -1.2 V, the knee of the characteristic curve is reached fairly quickly. The maximum input level for 1 percent harmonic distortion depends on the gain. Several typical values are listed in Table 2. A value of 85 mV for a gain of 50 dB may not appear particularly high. However, a dynamic microphone has a nominal level of 2 mV. If the amplifier can handle 85 mV. there is still 18 dB of headroom.

If you want to use this amplifier with a relatively high input signal level, an input attenuator should be used as shown in **Figure 9**. With the indicated component values, the attenuation is approximately 30 dB. If you want to have an exact value, or if you want to modify the attenuation, you can adjust the value of the 270- Ω resistor. Figure 9 also shows how a 48-V phantom supply can be implemented.

(020323-1)

Valve Amplifier Revival

return of the 'warm' sound

By Harry Baggen

Despite semiconductors being all around us, it seems that valves have never lost their appeal and even enjoy increasing interest from the 'all solidstate' generation. Among audio purists, valves have an excellent record for natural sound reproduction. The Internet is full of DIY valve amplifier designs, often complete with extensive description and photographs.

Despite the fact that semiconductors have been with us for more than 50 years, there is still a large number of audio lovers who prefer the warm sound of a valve amplifier over that produced by a transistor rig. Setting aside the air of nostalgia around valve amplifiers and their fine appearance, audio lovers will insist that valves are winners when it comes to sound reproduction with character. Despite the mechanical and electrical pitfalls encountered in the design and construction of valve amplifiers, like exotic transformers and dangerous supply voltages, thousands of enthusiasts enjoy tinkering with valves. Some do so with an innovative approach, while others focus on constructional aspects and strive to give their amplifier the best possible appearance.

On the Internet, a veritable galaxy of photographs may be found showing home built valve amplifiers. Once you start searching the net for websites covering valves, the amount of information is overwhelming. Besides true hobbyists you will also find companies professionally engaged in valve amplifier technology (still widely used in transmitters). Other firms specialize in kit, ready-built amplifiers or components. However these websites will not be discussed here since our aim was to see what was going on in the hobby department!

A great starting point for your explorations is the **World Tube Portal** [1], a website specialized in links to do with valves. Although these links have been sorted alphabetically, it is also possible to search in 105 categories.



Two of these links found under the heading 'DIY Tube Audio Sites', **DIY Tube Amplifiers** and **DIY Tube Audio Sites** [2] are of particular interest to hobbyists. Together these two links cover about 150 websites.

Once you start visiting a couple of the websites mentioned on the portal, you soon notice that there is much more to be discovered. We had to make a selection from the vast amount of information on offer, and have to limit ourselves to a couple of sites we found either particularly attractively styled or rich in useful content.

To begin with, we should recommend **Audio Bizarro** from tube 'fanatic' Ralph Power. On his website Ralph shows a lot of photographs of valve amplifiers [3], which are a sight for sore eyes! Some amplifiers are complemented by a circuit diagram. However the feature *par excellence* of Ralph's website is his set of guidelines for



building valve amplifiers. Of particular interest are his extensive descriptions and carefully finished photographs that go with the designs. There are also quite a few tips for music to play through your valved stereo apparently Dick is a great fan of Mobility

ELECTRONICS ON INF

Fidelity Soundlab. Finally, two websites with lots of schematics of valve amplifiers from the past — an excellent source of inspiration for new designs! The British website **The Circuit Archive** [13] holds an extensive archive of all designs by Dynaco and Heathkit, which were once famous for their kits. The schematics cover a lot of valve designs.

The company **Triode Electronics Online** [14] has a website with an extensive archive of antique schematics. Most amplifier brands are found in the archives. If not, a link is given to another collection of schematics on the net.

newcomers [4]. On 14 pages, Ralph tells you all you need to know to get started with valve amplifiers for home construction.

The hobby is international, too, as we soon discovered. **Bob Danielek** from the USA [5] designs and builds valve amplifiers large and small. Of the latter, the 'Darling', a super simple 1.5-watt stereo design, is fairly well known. Another, more unusual, design is a valve amplifier for use in a car — Bob has actually succeeded in fitting his sports car with a small, valved amplifier incorporating a power inverter for the high voltage supply.

The Japanese also have a soft spot for valves and the relevant websites we came across are often marked by original ideas. Of the many websites we came across we should mention **Explore the wonders of direct heating** [6] which describes the Sakuma principle.

The Canadian website run by **Rudy Godmaire** [7] allows you enjoy a visual impression of his beautiful designs of 70-watt power amplifiers. Rudy was obviously inspired by the designs of Menno van der Veen, an authority in the field and author of the book *Modern High End Valve Amplifiers* published by Elektor Electronics.

Claudio Bonavolta [8] from Switzerland is convinced that only

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valves can bring across musical listening enjoyment. Besides a description of his own audio system (using a 300B single ended SRPP final amplifier), his 'Electronics' pages present a wealth of circuit diagrams employing transistors, valves and even hybrids. Here you will find nearly everything you need for a complete audio system, ranging from MC preamp to power amplifier.

Of German origin, **Roehrenfieber** [9] by Ulrich Romanski offers other valve amplifier constructors an opportunity to present their design on his website. Tuning tips are also discussed, and there is an interactive introduction into the basic operation of the valve.

The Dutch, too, have a reputation to keep up when it comes to 'all things valved'. **Aren's Attic** [10] not only presents a description of two amplifiers and a line amplifier, it also provides an excellent link list to glance through.

For those interested in the renowned Williamson amplifier design, the web pages put together by **Bert van der Kerk** [11] are worth visiting. There, the construction of a Williamson amplifier is described.

