Remote control for a hi-fi system

Infra-red transmitter and receiver for switching and controlling a stereo pre-amplifier.

by **Steve Kirby**

Built by the author around the Advanced Preamplifier of Douglas Self, this infra-red remote control system may be easily added to most hi-fi amplifiers. It offers controls for input selection, volume, balance, treble and bass and can switch between loudspeakers and headphones. It eliminates the need for front panel controls.

I have used an earlier version of this circuit for a year, and it has performed satisfactorily. The circuits are fairly easy to add to existing pre-amplifiers. My system presently consists of the Advanced pre-amplifier by D. Self (*Wireless World*, November 1976) with the loudspeaker system by S. Linkwitz (*Wireless World*, May, June and December 1978) and five 25W power amplifiers. Signal metering is by the Q. Rice L.e.d. p.p.m. meter (*Wireless World*, August 1980).

The present circuit was developed to replace the original and give additional facilities, such as balance and tone controls, and to remove the duplication of the remote functions by the front panel controls. In one year's operation I have never used the panel controls, so the second version will have a completely blank front panel.

The remote transmitter

The transmitter uses a Plessey SL490 i.c. which has all the key reading, command encoding and output drivers stage integrated on a single chip. The only discrete components needed are for timing and for driving the transmitter infra-red l.e.ds with short high-current pulses (Fig. 1).

The i.c. has provision for transmitting 32 possible codes and there is a family of receiver i.cs which can use all of them. The ML922 used in this system only responds to 21 codes. Because of this the crosspoint matrix keypad has to be divided into two sections, giving 24 keys, one with no function and two with duplicate functions to adjacent keys. The same transmitter could be used for other receivers in the series (including a microprocessor-compatible 5-bit receiver and a toy controller) and include the full 32 keys, in a 4 × 8 matrix.

A variable resistor adjusts the internal clock which controls the command repetition rate and sets the time to go from 'off' to 'full volume'. The output stage provides a 1A pulse to give a high power infra-red signal. To prevent the l.e.ds from being destroyed, the 3n3 capacitor is discharged through the l.e.ds via the transistors. The green l.e.d indicates that a key has been pressed.

