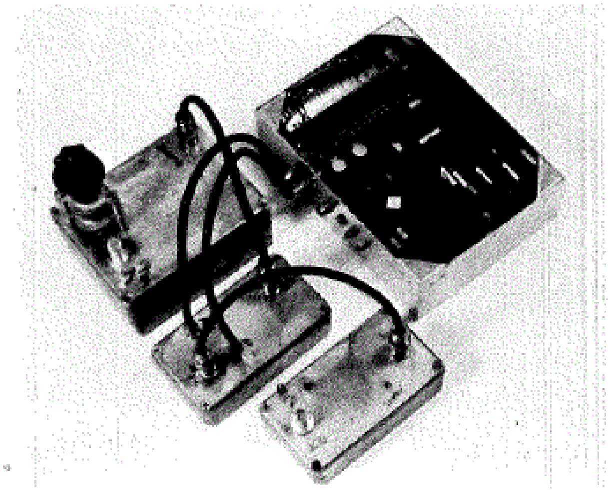


2-meter transmitter
uses Weaver modulation

Try the "third method" of SSB generation

Imagine a 2-meter SSB transmitter that contains no crystal filters, no IF amps, no heterodyne oscillators, no BFOs, and no broadband audio quadrature generator, either.

Impossible? *No.* The scheme described herein is based on a little-known technique usually called the "third method" of SSB generation, which I prefer to call the "Weaver Modulator," in honor of D.K. Weaver, its apparent inventor. First discussed in 1956, the technique has rarely been seen in the commercial or Amateur press.[1]

The purpose of this project was to demonstrate that the Weaver modulation technique could be easily and inexpensively applied to direct conversion sideband generation at VHF frequencies. It was not my intention to build a full-function rig, but merely to experiment with the architecture; therefore, the design does not include any T/R switching, ALC circuitry, or digital frequency display. Intrepid homebrewers can easily add these functions themselves.

Despite the fact the "filter" technique of SSB generation has been almost universally adopted for Amateur and commercial design, the Weaver technique offers the following advantages:[2,3,4]

- Much of the circuitry operates at audio frequencies, where layout is relatively non-critical. Components for these applications are inexpensive and easy to obtain.
- There is only one RF oscillator, and it operates at the center of the transmitted output passband rather than being offset by an IF frequency. The oscillator may be tested with ordinary Amateur equipment; a 2-meter receiver can be used as a detector. Also, a conventional frequency counter can be used as a digital frequency readout, since there are no BFO or IF offsets to account for.
- All of the mixers operate on baseband signals. The absence of heterodyne techniques mean that there are (theoretically, at least) no images or spurs. Any out-of-band radiation is a result of mixer and amplifier non-linearities, and not a result of any inherent limitations of the conversion scheme.
- Unlike the "phasing" technique, the Weaver modulator does not depend upon accurate phasing or balancing to achieve good control of the transmitter bandwidth. Phase and balance errors cause degradation of the audio quality only, not out-of-band components.
- No expensive or hard-to-find crystal filter is necessary. For the most part, no unusual components are required; the average junkbox probably contains most of the components needed for the design.

