output and for a 100 mV input an output of 400 mV. This can be shown to be a gain of 12 dB. Tests on the air revealed the FT-200 gain had to be set lower than normal due to this 12 dB, which puts in some extra 200 mV as opposed to the station microphone. Reports indicate speech quality to be slightly toppy, against normal reports using the station mike.

The circuit and components list is shown herewith. By using the pre-amplifier with the normal station microphone it may be held a little further away from the mouth for the same gain, which allows a little more freedom about shack.

THIRD-METHOD SSB EXCITER

USING LOGIC CIRCUITS

G. C. CLEMERSON, B.Sc. (G4CLT)

THERE are three possible ways of generating an SSB signal. These are the phasing, filter, and third methods, and are shown in block diagram form in Fig. 1. The phasing methods, once quite popular, has largely been superseded by the filter system, which is easier to set up, and can be used for transmission and reception—an obvious advantage in transceiver design. The "third method," which is sometimes described as a combination of the other two, has never attracted much interest among amateurs.

In the third method, two oscillators are used. One operates a radio frequency f_c. The other is an AF oscillator with a frequency f_p, which is at the centre of the audio passband. Each oscillator feeds a pair of balanced modulators *via* a 90° phase shift network. The two pairs of balanced modulators are connected together by two identical low-pass filters, which have a cut-off just below f_p. Because this is a low audio frequency, very high performance filters can be easily constructed using simple LC circuits.

In essence, two single sideband signals are produced. One is a lower sideband with a nominal carrier at $f_c + f_p$, and the other an upper sideband with a nominal carrier at $f_c - f_p$. Both sidebands are produced in each "arm" of the circuit, but with opposite phases. When it is correctly adjusted, either the lower or upper sidebands cancel in the output tank circuit, depending on the relative connection of the two 90° phase shift networks.

A simple way to produce the required AF and RF phase shifts is to use digital integrated circuits. Fig. 2(a) shows one suggested method. The main disadvantage of this circuit is that very high speed devices are needed to give an output above one or two megahertz. By using "double ended" drive to the clock inputs, the frequency division can be reduced to two, and higher frequency output obtained with cheaper circuits.

The generator described in this article uses two dual type-D flip flops, SN7474, and is capable of producing SSB output at above 6 MHz. The connection of the integrated circuits is shown in Fig. 2(b). To change from upper to lower sideband, it is only necessary to reverse the phase of one balanced modulator. This is most conveniently done by reversing the connections to the Q and Q outputs of one flip flop, preferably the one operating at audio frequency.

Construction

The unit is constructed on two 6½in. x 4½in. printed circuit boards. All the active components are mounted on one board, and the other one contains the two LC filter networks. The two boards are mounted one above the other in a small aluminium box. Power supplies are

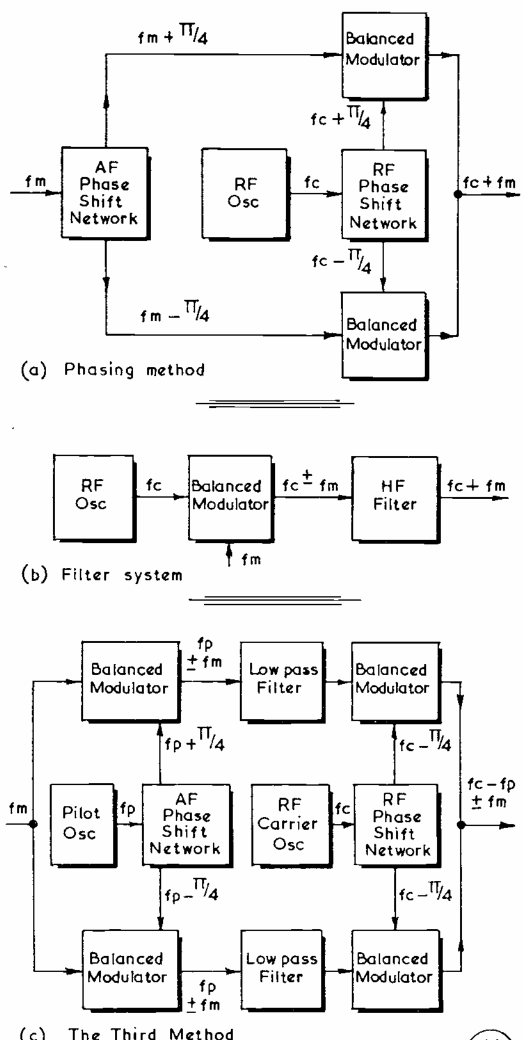
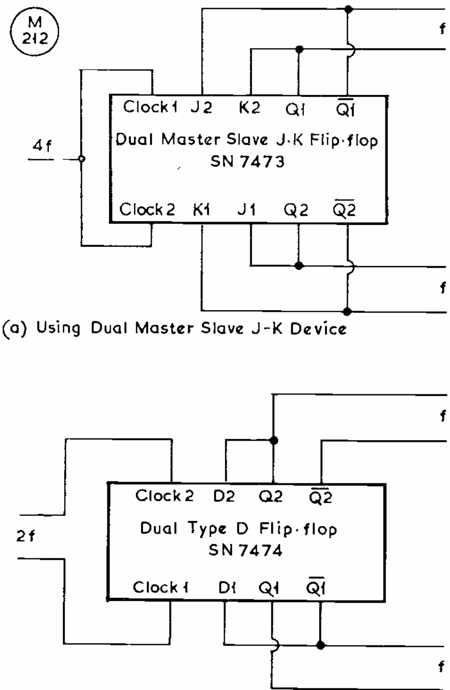


