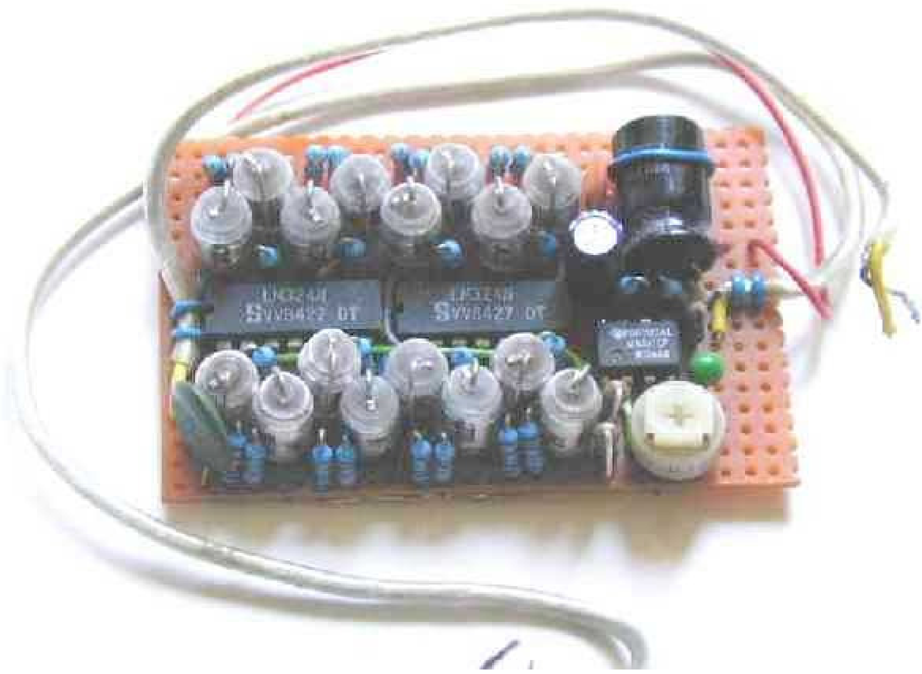
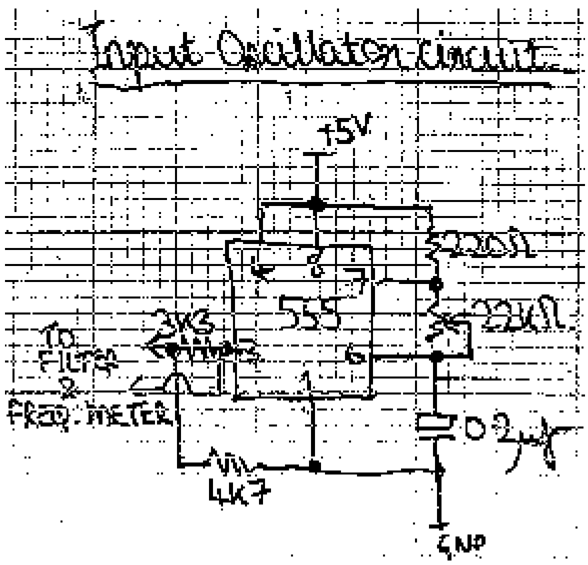
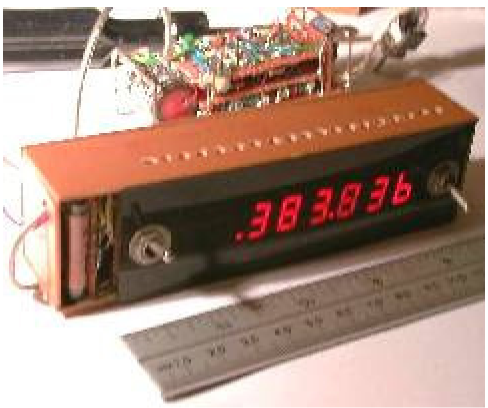
Phase-Shifting CW Filter



I saw this article about an unusual type of CW (morse code) filter in the university library copy of the August 1988 issue of Radio Communication, the monthly publication of the Radio Society of Great Britain. The article is reproduced here by permission of the RSGB.