Triode Dick's page [12] is the third but by no means least Dutch website we should mention in this article. On his homepages, Dick reports on his own ventures in

Internet addresses

- [1] World Tube Audio Portal: <u>www.worldtubeaudio.com/</u>
- [2] DIY Tube Amplifiers en DIY Tube Audio Sites:

www.worldtubeaudio.com/ diy_tube_amplifiers.htm www.worldtubeaudio.com/ diy_tube_audio_sites.htm

- [3] Audio Bizarro Constructors Page: www.spiritone.com/~tube/constructor.htm
- [4] Building your own tube amp: www.spiritone.com/~tube/ralphbuild.htm
- [5] Bob's Music/Vacuum Tube Audio/electronics Page: <u>www.geocities.com/TimesSquare/1965/</u>
- music_etc.html
 [6] Explore the wonders of direct heating:
 www10.big.or.jp/~dh/index.html
- [7]Rudy Godmaire DIYer zone: www3.sympatico.ca/r.godmaire/rg32e.html
- [8] Claudio Bonavolta: www.infomaniak.ch/~bonavolt/audio.htm
- [9] Roehrenfieber: www.roehrenfieber.de
- [10] Aren's Attic: <u>http://members.tripod.lycos.nl/Aren/</u> index.htm
- [11] The Williamson Tube Amplifier:
- <u>www.spiritone.com/~tube/constructor.htm</u> [12] Triode Dick's Page:
- www.triodedick.com/
 [13] The Circuit Archive:
- www.circuitarchive.co.uk/heath.htm www.circuitarchive.co.uk/dynaco.htm
- [14] Triode Electronics Online: www.triodeel.com/schindex.htm

Tube Box

that authentic valve sound — using diodes

Design by M. Radler

The Tube Box lends to a solid state amplifier that characteristic valve sound required by any self-respecting guitarist.



Guitar valve (or 'tube') amplifiers, as used on stage or in the studio, derive their characteristic sound from the clipping behaviour of the valves when overdriven. Whereas the effect of clipping in a solid state amplifier is very unpleasant (and for many this alone is reason enough not to touch them), it is common to drive valve amplifiers into clipping. But with a disadvantage: if a valve amplifier is to be overdriven, the volume control must be turned all the way up. This does not always go down well with the other band members (or with the neighbours!). With the Tube Box circuit described here, which sits between the guitar and the valve amplifier, this is not necessary. The tiny unit applies the desired distortion to the input signal of the amplifier. The controls on the Tube Box are two potentiometers and two switches, which allow the user to choose from a wide range of possible settings, making the unit suitable for the whole spectrum of guitar music from smooth, warm blues to screaming heavy metal.

Before and after the diodes

The heart of the circuit, the distorter itself, is built around IC2.A. Two pairs of diodes, connected in antiparallel, are responsible for creating the distortion. D1 to D4 clip the signal gently in a similar way to valve amplifiers, not suddenly as in solid state amplifiers. When the signal amplitude becomes very high, LEDs D5 and D6 also come into play and



Figure 1. Several filters and a distortion stage: the Tube Box circuit.

reduce the gain of the op-amp. The combination of these two distortions simulates fairly accurately the distortion typically introduced by valves, since the clipping diode adds odd (and therefore pleasant-sounding) harmonics to the notes.

We now come to the circuitry before and after the distortion stage. In order for the distortion stage to work as 'naturally' as possible, the input signal must be suitably preprocessed. The signal from the guitar is first taken through a highimpedance input circuit that forms a filter, and then to operational amplifier IC1.A. The op-amp functions as a buffer-amplifier and prevents too high a load being presented to the high-impedance guitar pickup.

From the output the signal goes two ways: first, via a filter to

footswitch S3, and second, via a filter to IC1.B. This op-amp amplifies the signal by a factor of about 5. The footswitch allows the distortion circuit to be bypassed, taking the guitar's signal practically unchanged to the output socket.

On the way to the distortion stage the signal goes via a bandpass filter, and then to another buffer-amplifier. The band-pass filter has the effect of boosting the important part of the frequency spectrum. The gain is adjustable via potentiometer P1. Note that this affects not the volume, but rather the degree of distortion. The further the potentiometer is advanced, so the more distortion is applied to the signal before it arrives at IC2.B. Here, we have a further potentiometer with which the volume can be adjusted so as to make the signal level at the output the same whether it goes through the whole circuit or whether it bypasses the circuit via S3. This ensures that there are no sudden jumps in volume when the guitarist switches between the distorted and the clean sound. Of course, P2 can also be adjusted to make the distorted signal louder than the undistorted, so that the guitarist's sound comes more into the foreground during a solo, for example.

There follows a fairly sophisticated tone control circuit with a range of possible functions. Switch S1 allows the user to choose between boosting and attenuating the middle-frequency range. Attenuation is especially suitable for 'harder' music, such as metal, while for the 'softer' blues solos the middlefrequency range should be boosted.

The tone adjustment is unusually constructed, but is nevertheless effective. Double-gang potentiometer P3 simultaneously

adjusts the treble and bass, while the position of S2 determines whether the bands are adjusted in the same or in opposite directions. Thus in one position the treble and bass are either boosted or attenuated together, while in the other position the bass is attenuated while the treble is boosted, and vice versa.







Figure 2. Layout and component mounting plan for the tiny printed circuit board (board available ready-made).



COMPONENTS LIST

Resistors: $RI = 5IIk\Omega$ $R2 = 680k\Omega$ R3 = 6kO8 $R4 = 15k\Omega$ $R5 = Ik\Omega 2$ $R6 = Ik\Omega$ $R7, R8, R10 = 8k\Omega2$ $R9,R29,R34,R35 = 2k\Omega2$ $RII = Ik\Omega 8$ $RI2 = 470k\Omega$ $RI3 = 3k\Omega3$ $RI4 = 100\Omega$ $R15, R26, R31 = 33k\Omega$ $RI6 = 220k\Omega$ $R17,R21,R23 = 47k\Omega$ $RI8 = 9Ik\Omega$ $R19,R20 = 43k\Omega$ $R22 = 100k\Omega$ $R24 = 22k\Omega$ $R25,R33 = 68k\Omega$ $R27 = 470\Omega$ $R28 = 180k\Omega$ $R30 = 10k\Omega$ $R32 = 5k\Omega I$ $PI,P2 = 100k\Omega$ potentiometer, logarithmic law $P3 = 50k\Omega$ potentiometer, stereo, logarithmic law

Capacitors:

