Anti Phaser

The Anti Phaser is a phaser with a difference: the notch filters sweep in opposite directions. Suitable for any amplified instrument.

By D. Bradshaw

DESIGNING a notch filter is not difficult, and there are several standard configurations to choose from. The situation is rather more difficult where a tunable notch filter is required, and much more difficult in an application such as this where the filter must be voltage controlled so that it can be swept by a low frequency oscillator. The main difficulty is that notch filters tend to require that several filter resistances or capacitances are varied in value and remain accurately matched in order to give usable results over a wide frequency range.

There are two conventional solutions to the problem. The most common one is to use a series of voltage controlled phase shifters, and to then mix the shifted and unshifted signals. At frequencies where the signals are out of phase they tend to cancel each other out and produce the required notches of high attenuation in the frequency response. Two phase shifters per notch are required, and matching of the two voltage controlled resistors (usually JFETs) is unnecessary. The other method is essentially the same but uses a different system to obtain the anti-phase signals. Rather than phase shifters an analogue delay line is utilised.

These days there are practical alternatives to the conventional systems, and it is one of these that is used in this unit. It is based on four operational transconductance amplifiers, two being required for each notch filter. **Fig. 1** shows the block diagram for the Anti-phaser.

Transconductance amplifiers can easily be used to act as bandpass, lowpass, or high pass filters, and it is the bandpass configuration which is used here. This is, of course, the exact opposite of what is required here, but a bandpass

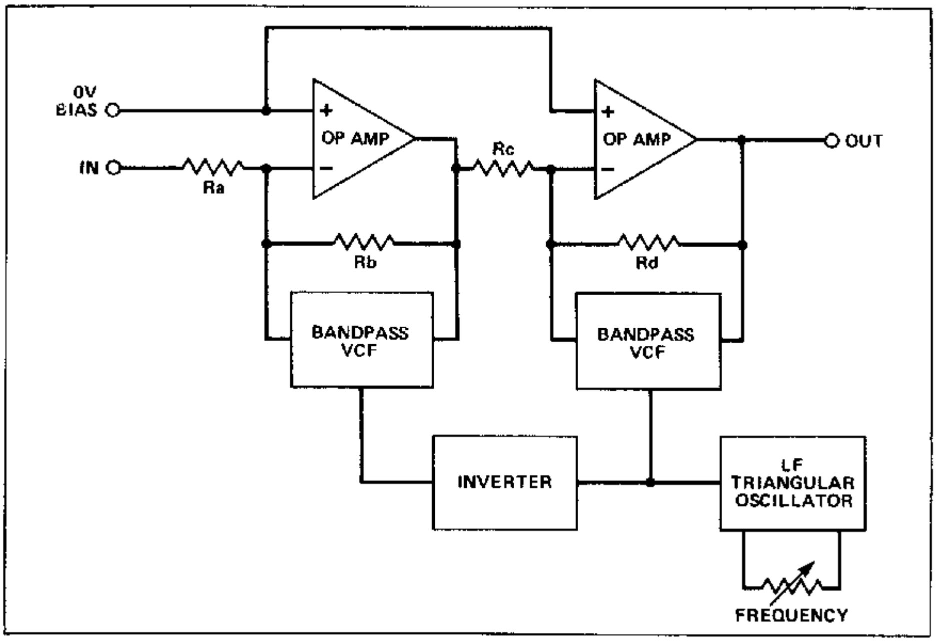


Fig. 1 The block diagram of the Anti-Phaser. This is for one notch filter. Two are used.

response can be converted to a notch type simply by using the filter in the feedback path of an inverting mode operational amplifier circuit. **Fig. 2** shows the basic inverting mode circuit, and the two resistors control the voltage gain of the circuit. The voltage gain of the operational amplifier itself is extremely high, being typically about 100,000 times at low frequencies. What it is actually amplifying is the voltage difference across its two inputs, with the output going positive when the non-inverting (+) input is at the higher voltage, or negative when it is at a lower voltage than the inverting (-) input. Due to the very high voltage gain of the device only a very small voltage difference of typically under 1 millivolt is needed in order to send the output fully positive or negative.