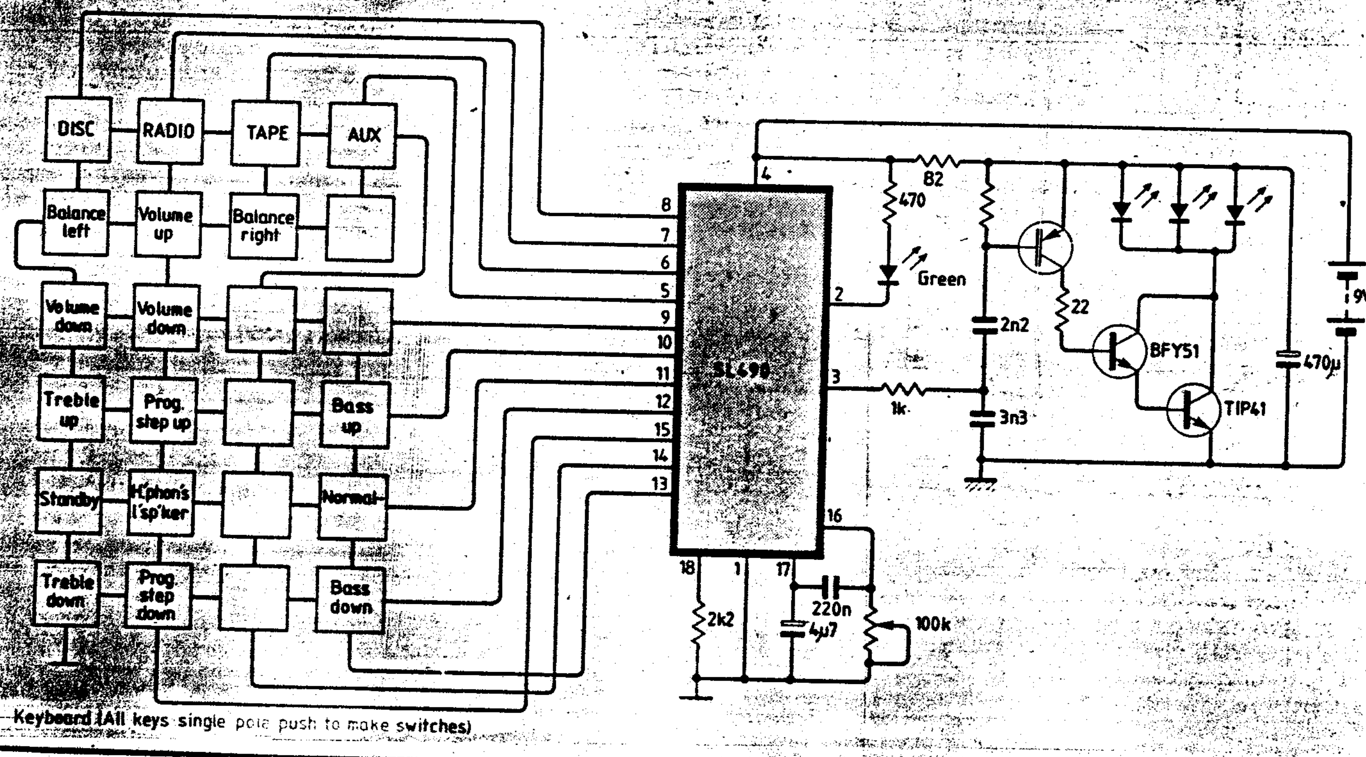
The receiver

The receiver is based on a Plessey ML922 i.c. which decodes the p.p.m. signal from the SL480 i.r. preamplifier. It gives a four-bit digital output and ten different codes are used. There are also three analogue current outputs produced by the internal 5-bit d-a converters.

Referring to Fig. 2, the infra-red pulses are detected by the receiver diode. Noise from visible light is filtered out by a Kodak infra-red transmitting filter, type 87C. The SL480 is a high gain (100dB) preamplifier specifically designed for this application. It can suffer from oscillation caused by feedback, so layout is rather critical and an electrical screen around the preamp section is essential, along with good supply decoupling. The p.p.m. output is decoded in the ML922. The standby output, which is low until the first command is received, presets the up/down counters and resets the four-bit latch.

When a digital code is received, the pulse line triggers the 555 mono-stable which expands the pulse to drive the green l.e.d. ('command received' indicator). The digital outputs have a +15V '1' level, which is buffered by the 4049 inverters to give a 0-5V t.t.l.-compatible output to drive the r.o.m.