CI,CI7,C24,C25,C26 = 100nF C2,C3,CI3,CI4 = 47nF $\begin{array}{l} {\sf C4,C5,C29,C30} = 10\mu{\sf F}\ {\sf I6V}\ {\sf radial}\\ {\sf C6} = 33n{\sf F}\\ {\sf C7,C11} = 470p{\sf F}\\ {\sf C8} = 1\mu{\sf F}\ {\sf I6V}\ {\sf radial}\\ {\sf C9} = 1\mu{\sf F}\ {\sf I6V}\ {\sf radial}\\ {\sf C10} = 2\mu{\sf F2}\ {\sf I6V}\ {\sf radial}\\ {\sf C12} = 560p{\sf F}\\ {\sf C15} = 3n{\sf F3}\\ {\sf C16,C27,C28} = 4n{\sf F7}\\ {\sf C18} = 100p{\sf F}\\ {\sf C19} = 39n{\sf F}\\ {\sf C20} = 22n{\sf F}\\ {\sf C21} = 220p{\sf F}\\ {\sf C22} = 22\mu{\sf F}\ {\sf I6V}\ {\sf radial}\\ {\sf C23,C29,C30} = 10\mu{\sf F}\ {\sf I6V}\ {\sf radial}\\ \end{array}$

Semiconductors:

D1-D4 = 1N4148 D5,D6 = LED, red IC1,IC2,IC3 = TL072-CP*

Miscellaneous:

K1,K2 = jack socket, 6.35mm, chassis mount*
K3 = solder pin
S1 = rocker switch, 1 x c/o
S2 = rocker switch, 2 x c/o
S3 = pedal switch
Enclosure: Hammond type 1590B*
PCB, order code 010119-1 (see Readers Services pages)

PCB layout available from Free Downloads section at www.elektor-electronics.co.uk

The output of the tone control circuit is connected via a buffer-amplifier. The signal now passes through a further high-pass filter and buffer, before being taken to the output connector via footswitch S3.

The distortion circuit can be powered from an external regulated mains supply, connected via lowvoltage jack K1. Alternatively, a 9 V battery can be used to power the circuit. For this option, the type TL072 op-amp should be avoided in favour of low-current quad op-amps such as the pin compatible rail-to-rail OP470.

Construction and Test

Given the small dimensions of the circuit board (**Figure 2**), the component density is very high. The majority of components are mounted vertically, and so it is particularly important to fit the smaller components first and the taller ones later. So, first fit the diodes, the LEDs, the IC sockets and capacitors (with the exception of the electrolytics), the wiring to the potentiometers and switches, and finally the resistors, electrolytics and solder pins for the signal inputs and outputs, power supply and ground (screen).

Figure 3 shows how the controls are connected. The circuit board should be fitted in a suitably-sized metal enclosure to screen the electronics from interference. For the same reason, the circuit ground (on K4) should be connected to the enclosure. This can be done with a solder tag to the base of the enclosure. For our prototype we used a Hammond enclosure type 1590B, which is very small and does not have space for a battery. To save space, instead of chassis-mounted jack sockets we used a short length of screened audio cable to connect the inputs and outputs to in-line jack sockets. Strain relief is essential for these cables.

If you would prefer to use chassismounted jack sockets, a larger enclosure will be required. Then no solder tag ground connection to the enclosure will be required, since jack sockets which are not isolated from the enclosure will do the job instead.

As well as screening the electronics, the metal enclosure has another advantage: it is very robust



Figure 3. The connections at a glance.

and so can stand up to rough treatment on stage.

To test the unit, connect up the Tube Box and set volume control potentiometer P2 to its minimum position. Strike a chord on the guitar: if nothing can be heard, press the footswitch to bypass the unit. Press the footswitch again, and the distortion circuit is switched in and P2 can be adjusted to set the desired volume. Finally, set the degree of distortion using P1 and adjust the tone as required using S1, S2 and P3.

(010119-1)



ECC86 Valve Radio



B. Kainka

Actually, the age of valves is already past but valves just refuse to go away! There's many a valve radio still in use, and there are many valves lying in the 'junk box' waiting to be rediscovered. If only we could do without the high voltages! However, there is a valve that can manage with only 6 V — the ECC86. At the beginning of the 1960s, the electronics industry was faced with a problem. The transistor had just been born, so it was finally possible to build car radios without vibrators and large transformers. However, the cut-off frequencies were still too low to allow usable VHF mixer stages to be built using transistors. This meant that a valve had to be used in a transistor circuit. This valve was the ECC86, which

was intended to be used for short wave input stages and selfoscillating mixer stages in car receivers powered directly from the car battery. According to the data sheet, an anode voltage of 6.3 V or 12 V may be used. The heater voltage is always 6.3 V. We owe the ECC86 low-voltage valve to this unique bottleneck in the history of electronics technology. Our circuit is a nearly classical valve audion for the medium-wave range. Power is supplied by a 6-V lead-acid gel battery. The circuit is nearly the same as that of a twostage amplifier. The first stage provides the demodulation and preamplification. The second stage is the audio output amplifier, which directly drives a headphone with an



impedance of 2 k Ω . A 500-pF capacitor between the two stages ensures that RF signals will not be further amplified. Otherwise the valve might easily recall its original intended use and start oscillating in the short-wave range. A ferrite rod with a diameter of 10 mm and a length of 100 mm, with a winding of 50 turns of enamelled copper wire, serves as the aerial.

The radio has a good sound and can receive local signals. In the evening, with a sufficiently long external aerial, it can receive numerous MW stations. It feels just like being back in the good old days.

(014069-1)

High Voltage PSU

a universal supply for valve amplifiers

Design by Nils Gohr

This power supply unit (PSU) provides hum-free and stable operating voltages for valve preamplifiers and power stages. Negative bias voltage included!



Editor's note:

The circuit is published as suggestion only. It has not been tested or constructed by Elektor's in-house design laboratory, hence the absence of an Elektor-style PCB layout and a parts list.

Most valve amplifier circuits require a high-voltage power supply. Here, such a circuit is proposed by one of our readers, Mr. Nils Gohr, who claims to own a working prototype.

The concept of the PSU is just fine for a power output stage using KT66 or KT88 power pentodes in push-pull or A/B mode requiring a relatively high plate voltage (here, 420 V) I combination with a grid bias voltage of -45 V for the quiescent current setting.

The preamplifier stage may be built around a double triode with two mutually decoupled 250-V anode voltages being available on the PSU.

Three voltages

The circuit requires a mains transformer with secondary windings

Figure 1. Three voltage regulator for three supply rails typically required in valve power amplifiers.

supplying 315 volts and 45 volts connected to the circuit via terminal block X1. Each output voltage has its own stabilizer. The resultant three regulators circuits are almost identical and based on the adjustable voltage regulator type LM317. The two high-voltage branches additionally feature soft start circuits based on FETs. These sub-circuits prevent the smoothing (reservoir) capacitors from being damaged by the current surge that occurs at switch on.