In the circuit of **Fig. 2** the negative feedback from the output to the inverting input results in the inverting input being maintained at the same potential as the non-inverting input. If an input signal was to take the inverting input slightly positive, this would unbalance the inputs and send the output negative. The coupling through R2 would result in the inverting input being taken negative, counteracting the positive input signal and maintaining the balance. Just how negative the output has to go for a given input voltage depends on the values of R1 and R2. If we

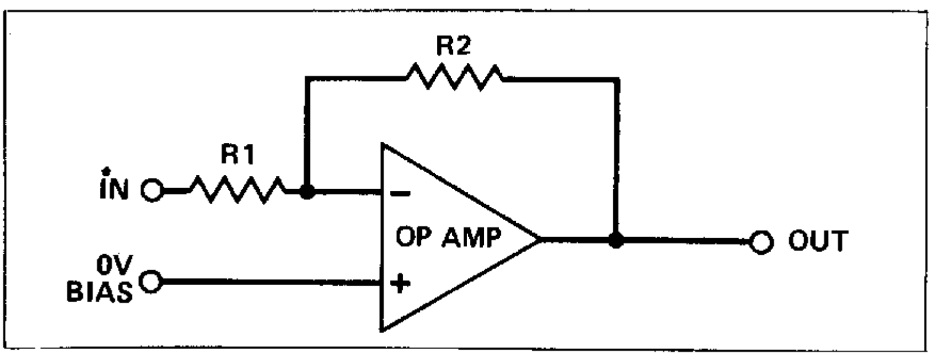


Fig. 2 The basic circuit which inverts the bandpass response into a notch type response.

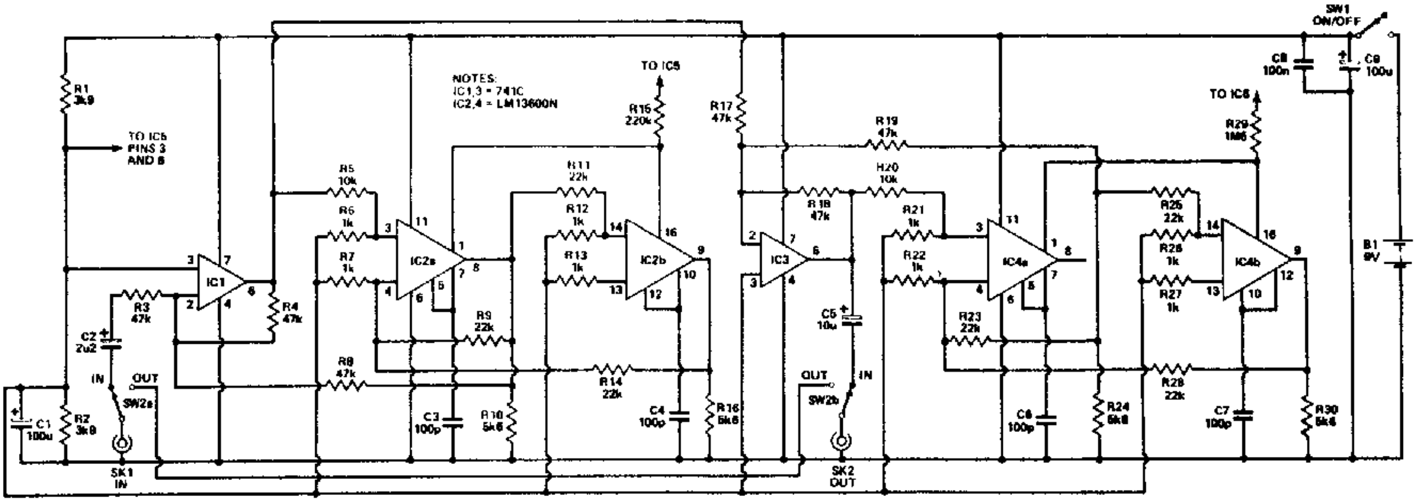


Fig. 3 The signal processing stages of the circuit. The two notch filters each use two ICs and are identical.

assume that they have the same value then any increase in voltage at the input will produce and identical but opposite change in the output voltage. If R2 is made higher in value than R1 it becomes necessary for the output voltage to change by a larger amount in order to maintain the input voltage balance by what is really just a simple potential divider action.

What is of greater importance is this application, if R2 is made lower in value than R1, the output has to change by a smaller amount than any change at the input in order to maintain the balance. The circuit can therefore provide both voltage gain and attenuation, depending on the feedback resistor values. The voltage gain/attenuation of the circuit is simply equal to the value of R2 divided by the value of R1. For instance, if R1 and R2 had values of (say) 100k and 1k respectively, this would give a gain of 1/100, of some 40dB of attenuation if you prefer.