Fig. 1. The remote control transmitter



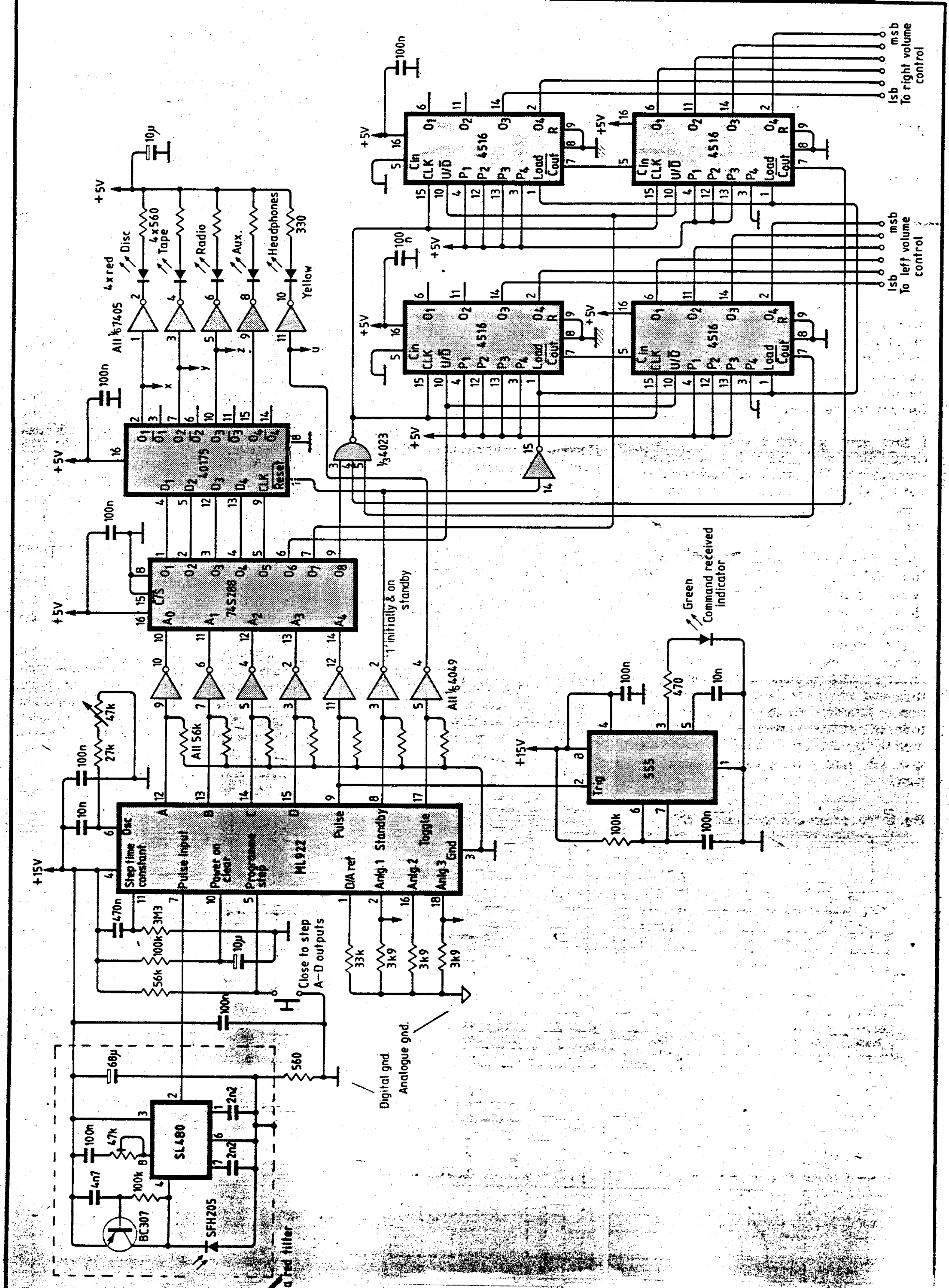


Fig. 2. *The remote control receiver and decoder*

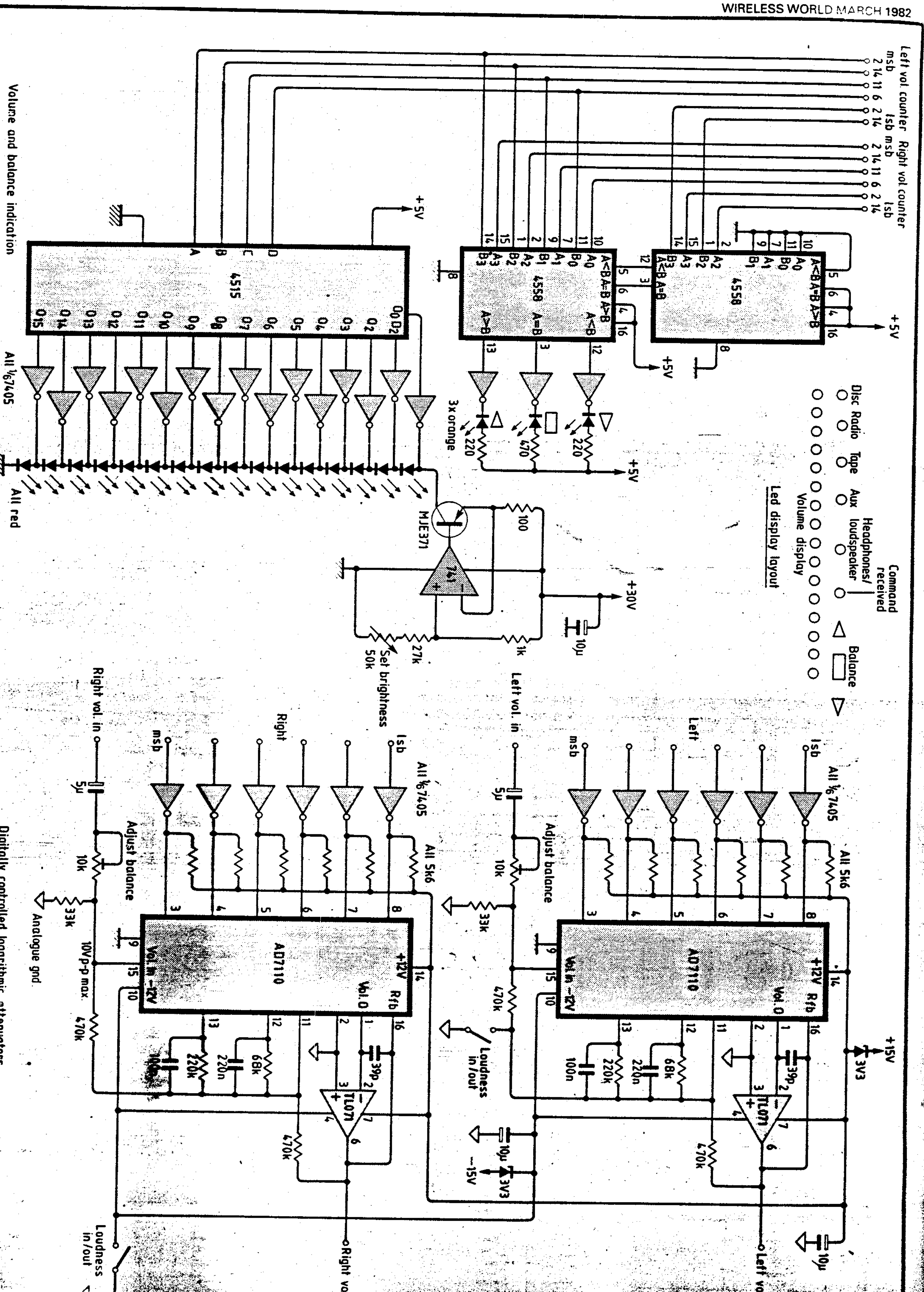


Fig. 3. Volume and balance control and display

The analogue current outputs at pins 2, 16 and 18 are converted into voltages by the associated resistors with a step size set by the resistor at pin 1. There are transmission codes to step them up or down, or to go back to a preset level, (3/8 f.s.d). In our application they are used to drive the tone controls (Fig. 4).

The digital word is decoded by a 5 × 8 p.r.o.m. (74S288). A '1' on one of the four first p.r.o.m. outputs is latched by a pulse on the fifth output. When a input or headphone/speaker command is received it is displayed on the appropriate l.e.ds, driven by the latch.