Fig. 1 THREE METHODS OF PRODUCING SSB



(a) Using Dual Master Slave J-K Device

(b) Using Dual Type D Device

Fig. 2 PHASE SHIFTING WITH INTEGRATED CIRCUITS

not included, and a stable source of 5v. at about 70 mA is required.

The selection of components for use in the prototype was made partly on the basis of what was readily available. Other constructors might wish to make substitutions, particularly for the transformers and filter inductors. Close-tolerance units are only required in the pilot tone oscillator and the filters, and a method of selecting suitable values for the latter is given below.

Circuit Description

The system will function as either a generator or detector of SSB signals. For simplicity, only the generator action will be described. When used for reception, the input and output are reversed, as is the signal path, but otherwise the circuit action is similar. The preset balancing controls are adjusted for best SSB output, and should not need resetting for use as a detector.

Fig. 4 is the circuit diagram of the main unit. Audio input reaches both "arms" of the circuit *via* RV1, the Audio Balance control. In the upper "arm" it is taken *via* T1 to the double balanced mixer D1-D6. Because both the audio input and the audio carrier (*i.e.*, pilot tone) are in the same frequency band as the DSB output, a single balanced mixer is not possible here. Diodes D1-D4 provide balanced switching, and diodes D5 and D6 act as clamps, eliminating the need for balancing presets. The arrangement gives a higher output than a ring mixer, and needs no adjustment.

The reference level for both the double balanced mixers is set by the emitter follower Tr1. If a higher degree of pilot carrier suppression is required, the potential divider R5, R6 can be replaced by a small preset potentiometer.

The audio pilot tone is generated by the multivibrator Tr2 and Tr3, which operates at 3·3 kHz. Since the accuracy of the audio phase shifting depends on the mark-to-space ratio of this waveform, close tolerance components are used for R8, R9, C2, and C3. Outputs are taken from both collectors to IC1, which produces quadrature outputs at 1650 Hz to feed both balanced modulators.

The output from the centre tap of T1 consists of audio frequencies from zero to 3·3 kHz. These are normal and inverted double sidebands of the pilot tone at 1650 Hz. It is taken through one of the low-pass filters, which removes the upper sidebands and any residual carrier, and reappears on the main board at C4. Similarly, the output from T2 passes through the other low-pass filter, and is applied to C5.

The audio signals at C4 and C5 are taken to the modified ring mixer, D13—D16, *via* T3 and T4. This circuit is identical to the type used in many phasing exciters. Quadrature RF drive is obtained from IC2, *via* T3 and T4, at 6 MHz. RV2 and RV3 provide RF Balance controls for both arms of the circuit. SSB output is taken from the common tank circuit *via* a link winding.