The raw direct voltage obtained from the rectifier on the 315-Vac winding is smoothed by C1 and applied to the voltage regulator via the FET. The direct voltage at the regulator input will be approximately 444 V. The target output voltage of 420 V is adjusted with the aid of preset P1 at the 'adjust' input of the IC. The 'adjust' voltage is always 1.2 V below the output voltage. Consequently, series network R1-R2-P1 drops about 420 V. Capacitor C2 is connected in parallel with this network to afford some buffering of the adjust voltage. Diode D1 counteracts the effect of C2 by ensuring that the adjust voltage can become positive with respect to the output voltage buffered by C5. In similar fashion, D2 protects the voltage regulator by preventing the output voltage exceeding the input voltage. All



electrolytics in this section of the PSU must be rated at 450 volts!

The soft start is implemented using an R-C network R4-C4. Capacitor C4 will charge relatively slowly via R4, causing the current through the FET to increase proportionally. Diode D10 prevents the source-gate voltage from exceeding 15 volts.

With the exception of diodes D5, D6 and D7, the 250-V stabilisation is identical to the 420-V section discussed above. The three zener diodes keep the gate of FET T2, and with it the input of the voltage regulator, at about 268 V. Because most valve amplifiers have an input stage based on a double triode, two mutually decoupled 250-V rails are available at the output. The decoupling elements are R8-C8 an R11-C7.

Finally, there's a low-voltage section based around regulator IC1, whose task is to furnish a negative bias voltage for the input grid (G1) of the output valves. The current requirement on the output should not exceed a couple of milli-amps. Because the output voltage is to be negative with respect to the PSU and amplifier ground, a separate rectifier is required to enable the enable the regulator output potential to be connected to ground. Preset P3 should be adjusted for a lower reference potential of -45 V. Although the grid bias section of the supply is also based on an LM317, it should be noted that it differs from the two HV sections in that a low-current version in a TO92 case is used.

PCB layout

The other two regulator are housed in a TO220 case and are suitable for an output current of 1.5 A. Because of the anticipated heat dissipation, they should be mounted on heatsinks. The same applies to the FETs in the soft start circuits. On the PB designed by the author, an elegant solution has been found in that the two associated TO220 parts are mounted back to back on PCB-mount heatsinks type SK129 from Fischer (Dau Components). Unfortunately there are unequal potentials at the cooling fins (LM317: output; IRFBC40: drain), so that insulating washers are required.

The PCB layout only just meets the electrical safety requirement for tracks carrying voltages of the order of 400 V and more. The requirement is a track distance of at least 0.9 mm.

Figure 2. The generously laid out PCB designed by the author (not available ready-made through Readers Services).



Warning

This circuit carries dangerous and potentially lethal voltages and should only be built and

used whilst in compliance with relevant electrical safety precautions.

(020096-1)

For the Valved RIAA Preamplifier and other applications High Voltage Supply 330 V from 12 V

Design by T. Giesberts

Although this supply was primarily designed for use with the Valved RIAA Preamplifier, we found that the inverter stage is useful in many other applications. With only a small modification this circuit can be used to power a 20 W PLCE (low energy) lamp from a 12 V car battery.



The Valved RIAA Preamplifier uses two valves, just like the Valve Preamplifier that was published in the June 2000 edition of *Elektor Electronics*. Since the valves' filaments have again been connected in series, the preamplifier requires two DC supply voltages: 12.6 V for the filaments and 330 V for the high voltage supply.

In order to avoid the need to use a custom transformer the circuit has been designed to use a standard 15 V/3 A mains transformer. As we'll see later, the supply circuit consists of two distinct sections: a conventional 12.6 V filament supply and a step-up converter which boosts the 12.6 V to 330 V. In other words, the filament supply is also used to power the inverter.

Since each section is built on a separate PCB it becomes possible to use them individually in other applications. This is especially useful in case of the inverter, since it makes a great camping light when used in conjunction with a 12 V car battery and a PLCE lamp. These lamps tend to work very well off a 300 V DC supply!

POWERSUPPLY

The 12.6 V supply

As we can see in **Figure 1**, this section is a very basic circuit. The 3 A fixed-voltage regulator (TO-220 case) is made to deliver a slightly higher output (12.6 V) by adding an extra diode (D1). The bridge rectifier uses 6 A diodes and is followed by some substantial smoothing capacitors (C4, C5, C6). The bridge rectifier is RF decoupled by C7-C10 and LED

COMPONENTS LIST

12.6 V Supply

Resistors:

 $RI = Ik\Omega 2$ $R2 = 6k\Omega 8$

Capacitors:

C1 = 10μ F 63V radial C2,C3 = 100nF C4,C5,C6 = 2200μ F 25V radial C7-C10 = 47nF ceramic

Semiconductors:

D1 = IN4148 D2 = red high-efficiency LED D3-D6 = FR606 (or similar 6A diode) IC1 = KA78T12 (3A)

Miscellaneous:

K1,K2,K3 = 2-way PCB terminal block, lead pitch 5mm
For IC1: heatsink type SK129 63,5 STS, 3.5 K/W (Fischer) (Dau Components)
Isolation material for IC1
PCB, order code 000186-2 (see Readers Services page)



Figure 1. The 12.6 V supply incorporates the well-known diode 'trick', which causes the output to increase by 0.6 V.

D2 functions as the power indicator.

Construction of the 12.6 V supply shouldn't cause any problems when the PCB shown in **Figure 2** is used. The heatsink for IC1 (Fischer type SK129 from Dau Components) is placed directly onto the PCB, which results in a compact module. It is very important that an insulating washer is used between IC1 and the heatsink.

There are two PCB terminal blocks (K1, K2) that provide the 12.6 V output voltage. One of these supplies the inverter and the other powers the two in series connected filaments. The third terminal block (K3) is for the 15 V transformer, which should be rated at least 50 VA.