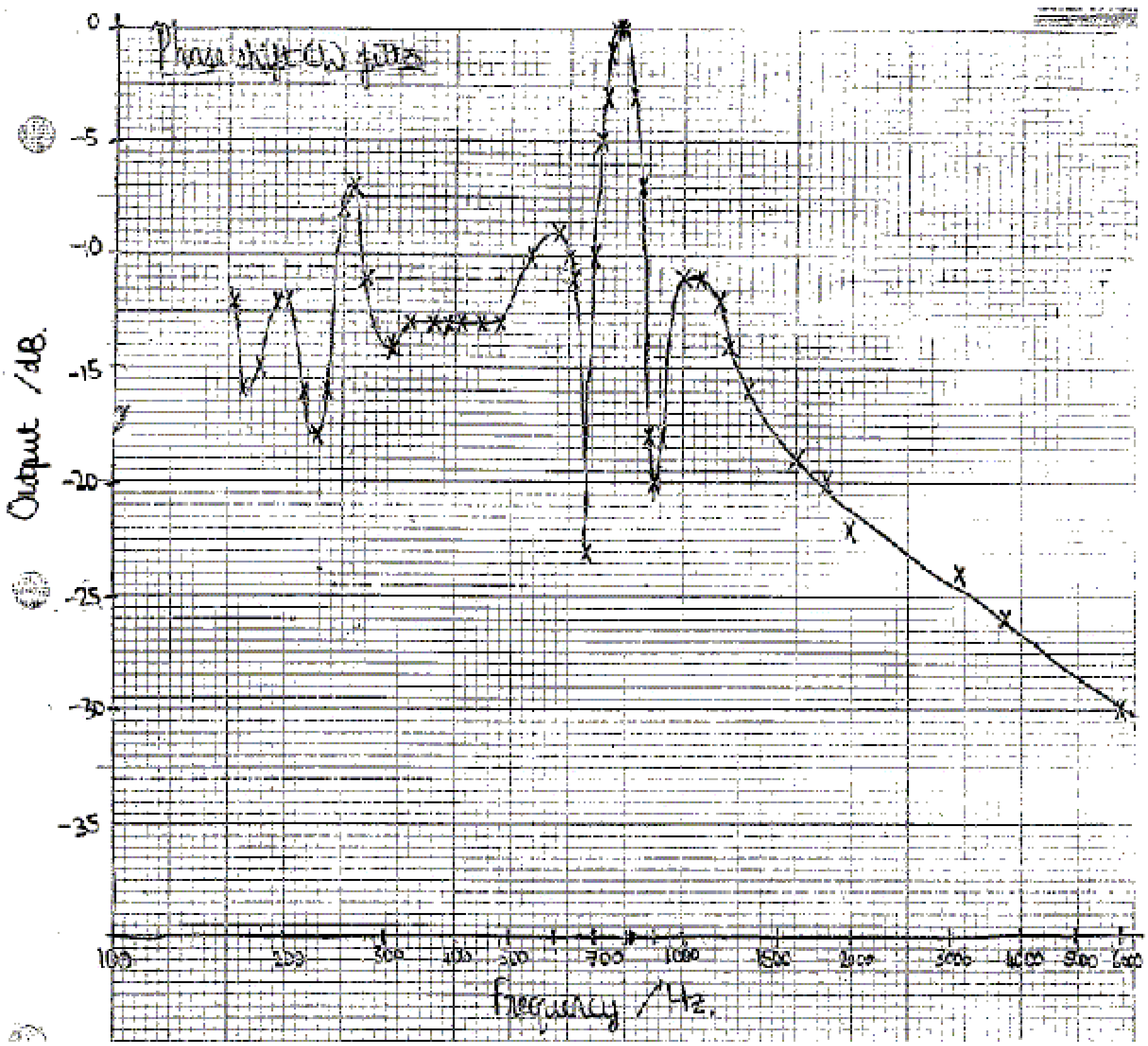
It seemed like a good idea, and so I decided to make it.

The photograph at the top of this page shows the results of my efforts. I used LM324 quad op-amps instead of the TL074 in the article, with a 741 for the final adder. I didn't build the audio output stage because I planned to feed the output into another amplifier. I used high quality polystyrene capacitors and 1% tolerance resistors. When I had finished making the circuit I felt I needed to test it, so I built the simple test oscillator shown in the circuit above on the right. This is a simple oscillator circuit using the well known 555 oscillator. I could vary the 22K resistor to set the frequency, which I measured with my recently-completed panel mounting frequency counter. The signal strength at the input and output of the filter I *think* I measured using an old digital AC voltmeter.

