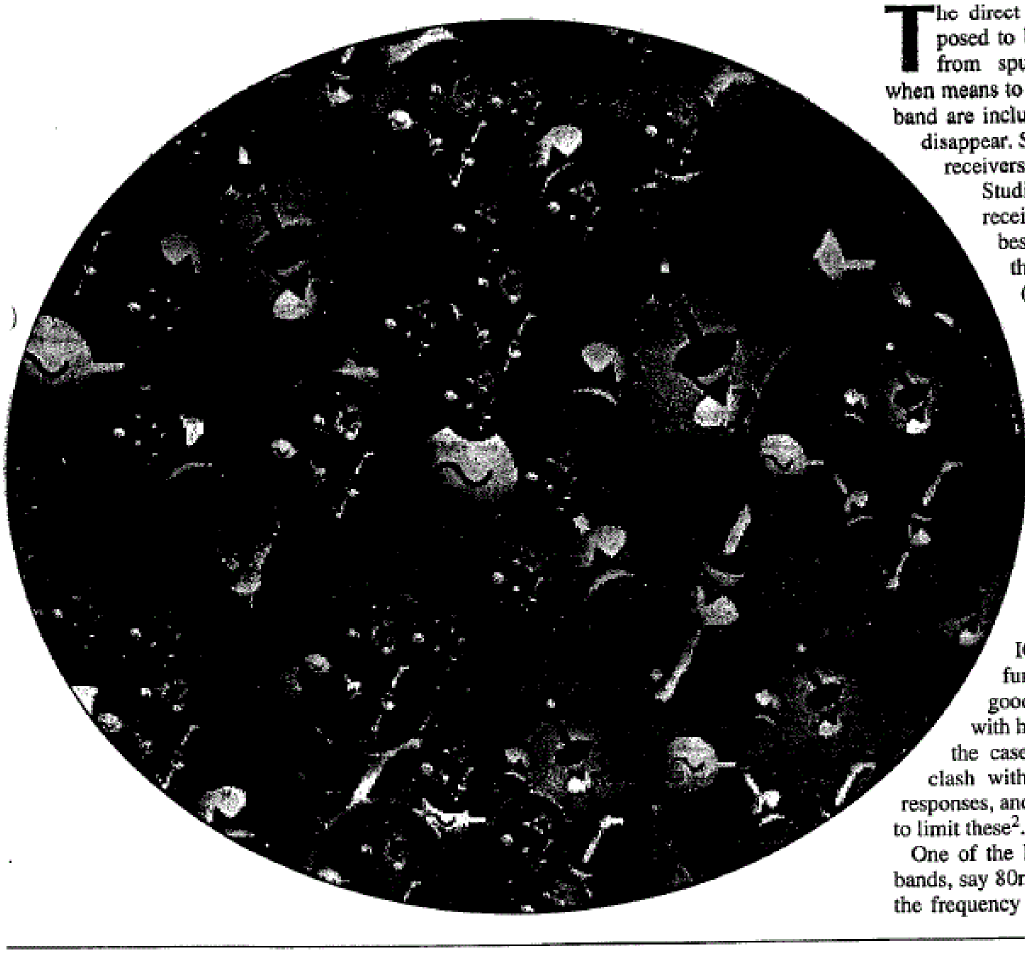
Direct conversion ssb receiver

Direct conversion for rf reception is now widely used in integrated form because it delivers good performance without the use of mechanical filters and other expensive, bulky items. Frank Dorey has put together a development breadboard for direct conversion SSB use, the principles of which can be adopted for many other applications.



The direct conversion receiver is supposed to be simple and relatively free from spurious responses. However, when means to eliminate the unwanted sideband are included, the simplicity seems to disappear. So practical designs for dc ssb receivers are still rare.

Studies on direct conversion receivers have concluded that the best option for ssb reception is the "Weaver type" receiver (known to radio amateurs as the *third method*). But it has to be ac coupled to get over the problem of a constant tone out of the second oscillator when dc drift occurs in the first balanced modulator. While there is a "hole" in the audio response, this, if not too wide, does not seriously detract from speech intelligibility[1].