The 330 V inverter

This part of the supply (see **Figure 3**) is a push-pull-converter that uses an old favourite of ours: the SG3525A. This regulator is an industry standard part that is used in many

switch mode supplies. We have used it before in the 'In-Car Audio Amplifier', which we published in 1994. The proper description of this IC is a 'regulating pulse width modulator', which sums up its function perfectly. A special transformer is driven with an alternating voltage by one or more switched transistors, with the driving voltage obviously limited to a safe value. By varying the pulse width of the signal, the amount of power is controlled. The output at the secondary of the transformer is rectified and fed back to the PWM regulator in order to keep the output stable. That completes the feedback loop of the regulator.Since the SG3525A has been described in depth before in Elektor, we will limit ourselves to a brief overview of the device. The regulator uses a reference voltage of 5.1 V. Various internal circuits use this reference: error amplifier, oscillator, PWM comparator and the current source for the soft start. An extra delay circuit has been added to give valve amplifiers enough time to warm up before the HV supply is applied. Because valve amplifiers generally have substantial smoothing capacitors, the soft start period has been increased and the value of 100 μ F for C5 is a fair bit higher than usual.



Figure 2. The heatsink just fits on the PCB, which results in a nice compact 12.6 V module.

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Figure 3. The main parts of the inverter are the integrated regulator (IC1), transformer and bridge rectifier.



Figure 4. The PCB for the inverter is also tidy and compact.

We've used a standard ETD29 type former with N27 core material for the (home wound) transformer. The switching frequency has been kept relatively low (30 kHz) in order to save on smoothing capacitors at the primary side. Furthermore, three of them have been connected in parallel, which splits the current between them. This design can deliver a power of about 30 W.

The oscillator frequency can be set with P2 within a wide range $(\pm 7 \text{ kHz})$ to compensate for the tolerance of C4 (1 nF MKT), although the exact frequency isn't critical. To facilitate maximum power transfer, the dead time has been kept to a minimum by connecting the discharge output directly to CT and by keeping the value of C4 as small as possible.

The reference voltage is decoupled by C3 and fed to the noninverting input of the error amplifier by R5. The output of the error amplifier (COMP) is also the input of the PWM comparator, which determines the pulse width. C2 limits the bandwidth and provides stability. The 330 V output voltage is fed to the inverting input of the error amplifier via potential divider P1/R1/R2/R3,
COMPONENTS LIST

330-V converter

Resistors:

R1,R2 = $120k\Omega$ R3 = $4k\Omega7$ R4 = $470k\Omega$ R5 = $1k\Omega$ R6 = $15k\Omega$ R7 = $6k\Omega8$ R8,R9 = $680k\Omega$ R10 = $330k\Omega$ R11,R12,R13 = $2\Omega2$ R14 = 100Ω R15,R16 = $82k\Omega$ P1 = $100k\Omega$ preset H P2 = $10k\Omega$ preset H

Capacitors:

C1 = InF ceramic, lead pitch 5mm C2,C9 = I0nF ceramic, lead pitch 5mm C3,C7,C10 = I00nF ceramic, lead pitch 5mm C4 = InF MKT, lead pitch 5mm C5,C6 = 100μ F 16V radial C8 = 100μ F 25 V radial C11 = 220μ F 25V radial C12 = 4nF7 C13,C14,C15 = 2μ F2 450V radial, lead pitch 5mm, diameter 10mm C16,C17,C18 = 1000μ F 25V radial

Inductors:

L1 = suppressor coil 40µH 3A, type SFT10-30 (TDK) L2,L3 = 47mH, e.g., 2200R series type 22R476 (Newport Components)

Semiconductors:

DI-D4 = BY329-1000 (Philips) D5 = BAT85 TI = BC550C T2,T3 = BUZ11 IC1 = SG3525A(N) (ST Microelectronics)

Miscellaneous:

- KI = 2-way PCB terminal block, lead pitch 5mm
- K2,K3 = 2-way PCB terminal block, lead pitch 7.5mm
- FI = fuse 2AT (time lag) with PCB mount holder
- TRI = ETD29 (Block) * primary: 2 windings II x (3 x 0.5 mm parallel) ecw secondary: 1 winding 300 x 0.3 mm ecw
- PCB, order code **000186-1** (see Readers Services page)

* see text

where R4 determines the open loop gain. P1 is used to adjust the value of the output voltage. The range has purposely been made fairly large (theoretically 270 V to 370 V), which gives the inverter plenty of scope for use in other applications. Slightly lower or higher output voltages can be obtained by varying the number of secondary turns proportionally (e.g. 273 turns would give 300 V). Keep in mind that if you use too many turns you can still get the correct output voltage, but at a reduced efficiency because the output is peak-rectified. The surplus energy will then be lost and dissipated in the transformer.

The modest circuit around T1 provides a delay of about 45 seconds between the application of the 12.6 V supply and taking the shutdown input low; this has to be below 0.6 V to enable the inverter. C6 is charged slowly by the potential divider of

R8/R9/R10, which causes the voltage at the base of T1 to rise slowly and causes it to conduct. D5 causes C6 to discharge quickly when the supply is switched off.

R11, R12, R13 and C7-C11 are used to decouple the supply to the PWM regulator. R14/C12 reduce the spikes that are caused by the fast switching of transistors T2 and T3. The alternating voltage at the secondary is rectified by four fast soft-recovery diodes (D1-D4). Smoothing is carried out by 450 V radial electrolytics (C13, C14, C15). These are followed by low pass filters that reduce the ripple of the switching frequency even further. We've assumed that the inverter will be used with a stereo amplifier so we've provided two supply outputs, each with its own filter network (L2/C14, L3/C15). For the inductors we've used the 2200R-series from Newport Components, but the board will also accept the 8RB and 10RB series from Toko (available from Cirkit).

L1 filters the input supply and F1 protects the input supply from overload, which is important when, for example, a car battery is



POWERSUPPLY



Figure 5. This shows an exploded view of the transformer parts.

used as source. For an even cleaner supply you should thread both 330 V cables through a large ferrite bead, which reduces common mode interference.

Construction of the inverter

The PCB for the 330 V inverter is shown in **Figure 4**. Because of the high voltages present, the layout is such that there is a minimum separation of 3 mm between the HV tracks and the earth plane. It is for this reason that the wire link between the cathodes of D1 and D2 is routed away from the low-voltage section (otherwise it could have been a track between the diodes).

Populating the board is simply a matter of carefully going through the parts list and soldering the components in place. The only part that could cause problems is transformer TR1. But this isn't as complicated as it might appear, since we can use a transformer kit supplied by Block (EB29 — see **Figure 5**). This also contains a pre-cut insulating foil, which is used to isolate the three windings

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from each other. The laying of subsequent windings is made easier by the tightly placed insulating foil. The ends of the secondary windings can be covered with the supplied insulating sleeve, which reduces the possibility of shorts between the windings.