Returning to the block diagram of **Fig. 1**, each bandpass filter is connected in the feedback path of an inverting mode operational amplifier circuit. In effect the bandpass filters replace R2 of **Fig. 2.** At pass frequencies the filter provides a low resistance, and consequently a very low level of voltage gain. Outside the passband it provides a very high feedback resistance and therefore a very high voltage gain. In this application we only require a low level of gain within the filter's passband, and about unity gain rather than high gain at other frequencies. Rb and Rd are therefore used to limit the feedback resistance to a suitable level so that the response of the circuit as a whole is tamed to an acceptable degree.

The sweeping of the filters is accomplished using a low frequency oscillator having a triangular output waveform. The sweep frequency is continuously variable from about 0.1Hz to 10Hz. One of the notch filters is driven direct from output of the oscillator, but the other is driven by way of an inverter stage which provides the anti-phase operation of the second filter.

Circuit Operation

Fig. 3 shows the circuit diagram of the signal processing stages while the oscillator/inverter circuit diagram appears in **Fig. 4.**

Taking **Fig. 3** first, R1, and R2 and C1 provide a centre tap on the supply rail, effectively giving dual 4V5 supplies from the single 9 volt input. The circuit is in fact powered from a 9 volt battery and the current consumption is only about 7 miliamps.

The two notch filters are essentially the same, one utilizing IC1 and IC2 while the other is built around IC3 and IC4. We will consequently only consider the operation of the first filter. The two transconductance amplifers of IC2 are connected in a standard transconductance amplifier state variable filter configuration. Lowpass filtering is produced at the output of IC2b, and the bandpass filtering needed here is obtained at the output of IC2a. This bandpass filtering is connected in parallel with feedback resistor R4, which is the equivalent of Rb in **Fig. 1.** The two transconductance amplifiers are used here as voltage controlled resistors which act as lowpass filters in conjunction with filter capacitors C3 and C4. Feedback through R9 and R14 is used to convert the response at the output of IC2a to the required bandpass type.

SW2 provides a means of bypassing the unit and switching out the effect. In practice this is a heavy duty push button switch so that it can be operated by foot.

Turning to **Fig. 4** now, IC5 is used as a conventional triangular/square-wave oscillator with IC5a acting as the Miller Integrator and IC5b being used as the

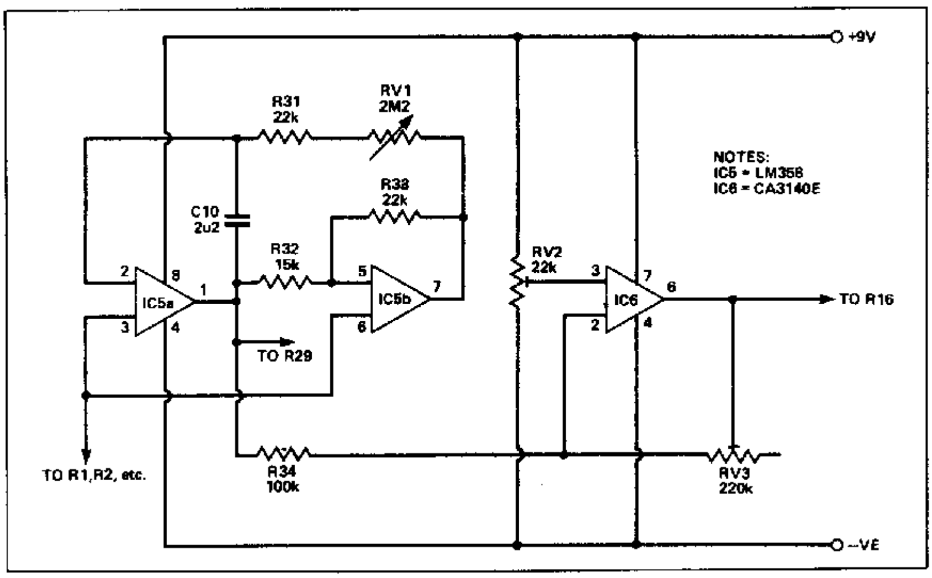


Fig. 4 Part of the circuit, showing the operation of IC5, the L.M. 358.

Schmitt Trigger. RV1 controls the charge/discharge rate of timing capacitor C10 and it acts as the sweep frequency control. IC6 is used as the inverter stage, and its voltage gain can be varied by means of RV3. The bias voltage at the non-inverting input can also be adjusted with RV2 providing the adjustable voltage. This enables the sweep range of the first filter to be set so that it exactly matches the sweep range of the second filter.