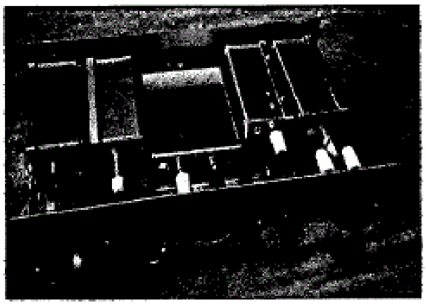
Although the use of digital ICs to realise some of the circuit functions required may seem a good idea, the use of square waves with high harmonic content tends, in the case of the Weaver receiver, to clash with the aim of no out-of-band responses, and extra work might be required to limit these[2].

One of the lower frequency HF Amateur bands, say 80m, seems to be a good place in the frequency spectrum to try out a design, using only analogue techniques so as to ensure freedom from spurious responses. As this is a design study, most of the circuitry is divided between five separate modules: rf amplifier and dual first mixers; 2-phase rf oscillator; dual 7-pole low-pass filters; dual second mixers, audio amplifier and agc circuit; two-phase second oscillator and power supply regulator.

Circuit operation

Referring to the block diagram, suppose that a transmission on a nominal carrier frequency of 3600kHz is in fact lower sideband with an audio range which would just fit this particular receiver. Then it would contain frequencies from 3599.8kHz down to 3596.4kHz. The first oscillator in the receiver should be tuned to 3598.1kHz to receive this transmission.

Sidebands of the local oscillation are produced extending for 1.7kHz on each side of it, a range of frequencies 3.4kHz wide. But what comes out of the balanced modulator consists of two sets of audio frequencies as superimposed signals, each in the range zero to 1.7kHz being the two halves of the original audio spectrum fed into the transmitter modulator. The ac coupling at the output of the first balanced modulator deliberately curtails the zero end of each band of frequencies, removing a dc component but putting a gap in the middle of any subsequent reconstitution of the original modulation frequency range. Also, one of the signals has a reversed frequency spectrum compared to the original modulation.

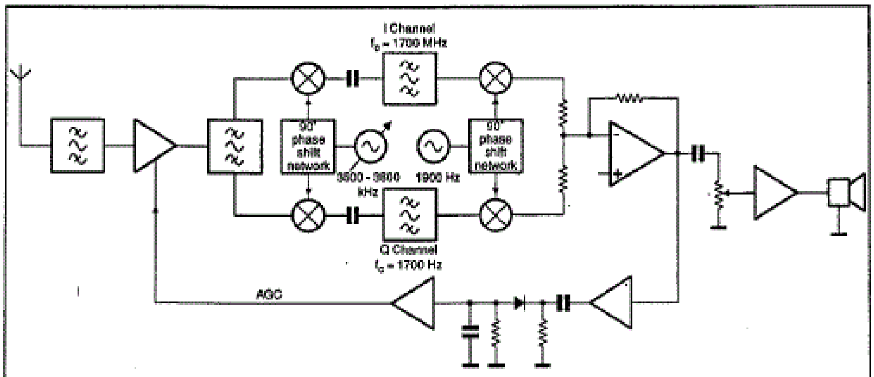


The prototype unit was built up as a series of modules wired together. This achieves a high level of screening, important even to basic direct conversion equipment.