The results of testing my version of the circuit are shown in the table and graph below. The peak between 200Hz and 300Hz is caused by the square wave output of the test oscillator. The third harmonic of the fundamental frequency at 271.67 Hz occurs at the filter centre frequency of 815 Hz, which is passed through the filter. Ignoring this spurious peak, my test results look very similar to the graph plotted in the article.

f=:	dB
162	-12
169	-16
180	-15
193	-12
202	-12
217	-16
227	-18
235	-16
251	-8
263	-7
276	-11
306	-14
354	-13
363	-13
383	-13
410	-13
444	-13
478	-13
546	-10
604	-9
630	-10
640	-11
681	-23
699	-10
716	-5
725	-3
745	-1
755	0
771	0

F=:	dB
782	0
787	0
819	-3
845	-7
873	-18
897	-20
920	-18
998	-11
1311	-16
1591	-19
1796	-20
2497	-22
3111	-24
3727	-26
5993	-30
17490	-33
15419	-40
1206	-14
1165	-12
1070	-11
989	-11



Phase shift cw filter
Output /dB.
0
-5
-10
-15
-20
-25
-30
-35
100
200
300
400
500
700
1000
1500
2000
3000
4000
5000
6000
Frequency /Hz.

LIFTING CW OUT OF THE NOISE

DESIGN FOR A CW FILTER WHICH AVOIDS THE NEED FOR HIGH-Q RESONANT CIRCUITS WHICH CAN IMPAIR INTELLIGIBILITY THROUGH RINGING. IT'S A PHASE-SHIFT CIRCUIT BASED ON 074 OPERATIONAL AMPLIFIERS

BY DAVID DAVIES, C.ENG. IERE, G4YKT

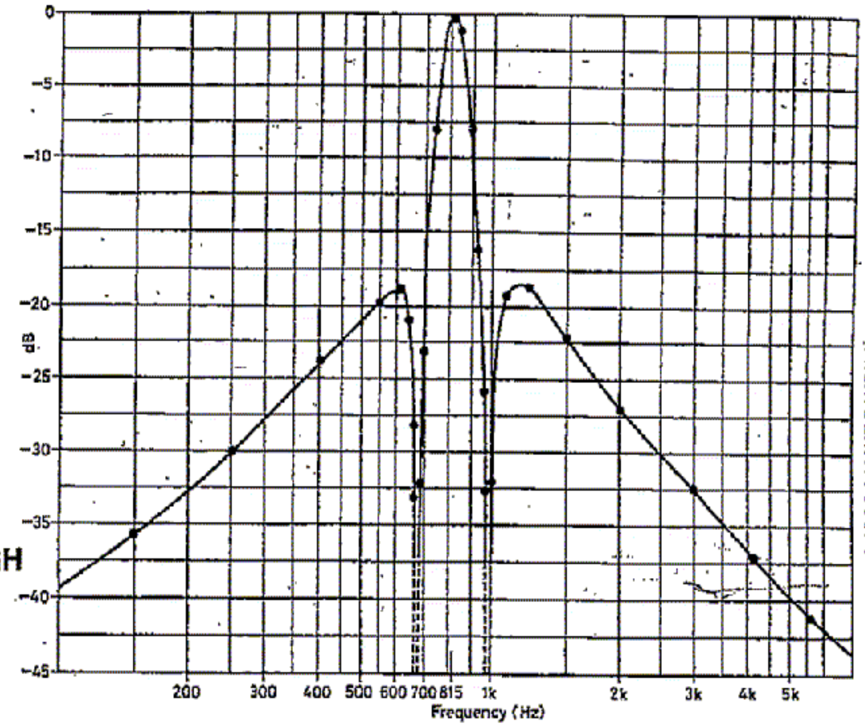
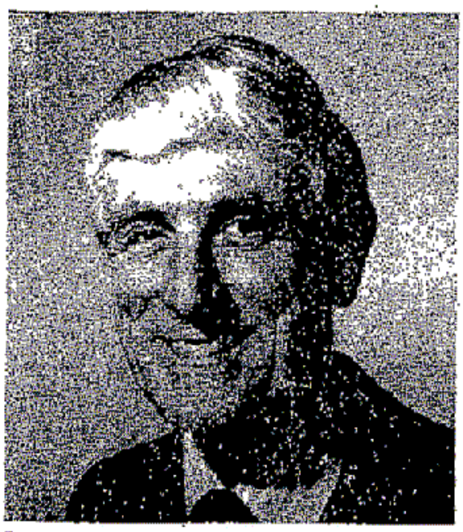