The remaining three r.o.m. outputs drive the up/down counters which are used to drive the 'volume control' attenuators in Fig. 3. They are counted up or down in parallel or in opposition to control both volume and balance. There is a 'standby' command to return the volume to a set level.

Volume and balance display

When the volume control knob is removed there is no longer any visual indication of the volume setting. For many people this may not be a problem; all that matters is how loud the music is! Others may like a display to set up a specific level for comparison. A balance indication can also be useful to answer such questions as to whether the second violins are really trying to drown the double basses.

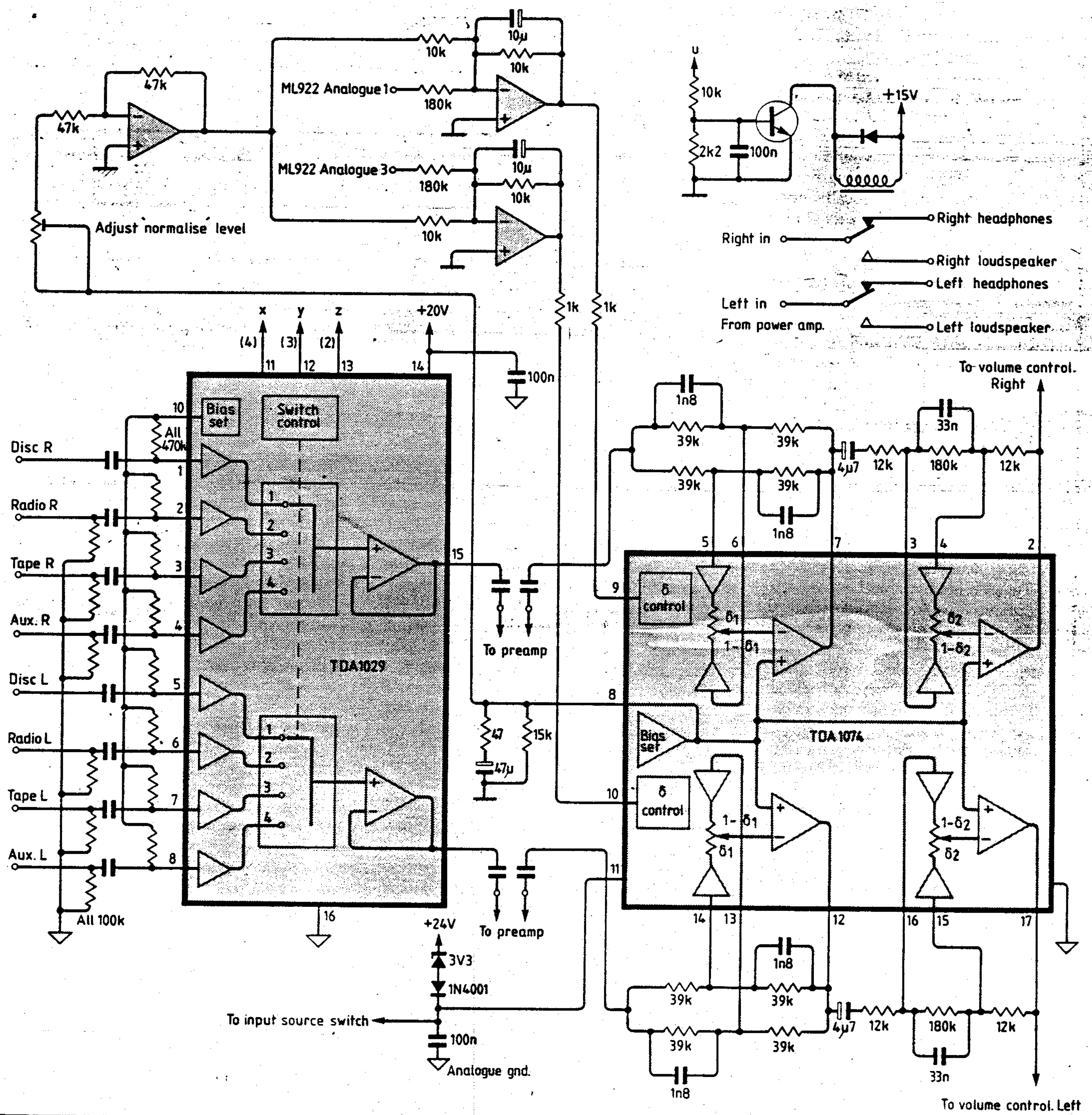
The circuits on Fig. 3 give simple indication of volume and balance settings. A 4-to-6 line decoder is driven by the top four bits of the left volume control counter.