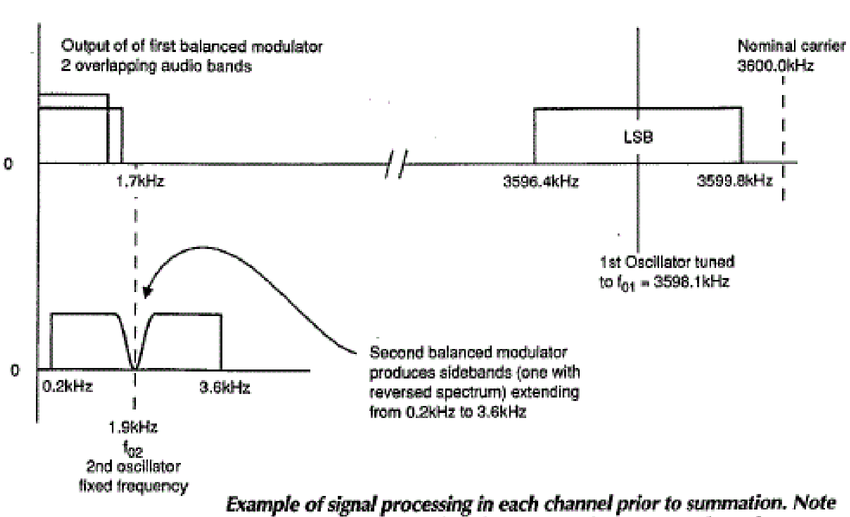
What has just been described is happening, of course, in two channels, the I (in-phase) and the Q (quadrature) channels. This permits signal processing by using another pair of balanced modulators fed from a 2-phase second oscillator. This is followed by a summing op-amp to re-invert one half of the audio spectrum and put together again the full range of modulation frequencies (except for a small gap in the middle).

The sideband range of the second oscillator, running at 1.9kHz, contains frequencies from 0.2kHz (ie. 1.9–1.7) to 3.6kHz (ie. 1.9+1.7). Correct phasing of the second oscillator inputs to the second balanced modulator channels will ensure that the output summation will reinforce wanted signal components and cancel the unwanted ones.

Note that incorrect phase choice will result in USB instead of LSB reception; also that success of the process depends on matched gains in the two channels up to the summing op-amp. For the maths to bear this all out, see the appendix to ref.1).



Receiver block diagram. Most of the circuitry is divided between five separate modules: rf amplifier and dual first mixers; 2-phase rf oscillator; dual 7-pole low-pass filters; dual second mixers, audio amplifier and agc circuit; two-phase second oscillator.



Example of signal processing in each channel prior to summation. Note the 'hole' caused by the fact that frequencies near zero from the overlapping audio bands, cannot pass through the ac coupling inherent in the circuit.

Prototype details

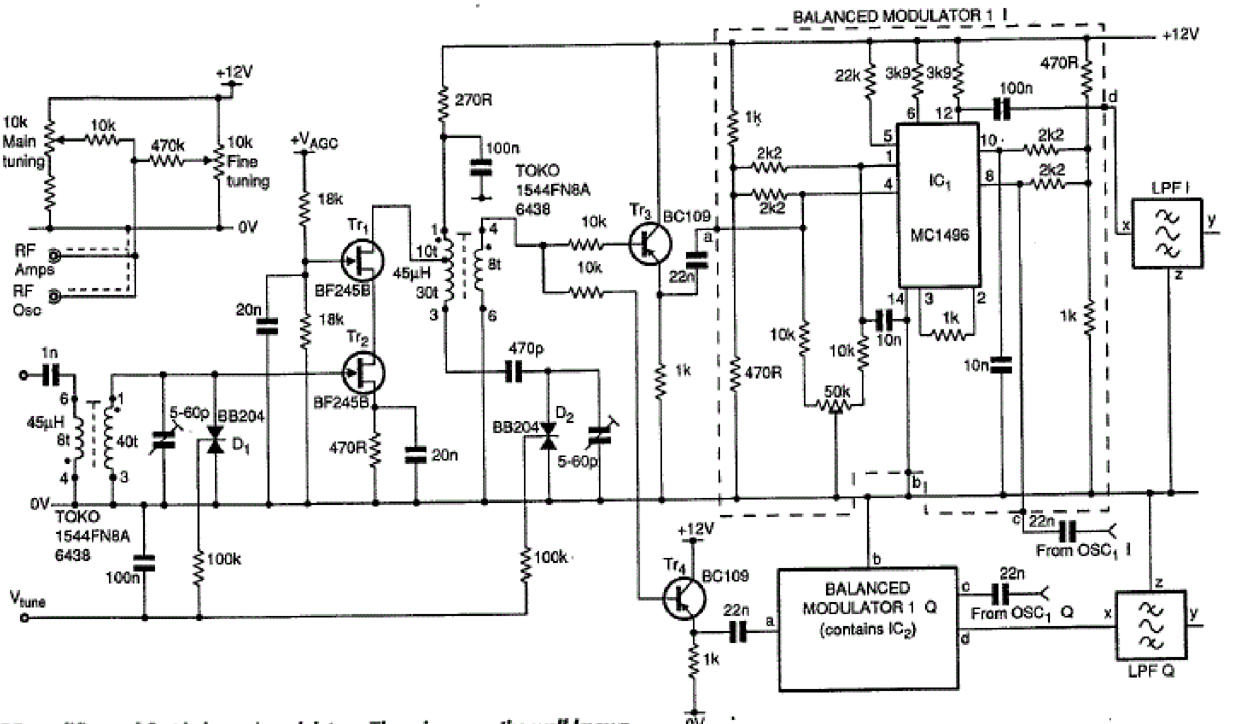
The signal interconnections between modules were made with phono connectors, colour-coded to identify channels. Each module was built on a printed circuit board contained in a small aluminium box. The five modules were housed in a case with panel mounted controls. No display of frequency is included. Since the channel to which the receiver is tuned is always (fo1+1.9)kHz, it is convenient to connect a monitoring frequency meter to an auxiliary output from the first oscillator.