Tr4 operates as a xtal oscillator at 12 MHz, and drives Tr5 and Tr6 via the centre tapped transformer T5. It would be possible to provide fine adjustment to the RF phasing at this point, but if the centre tap of T5 is not decoupled to RF, it is not necessary. The transistors Tr5 and Tr6 act as switches, and provide suitable outputs to drive IC2.

Filter Details

The circuit of the filter board is shown in Fig. 5. It was designed to have a cut off at 1350 Hz, which gives an audio response from 300 to 3000 Hz. The actual response is not as critical as the matching of the two filters. Any imbalance here will cause different audio phase shifts, and spoil the unwanted sideband suppression.

For the prototype, all the inductors were wound on ferrite pot cores, and final adjustments made with a commercial inductance bridge. An alternative method, which gives equally satisfactory results, is to connect a 1 μF capacitor across the inductor, and then measure the resonant frequency with an AF signal generator and an AC voltmeter. (Smaller values of shunt capacitance may cause errors, because of the self-inductance of the windings).

A similar method can be used for the capacitors. Each capacitor is connected across a known inductance, and the resonant frequency checked. Matched values of capacitance should then be used in the corresponding locations in each filter.

Testing and Setting-Up

An oscilloscope with a bandwidth of at least 6 MHz is needed to set up the balance controls. The SSB output will contain harmonics of the 6 MHz carrier, and the