The two primary windings consist of 11 turns of three strands of 0.5 mmenamelled copper wires, which are wound in parallel (next to each other) as if it was one conductor. The first winding is made between pins 3 and 11. This should cover the former with one layer of 0.5 mm copper wire. A layer of insulating foil is placed tightly across this, after which the next primary is wound between pins 2 and 12. This too is covered with a layer (or two) of insulating foil. Both primary windings have to be wound in the same direction to make sure that they are more or less identical. This ensures that the field has the smallest possible offset, which improves the performance of the transformer.

If you make the secondary winding very carefully, it is possible to use 0.4 mm wire, which reduces its resistance (resulting in better efficiency). But note that a sloppily wound 0.3 mm winding can fill the former almost completely. The transformer doesn't have an air gap!

And finally...

Once both boards have been populated and tested, they can be mounted with a 15 V transformer in an enclosure. It might appear easiest to mount the supply and Valved RIAA Preamplifier in one enclosure, but to obtain the best quality, and to keep interference to a minimum, we would recommend that the supply and the preamplifier are mounted in separate enclosures. If you do decide to mount them in one enclosure, you should at least have a metal screen between the supply and preamplifier sections, and the distance between the boards should be made as large as possible.

When we tested for interference suppression in our lab we weren't disappointed. When the Valved RIAA Preamplifier was used in conjunction with this power supply we measured the 60 kHz component at -90 dB, which is below the noise level of a typical phono signal. The 30 kHz component caused by the field of the inverter was at a level of -110 dB. Both measurements are relative to an output signal of 200 mV.

Valved RIAA Preamplifier

using an ECL86

Design by G. Haas

In spite of CDs and digital audio signal processing, there are still people who are devoted to the vinyl phonograph record. And they enjoy their analogue treasures the most in the company of contemporary valve technology.



For nearly a century, vinyl phonograph records played an important role as a storage medium for voice and music. Although nowadays the CD has taken over the leading role, analogue records are still widely distributed. Hundreds of millions of records are present in archives and private record cabinets. These include irreplaceable documentary recordings, collectors' items and many commonplace recordings that reflect the spirit of a particular era and the technology available at that time, as well as recordings that are technically and artistically outstanding. For friends of analogue recordings, there is still a market for phonograph records, in which re-pressings or new pressings can be obtained. Due to the small quantities produced, as well as the high quality, the prices are fairly high.

In a direct comparison on a good stereo installation, a high-quality phonograph record can easily match or surpass the quality of a CD. The only disadvantages of records are the mechanical noise from the pick-up needle and the crackling caused by dust and electrostatic discharges. Consequently,



Figure 1. Schematic diagram of the RIAA preamplifier with the optional input transformer.

there is a steady demand for good equalising preamplifiers, including those based on valve technology. This is where the circle closes. Before the coming of semiconductor technology, valve technology was used for everything. This put its stamp on the recording technology and the 'sound'. If we revert to using valves to amplify the phonograph signal, we come closer to the sound experience of earlier times. That is the *raison d'être* for this construction project.

A valved opamp

An equalising preamplifier using valves cannot be compared to one built with semiconductors. Modern opamps have very high open-loop gains, so there is lots of reserve gain available for negative feedback. It is very costly and difficult to achieve equally high levels of open-loop gain with a valve amplifier.

Dual triodes or audio-frequency input-stage pentodes are usually used in these designs. However, here again we chose to take a somewhat unconventional approach and use an ECL86, as can be seen from the schematic diagram of the preamplifier in **Figure 1**. A high-gain audio triode combined with a highcurrent audio output pentode can be regarded as a valve opamp (high gain, high input impedance and low output impedance). In order to better understand the operation of this circuit, let's first look at its functions in detail.

If the negative feedback network formed by R11, R12 and C10-C13 is disconnected, the open-loop gains of the valves are determined by their associated circuitry. The cathode resistor of the pentode is bridged by a large electrolytic capacitor. This means that the gain of the pentode is equal to $A_v = R_a \times G_m$ (see the box 'Gain Calculations'). With the triode, the DC operating point is determined by R5. Capacitor C7 represents a short circuit for AC voltages, so resistors R5 and R6 are effectively connected in parallel. Due to negative feedback applied via the cathode resistor, the triode will not achieve its full theoretical gain level. Without this negative feedback, the triode would have a gain of 77 in this configuration, while the pentode would have a gain of 90. The combined gain would be 6930, which is around 76.5 dB. The actual gain of the triode is significantly less than 77, due to the negative feedback. In addition, we have to take into account the range of variation in the characteristics of individual valves. Consequently, the basic gain is set to around 35 dB at 1 kHz. This is not breathtakingly high, but it has the advantage that you do not have to use a selected valve, but instead can use any one you wish.

It is not necessary to describe the schematic in more detail, since the functions of the individual components are evident. The operating principle of the ECL86 in this configuration has been thoroughly described in the Valve Preamplifier article. The quality of the components must match what is stated in the components list. If you adhere to this rule, you will have no trouble achieving performance figures equal to those given in the technical specifications.

With this circuit, the old debate about electrolytic capacitors in the signal path will flare up once again. The main advantage of electrolytic capacitors is that they provide very high capacitance in a very small space. This makes it possible to have low-impedance signal coupling, even at very low frequencies. In addition, a lot of technical progress has been made in the construction of electrolytic capacitors in the last few years — more than

Technical Specifications

	330 V
into 100 k Ω	200 m
at 20 Hz	2 kΩ
at I kHz	I 50 Ω
at 20 kHz	25 Ω
approximately	20 m/
	into 100 kΩ at 20 Hz at 1 kHz at 20 kHz approximately

MM (source impedance 750 Ω)

THD+N	BW= 80 kHz, 1 kHz	< 0.06 %
THD+N	A-weighted	< 0.014 %
S/N	22 Hz – 22 kHz	> 65 dB
S/N	A-weighted	> 76 dB
Gain	I kHz, U _N = 3.5 mV	35 dB

MC (source impedance 25 Ω , using an R-110 input transformer)

•	 · ·	•	
THD+N		BW = 80 kHz, I kHz	< 0.07 %
THD+N		A-weighted	< 0.018 %
S/N		22 Hz – 22 kHz	> 63 dB
S/N		A-weighted	> 74 dB
Gain		$I \text{ kHz}, U_{INI} = 0.37 \text{ mV}$	55 dB

Figure A shows the deviation from the RIAA curve. The relative deviation remains within +0.8 dB / -0.5 dB with a load of 47 k Ω . Due to the high output impedance, departures from these results may occur with other loads, particularly at low frequencies.