No attempt was made to include a power supply inside the case, the regulator input coming from an external unstabilised supply of at least 14.5 volts dc placed some way from the receiver. The current taken from this supply was found to be 190mA.

The cascoded junction fet rf stage uses Toko coils for 3.5 to 3.8MHz coverage, and incorporates diode tuning. Source follower buffers couple it to the following balanced mixers. The mixers use the well-known *MC1496*. Substitution for the later *NE602* should reduce the component count and simplify the layout.

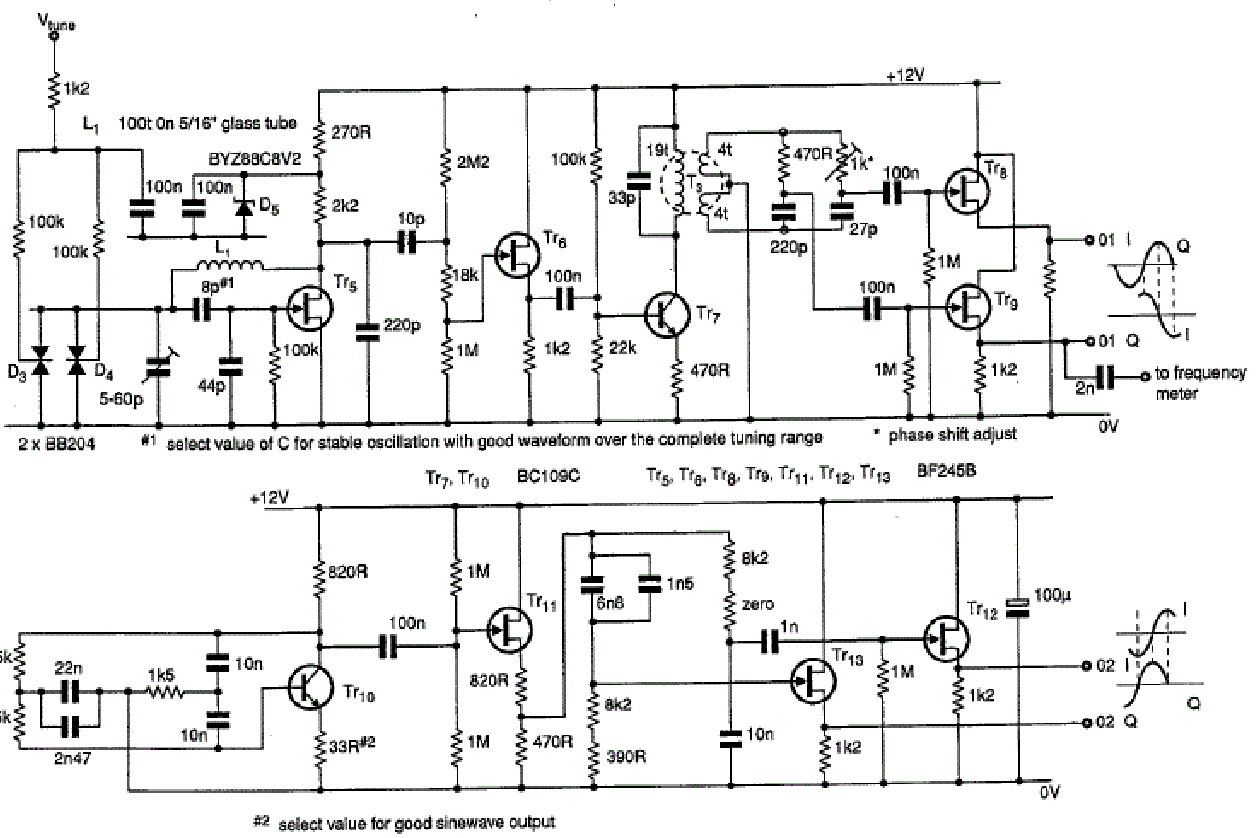
The fet Vackar first oscillator circuit is built around a coil wound on a 5/16 inch glass former with 100 turns of 0.15mm enamelled wire. The gate drive is adjusted so that oscillation is just stable over the complete frequency band, with a good clean waveform.