The outputs turn on the open collector inverters sequentially as the volume is increased. With only the first inverter on, all the current from the source, the 741, flows only through one l.e.d. As lower inverters are turned on, the current flows through extra l.e.ds before being diverted to ground. Power comes from the unregulated supply on the preamplifier.

Balance indication is by three l.e.ds, a central square one flanked by two triangular ones. The two sets of counts are compared and depending on whether the right counter is greater than, less than or equal to the left, the appropriate l.e.ds are lit.

More precise displays of volume or balance are possible but they consume a great deal more power, chips and money! Either or both displays could be omitted with no loss in the system performance.

Fig. 4. *Input switching and tone control*



Digital volume control

The logarithmic attenuators shown on diagram 3 are multiplying digital-to-analogue converters specifically designed for audio volume control. The attenuation is set by a six-bit word, with 59 steps of 1.5dB from 0dB to −85dB. Steps 60 to 64 give complete isolation between input and output. T.h.d. is better than −85dB over the audio range, depending mainly on the quality of the op-amp used in the output feedback loop.

Low volume loudness frequency compensation is provided but can be switched out. The attenuator has a current output, converted to a voltage by the output op-amp and an internal feedback resistor. The attenuator is powered from ±12V, derived from the ±15V regulated supply of the audio preamplifier. If the loudness function is not required, the −12V supply can be omitted. The 7405 inverters are used to change a logic '1' from 5V to 12V for driving the attenuators.

When the attenuator is turned off, capacitive feedthrough, due to the board layout is the main source of noise, so input and output should be carefully separated in the layout. Separate analogue and digital grounds are provided. Switching noise as the volume is changed is practically inaudible, and certainly better than a good many log pots I've used!