FIG 2. HOW THE CHOSEN 120HZ BANDWIDTH LOOKS IN PRACTICE

A majority of cw enthusiasts will, at one time or another, have considered audio filters as an aid to extracting weak signals from a noisy background or, as a means towards eliminating a strong interfering signal. Over the years literally dozens of designs have appeared in the amateur press offering versions of bandpass filters, notch filters, 'clippers' and 'limiters', not to mention spatial effect filters. Some of these devices can make a worthwhile contribution to reception but, in my experience, none has quite lived up to expectations. For example, narrow passband filters have a tendency to 'ring' giving a quite unnatural quality to the received signal. On the other hand, some designs can sound 'woolly' as if the signal were emanating from the bottom of a well. Then there are versions that require adjustments during reception which is, at best, inconvenient and, at worst, can mean that by the time the adjustment is optimised the contact is lost or conditions have changed.

After building a number of conventional filters of one kind or another, sometimes with disappointing results, I pursued the idea of a filter which could extract the wanted signal from a noisy background without changing its essential quality or destroying the intangible but nevertheless real, sense of contact that cw operators have with the signal environment. The objective was to avoid highly resonant elements and yet achieve a narrow steep-sided passband.

BIOGRAPHICAL NOTE



David Davies was born in Flint in 1927. While attending Holywell Grammar School he became interested in 'wireless' and built a number of one and two valve receivers, most of the components being home made. His early career included seven years as a Marconi Marine radio officer and several years as a tv service engineer. For the past 30 he has been employed by the British Steel Corporation as a Principal Research Officer largely involved in the design and development of electronic systems for the automatic inspection of steel products using ultrasonic, eddy-current and other techniques. Having always had an interest in amateur radio he finally found time to become licensed in 1984. He admits to spending more time modifying his equipment than he does on the air.

PHASE SHIFT FILTER

A block diagram of the design finally adopted is shown in Fig 1. Each of the elements IC1 to IC8 is a low Q (Q is approximately 2) unity gain passband filter centred at 815Hz. At the centre frequency each element produces a phase shift of 180°. At all other frequencies the phase shift will be greater or less than 180°. The design discriminates against frequencies which are not phase shifted precisely 180° through each element – ie everything except 815Hz.

Consider, first, an incoming signal of 815Hz. At the output of IC1 the signal will be phase shifted 180° and, after passing through IC2 a further 180° change, returned to its original phase. At the output of IC3 the signal is again 180° out of phase with the original and is identical to that at the output of IC1. Phase reversal continues on down the chain, points marked 1 3 5 and 7 (Fig 1) will show identical replicas of the incoming 815Hz signal but, 180° out of phase. At the same time, points marked 2 4 6 and 8 will show replicas in-phase with the incoming signal. At IC9 the 180° out of phase component from IC1 and the in-phase component from IC8 are additively combined via the inverting and non-inverting inputs of IC9 producing an output equal to twice the amplitude of the inputs.

A signal at 815Hz, is, therefore, passed through the filter virtually unmodified. At frequencies other than 815Hz the situation is, however, very different. For example, a signal at 700Hz will have a phase shift of some 170° through each element and the output of each element will differ from its predecessor, as indicated in Fig 1. At IC9 the inputs are again additively combined but, because of the phase difference, partial cancellation takes place. The degree of cancellation is dependant upon the relative phase difference at the inputs to IC9, when two signals are in-phase, but remembering that one will be inverted at IC9, total cancellation will take place.

In practice, total cancellation will take place when the two signals are not only in-phase but also have equal amplitudes. To accommodate minor variations in gain due to component tolerances RV2 is provided as a 'balancing' control. The capacitor, C10, preserves the DC conditions to the inputs of IC9.

The width of the passband is dependant upon the number of elements in the chain. The more elements the greater the phase change, at the end of the chain, for a given change in frequency away from 815Hz. In principle, the passband could be reduced to a few Hertz, but a reasonable compromise of some 120Hz was adopted. Fig 2 shows the overall response of the filter using eight elements; the passband is extremely steep and narrow owing to the rapidly changing phase between the outputs of IC1 and IC8 with any deviation from 815Hz. Also, total cancellation when the two contributions pass through their in-phase condition can be seen. This latter feature is helpful in eliminating an interfering signal close to the wanted station.