Construction

There are few difficulties here, but do not overlook the four link wires. Also, IC6 is a MOS input type and therefore requires the usual MOS antistatic handling precautions to be taken. Use an 8 pin DIL IC socket for this device, do not fit it into place until the unit is otherwise finished, and handle it as little as possible.

C10 must be good quality non-polarised component (ie not an electrolytic or a tantalum type), and the printed circuit board is designed to take a printed circuit type having 15mm lead spacing. However, it should be possible to fit other types onto the board without too much difficulty, but the component used in the C10 position must be physically quite small.

When all the link wires and components have been fitted onto the board connected the battery clip and fit pins at the points where connections to the other off-board components will be made.

Case

SW2 and RV1 are mounted on the top of the case (which in this case is the panel opposite the removable one). SW2 must be a heavy duty component since an ordinary push button type is unlikely to stand up to foot operation for long, SK1 is mounted slightly high of a central position on one of the long sides of the case, and SK2 is fitted opposite this on the other long side panel. On the prototype SK1 is a type which has DPDT contact, and two of these contacts are used as make types which act as on/off switch SW1. The unit is therefore automatically switched on when a plug is inserted into SK1, and switched off again when the plug is removed. This is a quite common practice with musical effects units, but a separate switch can of course be used for SW1 if preferred.

After the hard wiring has been completed the printed circuit board is mounted inside the case. There are printed circuit guide rails built into the case, but the board will not fit directly into these. It must be fitted into the guide rails via a set of four plastic adaptors which enable it to be mouted at right angles to the rails. This adaptors are not normally supplied with the case and must be purchased separately. The board must be mounted as low down in the case as possible so that it is not obstructed by the sockets or controls. An alternative method of mounting the board would be to simply bolt it to the base panel of the case, using short spacers to keep the connections on the underside of the board clear of the metal panel. There are plenty of small spaces on the board where mounting holes can be drilled.

There is sufficient space for a 9 volt battery at one end of the case, and some foam material can be used to trap the battery firmly in place there. Four small cabinet feet are fixed to the base panel of the case to prevent the unit from slipping when SW2 is operated.

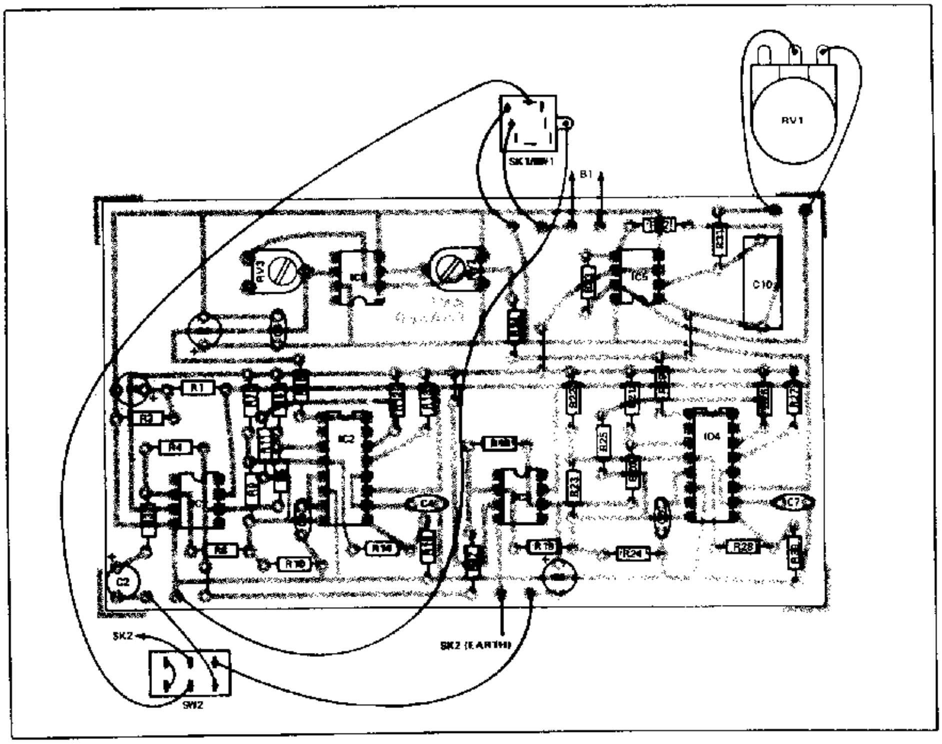
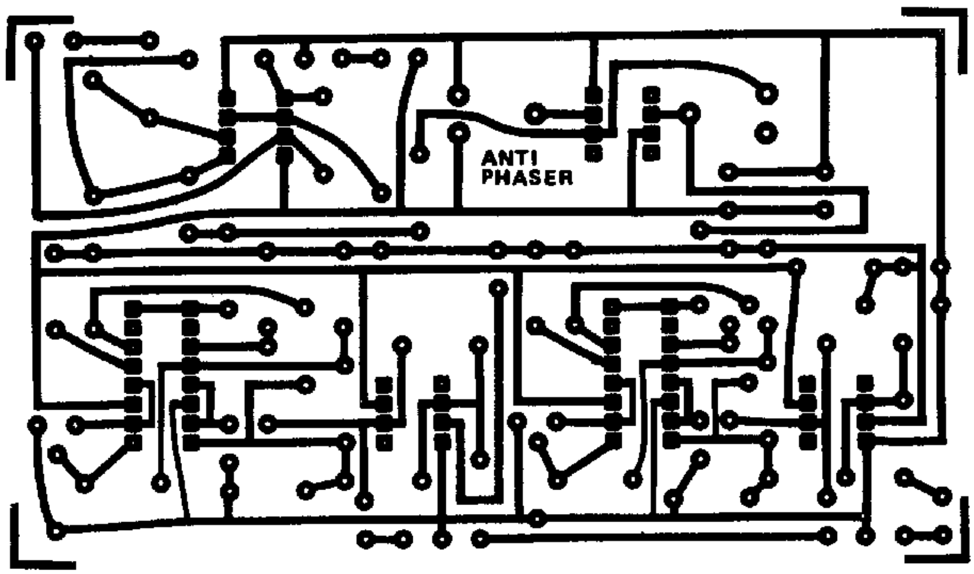