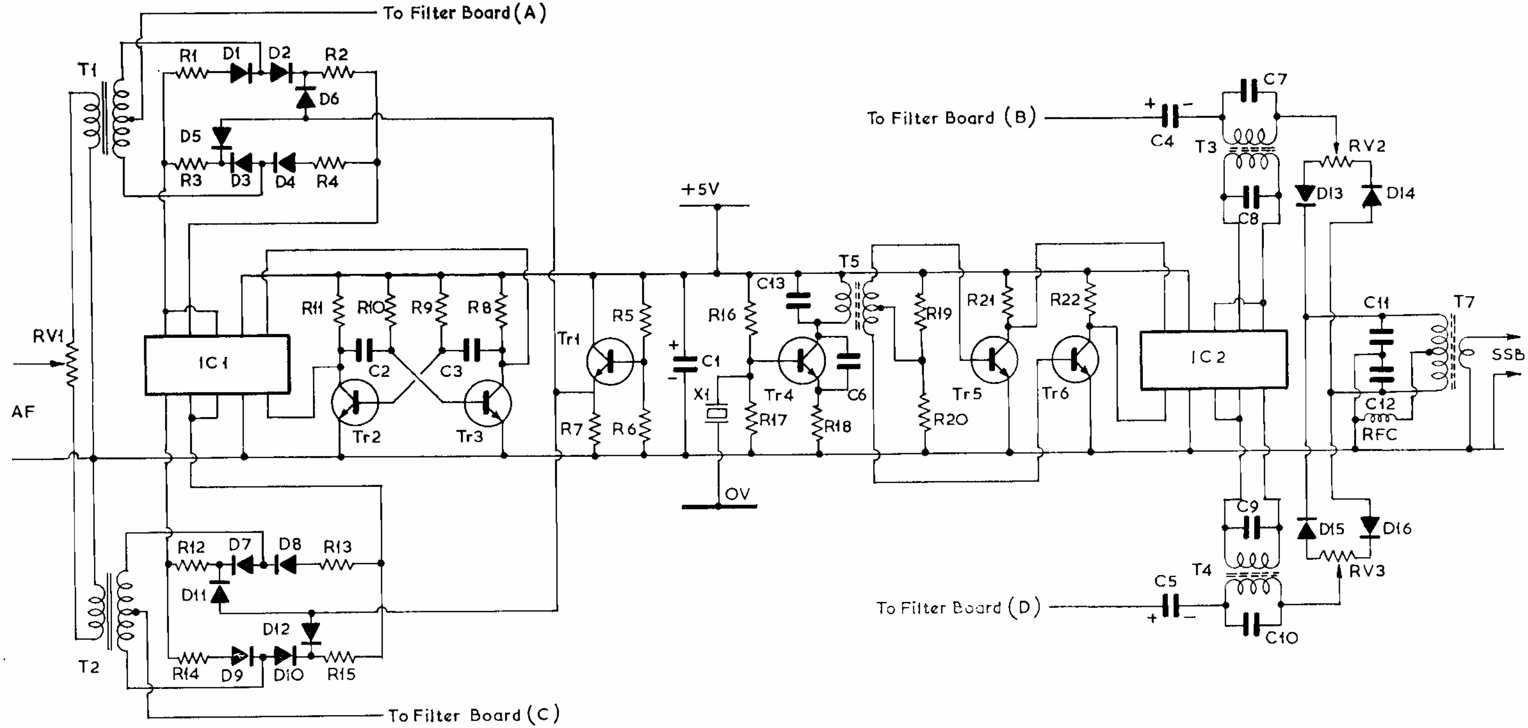
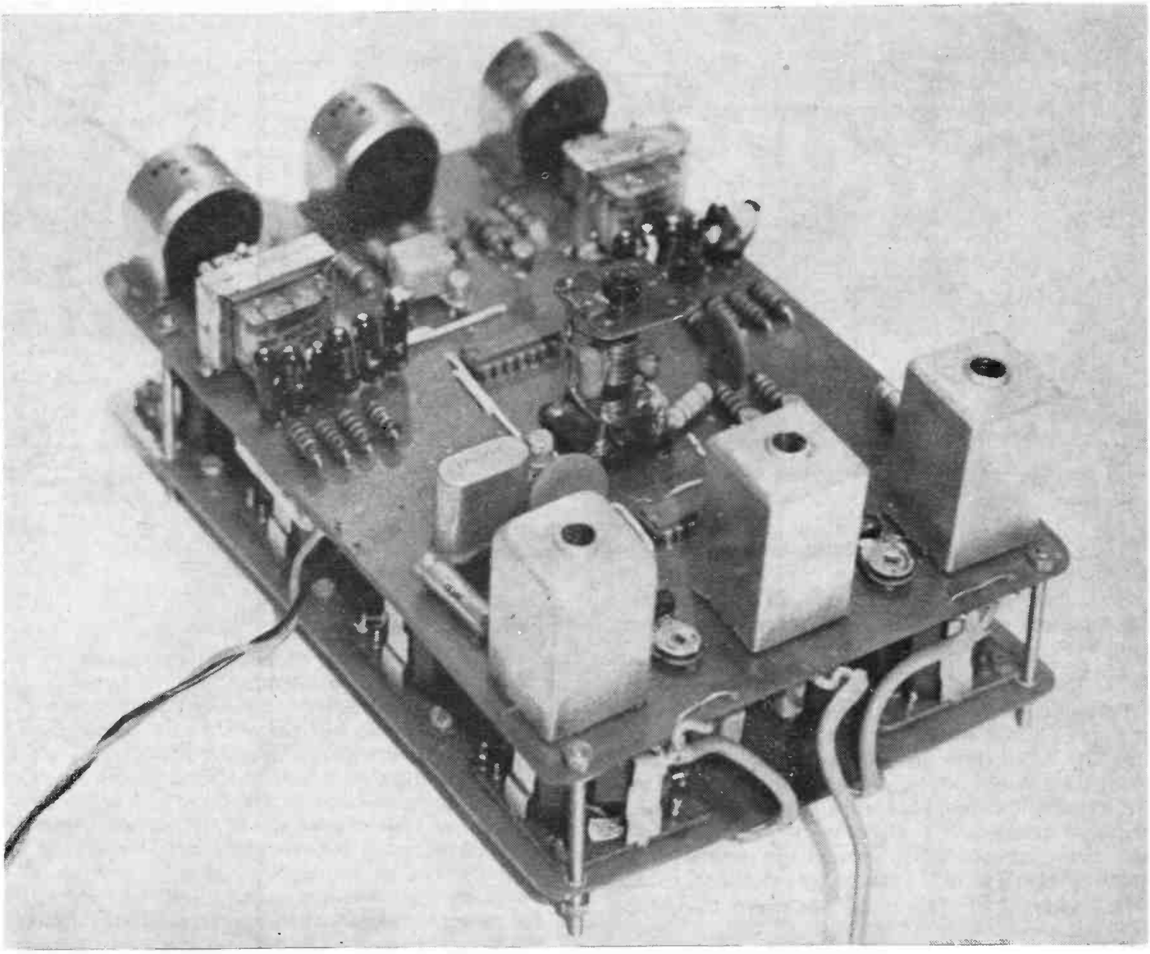