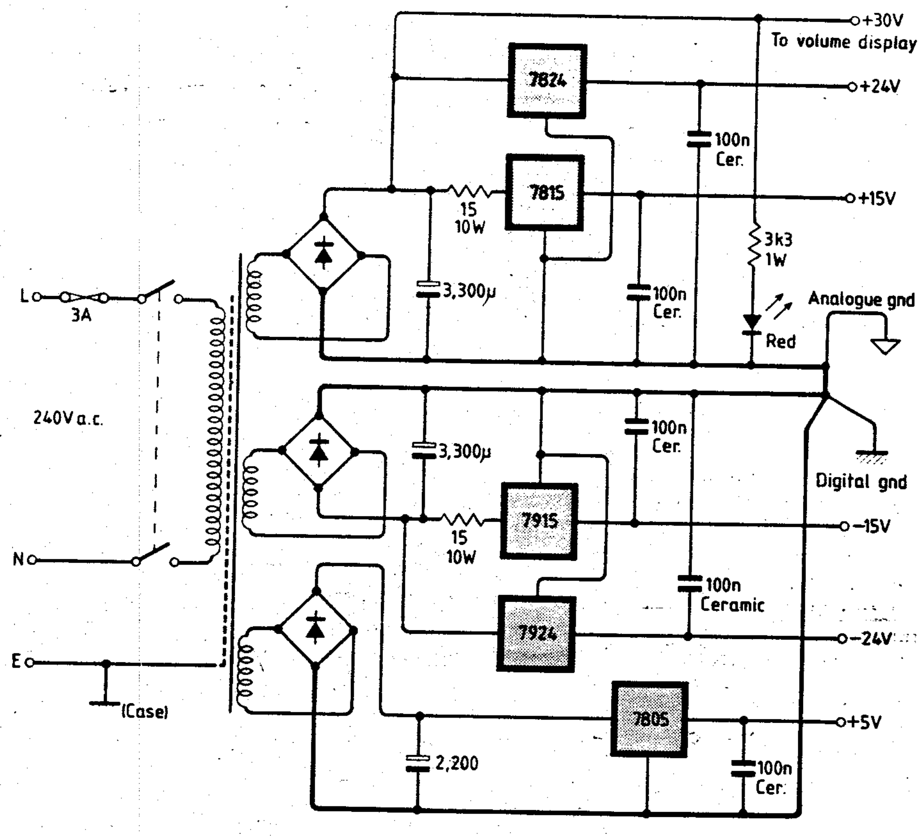


Fig. 5. *A suitable power supply*

Input switching

The low level inputs can be switched either by good quality relays, or electronic switches designed for audio. In an earlier design I used relays, which worked well, and the audible click as they change was a reassuring feedback. Audibly there is no difference (except that there is now no click). They are cheaper than four relays.

On Fig. 4, a four-to-one stereo switch is driven by the latch on Fig. 2. The input capacitors selected depending on the impedance of the source driving them. The disc input comes from the magnetic cartridge preamp. Resistors bias the inputs to half the power supply level for maximum range.

In the Self preamplifier, there are additional switches around the feedback network, selected according to the source. I have kept these as relays because of the large voltages that could be passing. They are driven in parallel with the input switches.

Tone control

Tone is not generally altered as often as volume, so for economy this section could be omitted, retaining the ordinary front panel controls. A Mullard TDA1074 is used in the remote controlled tone section. The i.c. contains four voltage-controlled potentiometers connected to high-gain op.amps. Frequency dependant feedback networks are connected around these to provide treble boost/cut of ±12dB at 10kHz and bass boost/cut of ±12dB at 100Hz.

The voltage controlling the potentiometers is relative to an internal bias, set to half of V_{cc} The analogue outputs from the Plessey receiver/decoder are summed with the bias voltage by the two op.amps. Each time the system is switched on, the analogue signals are 'normalised' to 3/8ths of f.s.d. This can be set to give a level response by the adjustment of the preset potentiometer. However, because the level is set at 3/8ths f.s.d. less room for cut and more room for boost is available. I haven't provided any indication of tone settings; I prefer my ears to do that, but the l.e.d bargraph i.c. from National Semiconductor could be used.

The system as a whole gives an output noise voltage of about 50μV and a total harmonic distortion of 0.05%, channel balance is within 0.5dB and channel separation 60dB. The dynamic range is not quite as good as Mr Self's original design.

Construction

The receiver and decoder were built on a Veroboard with an edge connector. The system is built around a mother board for future up-dating. The transmitter was built in a standard Verobox and a keyboard taken from a calculator. Layout is not critical except where already mentioned.

The components are readily available except for the p.r.o.m. which needs to be programmed. Many companies offer a programming service[1]. If all the sub-circuits are built the total component cost would be about £70, though various displays and the tone control section could be left out if the constructor wishes to reduce costs.

Future development of the system will replace the logic with a single chip microprocessor. Programmes for the remote control of solenoid operated cassette decks and synthesised radio receivers could then be easily configured.

Power supply

The power rails necessary will be found in many modern stereo systems with the probable exception of the +5V. This could be provided by a small extra transformer and a 1A regulator if no existing transformer taps will do. For those starting from scratch, Fig. 5 gives a standard power supply unit. Note the connections of grounds.

[1]Programmed r.o.ms may be obtained from Mr G. Long, York Electronic Centre, Department of Electronics, University of York, Heslington, York. The cost includes a programming and handling charge and will be about £4 in total.