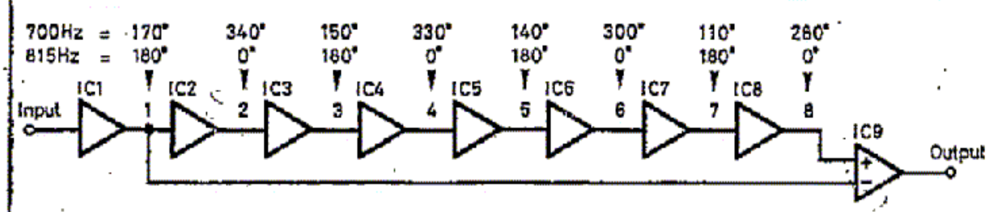


FIG 1. PHASE CHANGE AT EACH ELEMENT RELATIVE TO THE INPUT SIGNAL AT 815HZ AND 700HZ

R1= 75K C= 4n7
R2= 15K
R3= 150K

f = 782 Q = 2215 A = 1.0

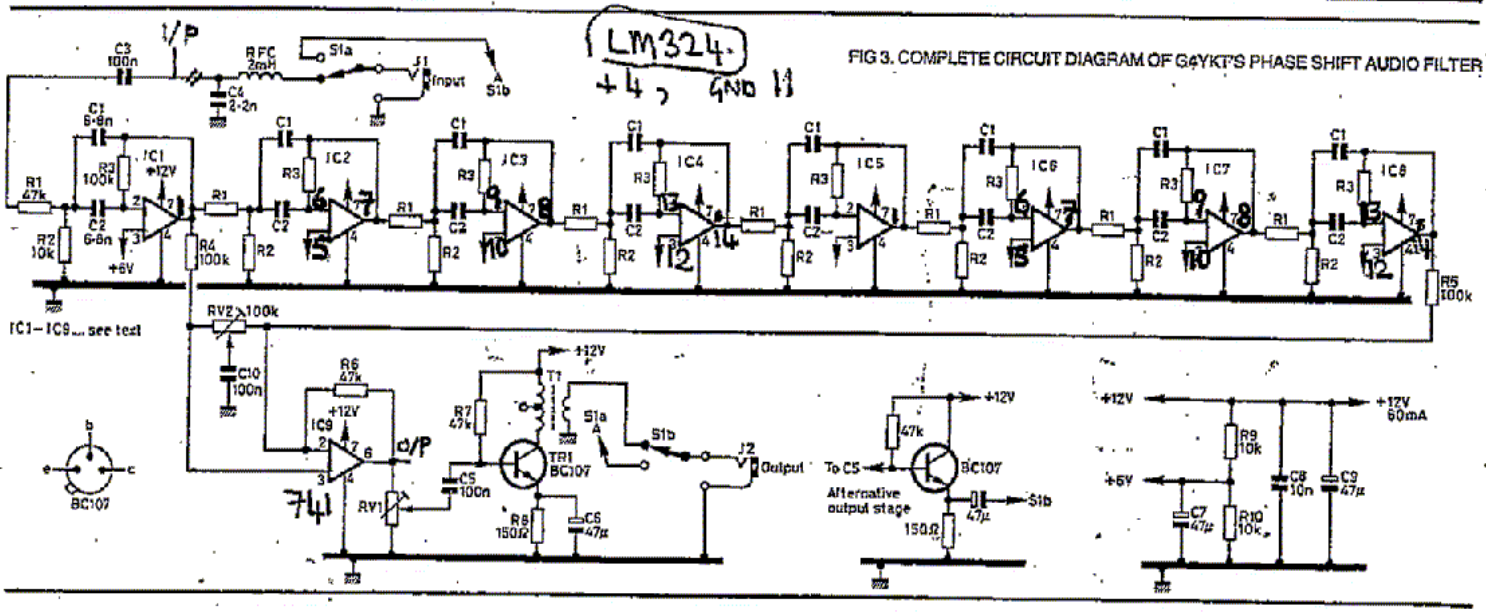


FIG 3. COMPLETE CIRCUIT DIAGRAM OF G4YKT'S PHASE SHIFT AUDIO FILTER

A centre frequency of 815Hz was chosen for two reasons. First, it lies within the optimum audible range for cw reception. Second – and this was the deciding factor – the frequency-determining components have standard easily obtainable values. Some constructors may wish to experiment with an alternative centre frequency, in which case the important parameters for each filter element are:

$$f=\frac{1}{2\pi C}\sqrt{\frac{R_1+R_2}{R_1 R_2 R_3}}$$

$$Q=\pi R3\,C10^5$$

$$A(\text{gain})=\frac{R_3}{2R_1}$$

$C_1=C_2=C$

$R=M\Omega$

$C=\mu Fd$

$f=Hz$

The first step is to choose a reasonable value for C and R3 then proceed by substitution; a once tedious process but now delightfully easy using a calculator.