Figure B shows the frequency spectrum for a 200-mV, 1-kHz signal with a 100-k Ω load. It can be seen that the THD+N value consists almost exclusively of the noise component. The peaks at 25 kHz and above are generated by the switching power supply (soon to appear in Elektor Electronics!). Lying at -90 dB, they are not only so low as to have no effect on the measurement, they are also far outside the range of human hearing.



with other types of passive components. We suggest that you first build the circuit and listen to the results. After this, you are free to experiment with other types of capacitors.

MM or MC?

Now we come to an interesting circuit detail at the input. If a conventional MM (movingmagnet) cartridge is used, it is connected directly to C3. Resistor R3 then provides the standard termination impedance of 47 k Ω . Resistor R2 functions only as a bleeder resistor for DC voltages, and C2 can be used as needed. A particular capacitive load is prescribed for each type of MM cartridge, in order to obtain a linear frequency response. Usually, the connecting cable of the phonograph is dimensioned so that it provides the correct load in combination with the stray circuit capacitance. If this is not sufficient, C2 must be used to provide compensation. A ceramic capacitor is usually used here, with a value ranging between 10 pF and several hundred picofarads.

If an MC (moving-coil) cartridge is to be used, it is both helpful and worthwhile to employ an audio input transformer. Moving-coil cartridges have significantly better reproduction characteristics than moving-magnet cartridges, due to their operating principle. Their disadvantage is that the output voltage is approximately a factor of ten lower. This can be compensated by using an R-110 input transformer, which is a toroidal-core transformer enclosed in Mu-metal. It boosts the low signal voltage by a factor of 10 (20 dB), with practically no noise and a very small distortion component. This type of transformer can only produce odd harmonics. The distortion factor depends on the primary voltage level and the frequency. The lower the frequency and the higher the voltage, the higher the distortion factor. The characteristics of the R-110 transformer are measured at 1 mV, which is a good standard value for MC cartridges.

The Mu-metal enclosure for the transformer is relatively expensive, but it is absolutely necessary. Without it, all induced magnetic disturbances would be correspondingly

amplified along with the signal and would degrade the signal-to-noise ratio. Using the transformer allows us to avoid trying to get the utmost out of amplifier technology, with all the associated disadvantages. In order for the input transformer to work perfectly, certain basic considerations must be taken into account. The transformer used here changes the voltage by a factor t = 10. The voltage is transformed upwards by a factor of 10, but the impedance is transformed by a factor equal to t^2 . If the transformer is terminated with 47 k Ω , for example, the termination impedance R seen by the movingcoil cartridge is (R3 / t^2) = (47 k Ω / 10^2) = 470 Ω .

In order for this low impedance to have an effect at the amplifier input, C3 must have a high capacitance. The 3-dB corner frequency lies at around 0.07 Hz in this case. In addition, the amplifier input sees the low source impedance, which has a beneficial effect on the noise behaviour. The transformer







has a screen between the primary and secondary windings, which diverts interference to ground.

In order to use a moving-coil cartridge correctly, you must observe the specifications given on its data sheet. For example, if a termination resistance of 1 k Ω is specified, R3 must be increased to 100 k Ω . In this way, the cartridge sees the required termination resistance, while the source resistance seen by the preamplifier is the internal resistance of the moving-coil cartridge. R2 can be omitted if an input transformer is used, since the transformer winding provides the DC path. The frequency response can be corrected using C2, C1 and R1. C2 represents the nec-

COMPONENTS LIST

Resistors:

(metal film, 0.7 W, 1% tolerance, MO = 5%, tolerance) RI = see text $R2 = IM\Omega$ (see text) $R3 = 47k\Omega$ $R4 = 220k\Omega$, MO, 2W $R5.R6 = Ik\Omega5$ $R7 = 4k\Omega7$ $R8 = 680 k\Omega$ $R9 = 220k\Omega$, MO, 2 W $RI0 = 560\Omega$ $RII = IM\Omega 2$ $RI2 = 68k\Omega$ RI3,RI4 = I8kΩ, MO, 4.5 W RI5 = 270Ω, MO, 2W $RI6 = 8k\Omega 2$ $RI7 = I0k\Omega$

Capacitors:

C1,C2 = see text C3 = 47μ F 35V bipolar C4 = 47μ F 450 V axial C5 = 2μ F2 400V, lead pitch 5mm C6 = 47μ F 450V axial C7,C8 = 220μ F 25V, lead pitch 5mm C9 = 47μ F 450V axial C10,C11 = 3nF3, 2.5 % polypropylene, min. 100V C12 = 1nF5, 2.5 %, polypropylene, min. 100V

Miscellaneous:

- VI(R"o I) = ECL86
- I noval (9-pin) socket, ceramic, for board mounting
- I PCB, epoxy resin strengthened, $70 \mu m$ copper layer
- I Moving-Coil transformer, type R-110, see text
- Solder pins

essary capacitive load, which depends on the cartridge employed and the stray circuit capacitance. Depending on the configuration, there may be a resonance peak in the audible range. This can be suppressed using R1 and C1, in order to make the frequency response linear. If C1 and R1 are needed, their values will lie in the range of 22 pF to 1 nF and 5 k Ω to 20 k $\Omega.$

Construction tips

The MC cartridge should be connected as symmetrically as possible

using XLR connectors. This is the only way to conduct the very small pickup signal to the amplifier input without introducing interference. Connect the screen braid to pin 1 in the XLR connector. The following small table shows how everything is arranged:

RIAA Cutting

RIAA is an acronym that stands for 'Record Industry Association of America'. This organisation has specified how phonograph records are to be cut (mastered) and reproduced, so that every record can be played on every phonograph anywhere in the world. What is the purpose of the well-known RIAA curve?

When a record is recorded, the first thing that has to be done is to decide on the groove spacing. The narrower the spacing, the more program material that can be placed on each side. The price of narrow spacing is reduced lateral excursion, which means reduced dynamic range. With a given groove width, the maximum excursion must be limited, since otherwise groove overcutting will occur. This is especially true in the low frequency range (below around 500 Hz), where the largest amplitudes appear.