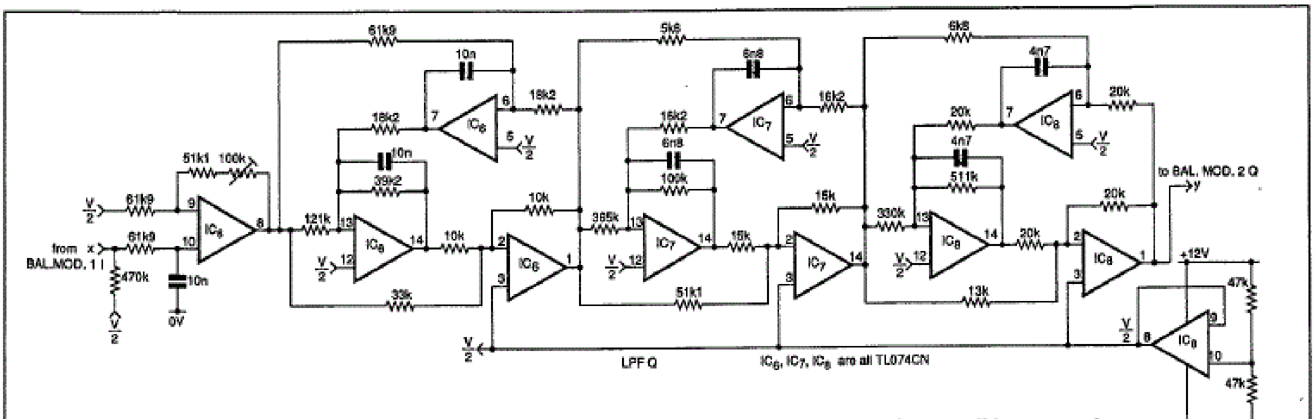
The rf phase-shift network was suggested by Ref 3. Operation was checked using an oscilloscope, having first ensured that no phase difference was indicated when the same signal was applied to both Y-channels. This frequency is too high for the bandwidth of the average X-channel so that a Lissajous figure method cannot be used unless the scope X-input can match the phase response of the Y-channel.



RF amplifier and first balanced modulators. The mixers use the well-known MC1496. Substitution for the later NE602 should reduce the component count and simplify the layout. High level parts such as the SL6440 type will improve strong signal performance but require care in matching.

First RF and second AF oscillators.





Low pass filter circuitry for I or Q channel. Values were calculated for a seventh-order elliptic filter with a cut-off frequency of 1700Hz, pass band ripple width 0.3dB, and minimum stop band loss 80dB. This last value puts it in a category where little or no trimming should be required if capacitors with a 5% tolerance are used and resistors are within 1% of the calculated value. Other channel is identical.

The low pass filter design is detailed in Ref.4. Values were calculated for a seventh-order elliptic filter with a cut-off frequency of 1700Hz, pass band ripple width 0.3dB, and minimum stop band loss 80dB. This last value puts it in a category where little or no trimming should be required if capacitors with a 5% tolerance are used and resistors are within 1% of the calculated value.