Fig. 3 CIRCUIT DIAGRAM OF MAIN BOARD

TABLE OF COIL DATA

T1, T2 = Any two identical small centre-tapped AF transformers, such as *Radiospares* TT7.
T3, T4 = Primary 30 turns, secondary 20 turns, spaced quarter-inch from primary.
T5 = Primary 10 turns, secondary 5 turns centre-tapped, spaced half-inch from primary.
T7 = Primary 16t. centre-tapped, secondary 3t. p.v.c. wire over middle of primary.
RFC = 5 mH RF choke of usual pattern.

Except for T1, T2 and RFC, all coils close-wound using 28g. enamelled on 0·3-in. formers with screening cans. In this circuit, there is no "T6" nor value for it.



Constructional form for the circuit Fig. 3, opposite

Table of Values

Fig. 3. The Main Board

C1	= 1·001 µF	R16	= 8,200 ohms
C2, C3	= ·022 µF, 1%	R17	= 2,200 ohms
C4, C5	= 10 µF	R19	= 22,000 ohms
C6	= 5 µµF	R20	= 470 ohms
C7, C10	= 82 µµF	RV1	= 2,000 ohms, 1in.
C8, C9	= 120 µµF	RV2, RV3	= 1,000 ohms, 1in.
C11, C12	= 470 µµF, 2%	D1-D6	= OA91 diodes
C13	= 100 µµF	IC1, IC2	= SN7474
C14, C18, C19, C23	= 0·25 µF*	Tr1, Tr2, Tr3, Tr5, Tr6	= BC-108
C15, C17, C20, C22	= 0·38 µF*	Tr4	= BSX-20
C16, C21	= 0·47 µF*	XL1	= 12·7 MHz, HC-U type
R1, R4, R13, R14	= 560 ohms	L1, L4, L5, L8	= 31·5 mH*
R2, R3, R12, R15	= 270 ohms	L2, L3, L6, L7	= 115 mH*
R5	= 3,300 ohms		
R6	= 2,200 ohms		
R7, R8, R11, R18, R21, R22	= 1,000 ohms		
R9, R10	= 10,000 ohms, 1%		

Notes: All resistors and capacitors 20% tolerance unless otherwise marked. Where marked * see text for method of selecting these components. C1 can be built up by 1 µF + ·001 µF.

oscilloscope should be connected to the unit by a small RF transformer tuned to 6 MHz. The controls are then adjusted to display the envelope of the RF output.

Initial adjustments are made without any audio input. Oscillator transformer T5 should be adjusted until output is obtained at 6 MHz, and then both T3 and T4 can be peaked for maximum output. The RF balanced modulators are now set by adjusting RV2 and RV3