A phonograph record is cut using an electrodynamic process. The signal currents for the individual channels are passed through separate coils that drive the cutting stylus, similar to how a dynamic loudspeaker is driven. A spring/mass system is driven by a coil/magnet system. The driven cutting stylus is free to move along only two axes. The impedance of the coils is frequency-dependent. If constant-amplitude drive is used, the cutting velocity rises with increasing frequency (see **Figure A**). With constant-cutting-velocity drive, the amplitude drops with increasing frequency (see **Figure B**.) Since the frequency range that must be handled runs from 20 Hz to 20 kHz, the amplitude ratio with constant cutting velocity is 1:1000 (60 dB).

Such an extremely large dynamic ratio would mean that no usable signal-to-noise ratio would be left at high frequencies. For the optimal utilisation of the surface area of the record, constantamplitude recording is ideal, but this means that the cutting velocity rises with increasing frequency. When the record is played back using a coil/magnet system, the law of induction thus says that the



output voltage will rise with increasing frequency. Figure C shows the RIAA recording curve with its corner frequencies and associated time constants. Constant-amplitude recording is used in the frequency range between f1 and f2, in order to limit the maximum excursion and thus avoid groove overcutting. This can be considered to be the same as bass attenuation. In the region between f2 and f3, the behaviour of the coil/magnet system is taken into account and constant-velocity recording is used. Above f3, up to the end of the range, constantamplitude recording is again used. The net result is that the bass frequencies are attenuated enough to avoid groove overcutting, while the surface area of the record is efficiently exploited. The mid-range frequencies are handled neutrally in comparison to the low frequencies, while the high frequencies are emphasised, which leads to a significant improvement in the signal-to-noise ratio. In the frequency range from approximately f3 upwards, the human ear has increased sensitivity to noise components. During playback, the high frequencies are reproduced too loud, so they must be attenuated. This is precisely the effect that improves the signalto-noise ratio. The bass frequencies, by contrast, must be emphasised, which leads to increased sensitivity to induced mains-frequency signals and their harmonics. If an equalising preamplifier can maintain the complementary reproduction curve within less than ± 1 dB, it is considered to be high-end equipment. The numerical values in the table, which represent the RIAA curve with reference to I kHz = 0 dB, are a useful aid for making measurements.

Hz	dB	Hz	dB
20	+19.3	800	+0.7
30	+18.6	lk	0.0
40	+17.8	I.5k	-1.4
50	+17.0	2k	-2.6
60	+16.1	3k	-4.8
80	+14.5	4k	-6.6
100	+13.1	5k	-8.2
150	+10.3	6k	-9.6
200	+8.2	8k	-11.9
300	+5.5	l 0k	-13.7
400	+3.8	15k	-17.2
500	+2.6	20k	-19.6

Standard lead colour	XLR pin	RIAA- Preamp
white	2	a left
blue	3	b left
screen	I	m left
red	2	a right
green	3	b right
screen	I	m right

If you do not use a moving-coil input transformer, you should use a quasisymmetric lead arrangement with Cinch connectors, instead of the usual asymmetric arrangement. Connect the screen and the blue or green lead together inside each Cinch connector. This keeps the screens free of signal voltages in this case as well.

The amplifier circuit is relatively simple, but it has very good basic characteristics. Very high-quality construction is absolutely necessary if these are to be fully realised. The amplifier circuit is designed to be monaural, so separate printed circuit boards (as shown in Figure 2) are needed for the two channels. The circuit board is not available from Readers Services, but it can be obtained from the author. Mount the two circuit boards in a generously sized, screened enclosure, well separated from interference sources. A high degree of channel separation can be maintained by spacing the two boards widely apart, or by placing a sheet-metal screen between the two boards if space is tight. This makes a lot of difference to the sound. The standard requires >26 dB at 1 kHz, which lies within the realm of what is possible.

The power supply does not belong in the amplifier enclosure. Only the amplifier boards are mounted in the non-magnetic metallic enclosure, while the power supply is housed externally and placed at a sufficient remove from the equalising amplifier. The well-filtered, floating supply voltages (filament and high voltage) are fed to the amplifier enclosure via separate cables. This avoids the superimposition of the filament and high-voltage currents, which could increase the base noise level. The input and output sockets of the amplifier are mounted isolated from ground. The negative poles of the high voltage and filament supplies are connected together, along with



the enclosure and signal ground, at a single point. This results in a nongrounded power supply and avoids hum loops. If the signal leads in the phonograph are isolated from the chassis, the phonograph chassis must be connected to the amplifier enclosure via the lead provided for this purpose (black).

The ECL86 has a filament voltage of 6.3 V at 0.66 A. The filaments must be connected in series to match the DC filament voltage of 12.6 V.

Once you have carefully built this preamplifier, you are ready to enjoy unimpeded listening pleasure. Even old monophonic records will sound distinctly better, since a stereo cartridge has significantly better tracking characteristics than a mono cartridge. The value of your record collection will be distinctly enhanced by this low-noise, low-distortion amplifier using primarily K2 components.

(000016-1)

Additional information is available from:

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Gain Calculations

The gain of a triode is determined by its transconductance, its internal resistance and the value of the anode resistor that is used. The gain formula is $A_v = R_a \times g_m$, where g_m is the dynamic transconductance. The values of the static transconductance and internal resistance can be obtained from the valve data book. The value of the dynamic transconductance, which ultimately determines the gain of the triode circuit, must be calculated using the formula

$$g_m = G_m \cdot \frac{R_i}{R_i + R_i}$$

g

where

g _m	=	dynamic transconductance
G _m	=	static transconductance
R _i	=	internal resistance
R _a	=	anode resistance

For pentodes, the formula is $A_v = R_a \times G_m$. A tolerance of 5% for the anode resistor is considered to be very precise, since many parameters of active components often shows a wide range of variation. For example, if the data book gives a value of 1.6 mA/V for the transconductance of an ECC83, a variation of $\pm 30\%$ is easily possible. The internal resistance amounts to 62.5 k Ω . You can easily calculate how much the actual gain can vary for a given value of the anode resistor. The ECL86 consists of half of an ECC83 together with an audio output pentode, which has a transconductance of 10 mA/V.