Using K=3 in each stage, including the first-order stage, gives each filter a pass band voltage gain of about 80. Adjustable gain in each first-order stage permits balancing the overall gains of the I and Q channels.

The second oscillator uses a twin-T circuit followed by a simple phasing network. The circuits were laid out so as to allow the late addition of series or parallel trimming additions to get the frequency exact and the phase difference exactly 90°, with equal amplitudes of the I and Q outputs. With a frequency in the audio range it was possible to switch the oscilloscope to its X/Y mode and trim for a near-perfectly circular Lissajous figure.

The audio amplifier is a simple unity-gain output stage following the summing amplifier and the agc circuit is fed from a gain-of-5 stage, using spare amplifiers in the *LM324* which already contained the summing amplifier at the end of the I and Q channels. The volume control and the audio output stage are fed direct from the summer.

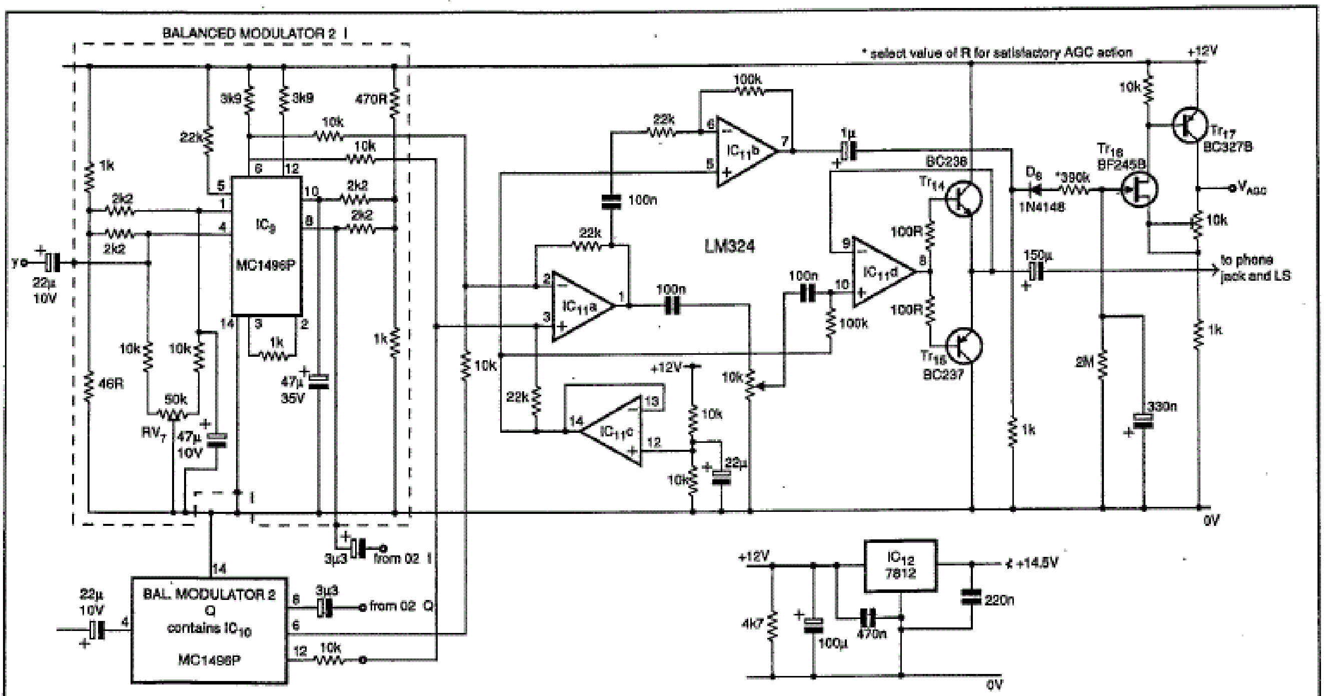
The agc circuit was suggested in Ref.5, and as stated there, it is necessary to choose carefully, how much control to use. Excessive values lead to a particularly annoying "pinging" sound as each word is spoken in this type of receiver.

Levels of carrier injection are 300mV rms for each of the rf carrier components and 50mV rms for each AF carrier component.