Fig. 5 The PCB layout. Assembly is straightforward, but see the note in the text about C10.



In use

The electric guitar or other instrument connects to SK1 by way of the usual screened jack lead, and as explained previously, the unit is automatically switched on and off as the plug is inserted into and withdrawn from SK1. The output signal is taken from SK2, and again, a normal screened jack lead is used to take the output to the amplifier (or whatever).

RV2 and RV3 should be set at a roughly mid setting initially, and the unit

PARTS LIST

Resistors
(All ¼W 5% carbon)

R1, 2	3k9
R3, 4, 8, 17, 18, 19	47k
R5, 20	10k
R6, 7, 12, 13, 21, 22, 26, 27	1k
R9, 11, 14, 23, 25, 28, 31, 33	22k
R10, 16, 24, 30	5k6
R15	220k
R29	1M5
R32	15k
R34	100k

Potentiometers

RV1	2M2 linear
RV2	22k 0.1W horizontal preset
RV3	220k 0.1W horizontal preset

Capacitors

C1, 9	100uF 10V radial elect
C2	2u2 63V radial elect
C3, 4, 6, 7	100pF ceramic plate
C5	10uF 25V radial elect
C8	100nf ceramic
C10	2u2 polyester or carbonate

Semiconductors

IC1, 3	741C op amp
IC2, 4	LM13600N or LM13700N dual transconductance amp
IC5	LM358 dual op amp
IC6	CA3140E MOS op amp

Miscellaneous

SK1	Standard jack with DPDT contacts
SK2	Standard jack
SW1	Part of SK1
SW2	Heavy duty DPDT push button switch
B1	9 volt

150 by 80 by 50mm diecast aluminium box; printed circuit board; control knob, four guide rail adaptors; battery connector; for 8 pin DIL IC holders; two 16 pin DIL IC holders; Veropins, wire, etc.

should then provide the anti-phase phasing effect. However, the two notches will almost certainly cover different frequency spans and will not exactly complement one another. By adjusting RV2 and RV3, using a process of trial and error, it should be possible to set the range of the first filter so that it matches that of the second. This adjustment is probably most easily accomplished with RV1 adjusted to give a middle modulation frequency of about 1Hz. It is also easier if the input signal contains a wide range of frequencies (a low frequency squarewave or a noise signal for example) since the phasing effect is then most apparent. Of course, you can use any settings for these two presets that given an effect you like, and you do not have to adjust them for matched sweeps.

The sweep range of the first filter should be satisfactory, but if necessary the range of frequencies covered can be moved up by making R29 lower in value or shifted down by using a higher value resistor here. ■