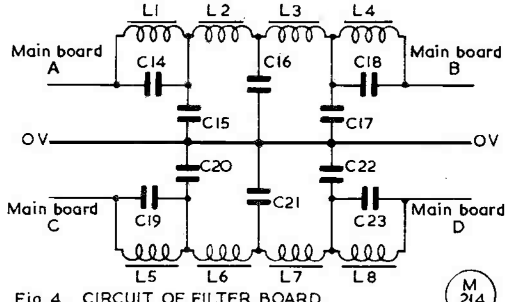


Fig. 4. CIRCUIT OF FILTER BOARD

M 214

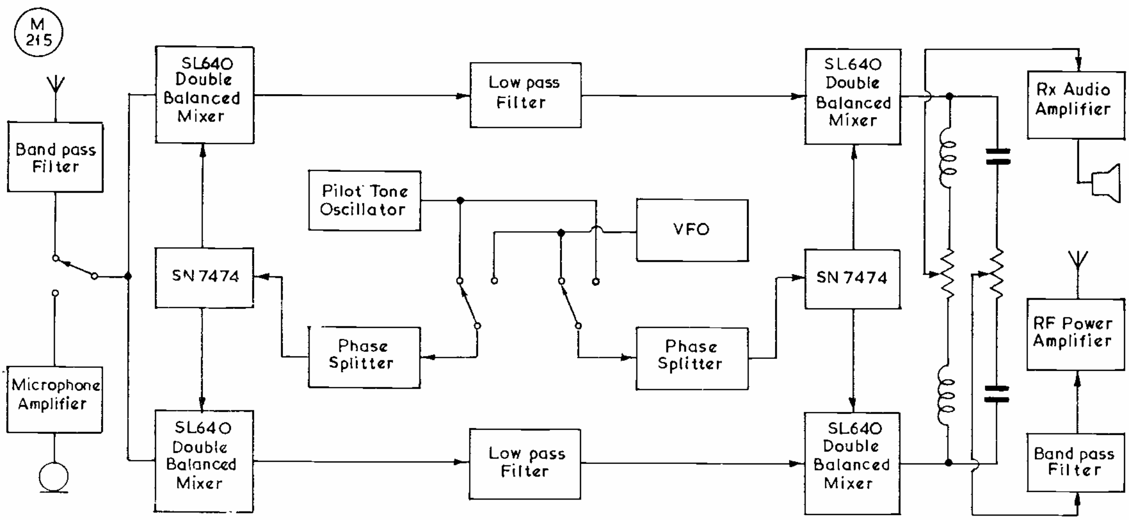


Fig. 5. PROPOSED THIRD METHOD DIRECT CONVERSION TRANSCEIVER

for *minimum* output. The two controls will interact somewhat, and several successive adjustments will be needed.

When the RF circuits have been set up, apply about 5 volts of audio at around 1000 Hz to the input, and carefully adjust RV1 for minimum ripple on the RF envelope. It should be possible to obtain less than 5% ripple with careful adjustment. If this is not achieved, try slight adjustments to the cores of T3 and T4. As a last resort, the coupling between the primaries and secondaries of T3 and T4 can be altered, but this should not prove necessary if the transformers have been wound carefully.

Performance

A poor Third-Method Exciter does not produce any output in adjacent channels, or at the carrier frequency. Imperfections show up as a steady output at the centre of the passband, and as inverted sideband in the same channel as the wanted sideband. The prototype had a residual output at some 45 dB down, and some inverted sideband at about 26 dB below the wanted sideband.

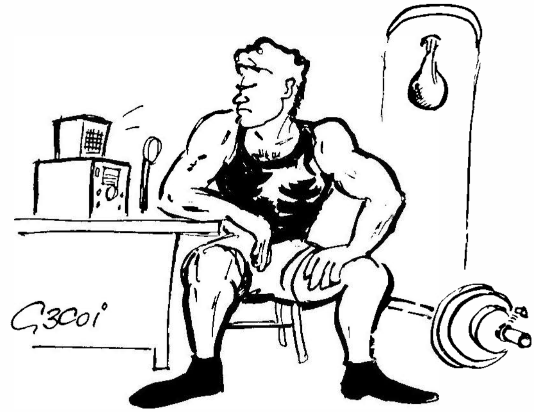
Any comparison with filter-type exciters is difficult, because the unwanted sideband does not have the same significance. In practice, it is almost completely swamped by the wanted sideband, and the only effect is to give the signal a characteristic sound which several operators have commented on.

When used for reception, the unit produces some output at the pilot frequency. This can be easily eliminated by the use of a simple band-stop filter in the audio amplifier. There will be a small "hole" in the AF passband, but this is not noticeable and is in fact already present due to the AC coupling between the low-pass filters and the RF circuits.

Further Development

One of the most interesting applications of the Third Method is in direct-conversion transceivers. The prototype was not suitable for this purpose because of the xtal oscillator, and the number of tuned circuits. Another version is currently under development which, it is hoped, will overcome these difficulties. A simplified block diagram is shown in Fig. 5.

The use of integrated circuit (IC) mixers eliminates some of the preset controls and reduces the size of the unit, but it does introduce some extra complexity in "transmit" to "receive" switching. DC coupling to the RF mixers removes the phasing errors which are caused by differences in the tuned transformers of the prototype. The system is ideally suited to use with a mixer VFO, since the SSB output has twice the stability of the tunable oscillator, and does not contain any unwanted products of the mixing process.



" . . . You sound pretty weak to me . . . "