Alignment

Check or adjust the drive levels of both the rf and af oscillators to ensure good sinewaves. Tune the first oscillator to the correct range of 3500 to 3800kHz by adjusting (alternately) the capacitive trimmer and the value of resistor in series with the main tuning control.

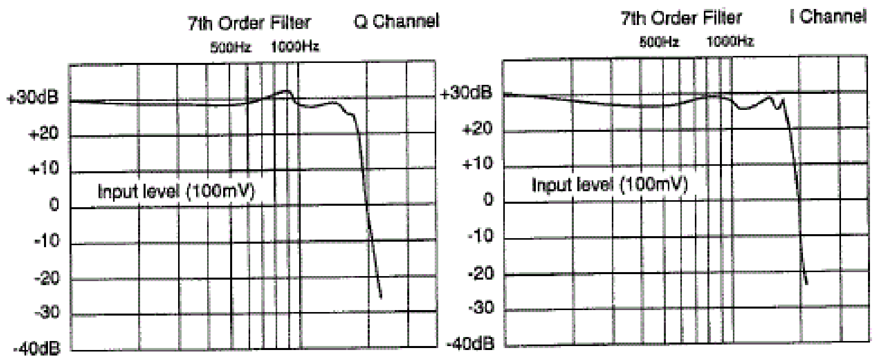
Set a phase difference of 90° between the rf carrier I and Q outputs by adjustment of the variable resistor in the phase shift network. The amplitudes should be reasonably equal. Trim the second oscillator frequency to 1900Hz by adjustment of the value of the auxiliary capacitor in the centre leg of the T-network with a resistive top.



2nd balanced modulators for I and Q channels and audio amplifier/power supply. IC11c provides a half-supply rail bias point for other op-amps in the system. The pre-amp sections would benefit from a lower noise device in place of the LM324 used in the prototype.

Trim the phase of the af carrier outputs, by adjustment of the values in the four arms of the phase shift network for a good Lissajous circle. Balance the first and second balanced modulators, observing their waveforms at the respective output pins. Apply an input CW signal, at a level of say 50µV, and observe that LSB reception is being achieved, ie. that the output tone increases in frequency as the signal generator frequency is decreased. If the opposite occurs the second oscillator carrier components should be swapped over at the balanced modulator inputs in the I and Q channels. Tune the rf amplifier circuits in the usual way, ie. coil cores at the low frequency end of the band and capacitive trimmers at the high end.

Observe levels due to a received cw signal at the outputs of the I and Q channel low-pass filters and adjust the gains of the first order stages until the channel amplitudes are equal. Then tune through the sideband range slowly. Probably, as the correct tone increases, a spurious output tone will be heard to correspondingly decrease. A careful attempt should be made to adjust the gain of one channel to maximise the loudness of the correct, ie. wanted, tone as compared to the spurious one. The only adjustment which remains is to choose the optimum value for the resistor in series with the agc diode to give satisfactory action on a strong received speech signal.



Low pass filter response curves for I and Q channels.

Performance

In use the receiver needs to be carefully earthed to minimise hum and feedback. Feeding inputs from a signal generator via a standard dummy aerial showed usable outputs from signals down to a few microvolts. On a short outdoor aerial the receiver compares well with a conventional superhet incorporating a mechanical filter, while it is pleasantly free from spurious whistles. ■

References

1. *Direct Conversion ssb receivers*, Dr.S.R.Al-Araji and Professor W. Gosling, *The Radio and Electronic Engineer*, Vol.43, No.3, March 1973, p.209.
2. *ICs simplify design of single-sideband receivers*, I.Hickman, *EW+WW*, November 1991, p.939.
3. *PA0KSB's 2-phase receiver*, a summary of the work of K.Spaargaren included by Pat Hawker,G3VA, in Technical Topics, *Radio Communication*, November 1970, p.761.
4. *Active Elliptic Audio Filter Design using Op-Amps*, D.H.G.Fritsch, G0CKZ, *Radio Communication*, February, p.98 and March, p.179 1986.
5. *ARRL Handbook 1988*, Chapter 28, *Audio and Video Equipment* page 28-11, Fig.14 and associated text.