The output stage is just about the simplest that could be devised to drive low impedance (8Ω) headphones. The writer has a particular preference for low impedance, 'high fidelity', headphones which have the advantage of revealing background noise and other spurious effects which are better eliminated at source rather than disguised by lesser quality headphones. However, if medium to high impedance headphones are preferred then the alternative output stage shown in Fig 3 may be used.

CONSTRUCTION

The complete circuit diagram is shown in Fig 3. It is important that the frequency determining components (R1 R2 R3 and C1 C2) in each of the filter elements are close tolerance items. In the prototype the capacitors were five per cent polystyrene types and the resistors one per cent metal film. Apart from this consideration there are no sensitive component values and the layout is not critical. The original was built on a piece of strip board measuring approximately 160 by 100 millimetres using individual 741 operational amplifiers. During the development stage alternative operational amplifiers, types 071 and 081 were tried, largely to determine if there would be any difference in the audible noise. In the event, there was no difference. With regard to noise, with no input and using high fidelity headphones it is just possible to discern enough noise to decide that the filter is switched on.

COMPONENT LIST

R1 (8 off)	47kΩ 1%
R2 (8 off)	10kΩ 1%
R3 (8 off)	100kΩ 1%
R4 R5	100kΩ
R6 R7	47kΩ
R8	150R
R9 R10	10kΩ
C1 C2	6·8nF Polystyrene 5% (16 off) IC1 to IC8
C3 C5 C10	100nF
C4	2·2nF disc
C6 C7	47µF
C8	10nF
C9	470µF
Op. Amps	See text
TR1	BC107
RV1 RV2	100kΩ Trimpot (linear)
T1	Miniature o/p transformer 6.6:1
RFC	2mH choke
S1	DPDT toggle switch

Misc: Metal Box, Strip Board, Jack Plugs & Sockets.

The final version was built using type 074 operational amplifiers, this is the 'quad' version of the 071. However, the use of quads or singles is a matter for individual preference, as is the use of sockets or direct soldering. If the constructor is not very experienced then individual, socketed, IC's is probably the best approach. The cost difference is slight and, certainly, layout and subsequent checking would be very much easier.

If the filter is to be used in conjunction with a transmitter when strong rf fields can be expected, these, if allowed to enter the filter chain, can produce extremely unpleasant audible effects, especially if the operator is using headphones! The rf choke (RFC) and sundry by-pass capacitors help towards avoiding this situation but if strong rf fields are going to be present, then it is essential that the filter be housed in a shielded enclosure.

The miniature output transformer in the output stage was salvaged from a defunct transistor radio purchased for 10p at a car boot sale. Similar transformers can be purchased from a number of stockists, being referred to as 'output' transformers.

CHECKING

A performance check on the completed unit is best carried out using an audio oscillator and an oscilloscope. With an input of 815Hz the output of each stage is checked for 180° phase change, and the filtering action observed at other frequencies. If an oscillator and oscilloscope are not available it is suggested that the filter is built using sockets and individual operational amplifiers. Checking is then performed using the output stage as a signal tracer. For example, provide an input from the station receiver, insert IC1, and probe pin six with the input of the output stage, a slight degree of filtering should be heard. If all seems well insert IC2 and repeat the procedure, following with IC3; and so on down the chain. The gain of each stage is essentially unity, so the signal amplitude will remain the same at each stage and only the filtering will become more pronounced as each IC is inserted.

Initially, the balancing control, RV2, should be set to mid-travel. When checking is complete and the pass-band is observed to be similar to Fig 2, RV2 may be adjusted to improve the depth of the nulls. This will require a compromise, as a given setting of RV2 will not be possible to achieve total cancellation at both nulls. In my experience adjusting the balance control to give cancellation at the lower frequency null offered the best operating advantage.

In use, the preset, RV1, is adjusted so that, while listening to a carefully tuned signal, the amplitude is the same for either position of the switch S1.

CONCLUSION

This aid to cw reception offers an audio filter having a narrow steep-sided passband without noticeable ringing or other objectionable effects. The filter is capable of retrieving a signal from a noisy background when, without the filter, the signal may be virtually inaudible. □