By **Norm Bernstein, N1COX**, 24 Foxfire Drive, Sharon, Massachusetts 02067

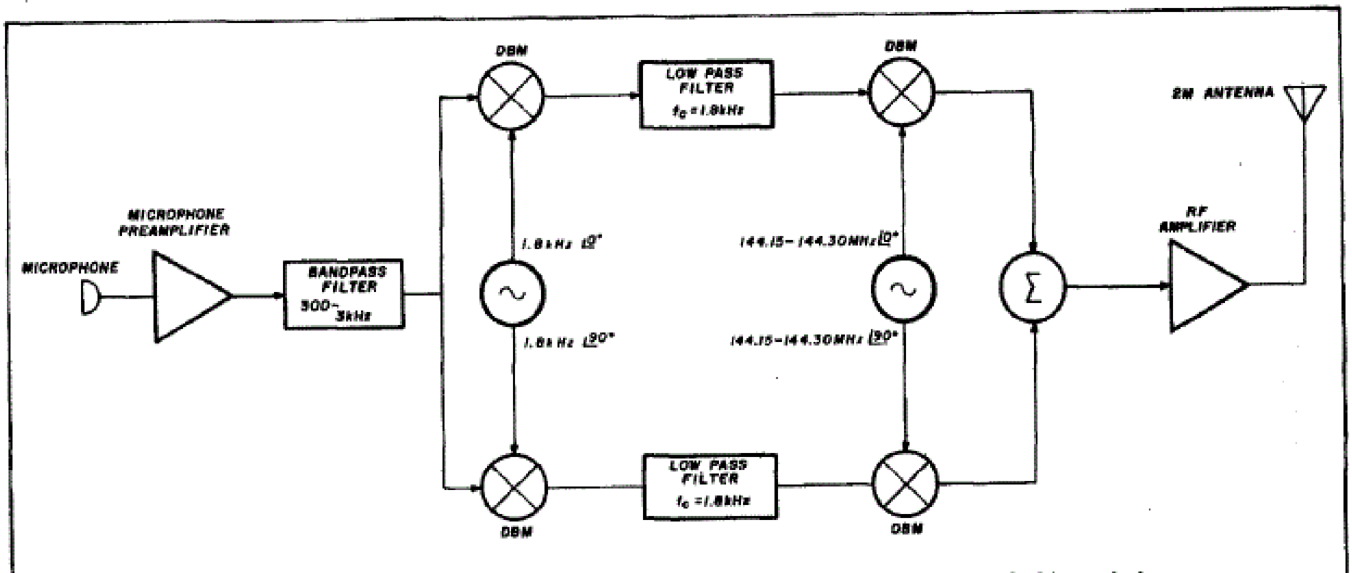


fig. 1. Block diagram of a single sideband transmitter using the Weaver modulation technique.

The only significant disadvantage of the Weaver modulator technique is local oscillator suppression. In a conventional SSB transmitter, any LO leakage is located at the position of the suppressed carrier, and a filter type receiver would normally be tuned to zero beat this signal for proper reception. In the Weaver modulator, however, the LO is set exactly in the middle of the transmitted passband, and good RF mixer balance is essential to avoid an unpleasant "whistle" on the transmitted signal. Fortunately, commercial DBMs have excellent balance characteristics, and the carrier leakage can be dealt with successfully. (Interestingly, this would not be a problem if the intended receiver used the Weaver technique as a demodulator, since the receiver's local oscillator would then zero beat with the transmitter's leaky LO signal, rendering the leakage inaudible.)

circuit description

Figure 1 shows a block diagram of the basic technique. The signal from the microphone is amplified and filtered for the normal 300 Hz to 3000 Hz communications bandwidth. It is then applied to a pair of double balanced modulators; the modulators are driven from an audio frequency local oscillator whose outputs are in quadrature. The AF local oscillator runs at 1.8 kHz, which is the center of the audio passband.

The outputs of the DBMs are then fed to a pair of low-pass filters, each with a cutoff frequency of approximately 1.8 kHz. These filters establish the basic transmitted bandwidth and are analogous to the crystal filter found in conventional rigs.

The outputs of the filters are then sent to another pair of double balanced mixers; these mixers are driven from a quadrature local oscillator operating at the desired RF frequency. The outputs of the mixers are then summed, with the resultant output being a single sideband signal. The signal may then be amplified in a conventional manner before being fed to the antenna.

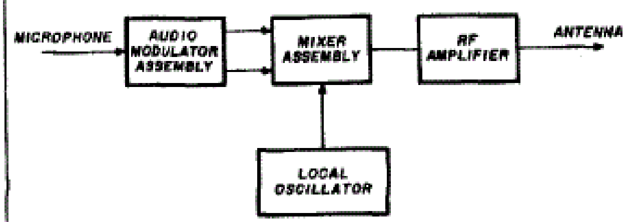


fig. 2. An audio modulator, mixer assembly, local oscillator, and post mixer amplifier comprise the modular construction design approach.

Selection of the upper or lower sideband can be made by switching the phases of either of the local oscillators or by swapping the outputs of either pair of double balanced mixers.

While the actual technique might be difficult to understand, its mathematics are relatively simple. Rather than attempt a complete description of the mathematics at this point, I recommend that interested readers consult the references listed at the end of this article, especially the original paper by Weaver.

designing the prototype

To minimize leakage effects and simplify testing, I decided to split the design of the prototype into four functional blocks: the audio modulator (containing the microphone preamplifier, the audio double balanced mixers, the filters, and the AF local oscillator generating circuitry), the local oscillator, the RF mixer assembly, and the post-mixer amplifier. The audio modulator was housed in an aluminum chassis box, and the remaining three modules were constructed in separate die-cast boxes, using BNC connectors for signals and feedthroughs for DC power. Both **fig. 2** and the photo show the interconnection of the four modules.

The audio modulator assembly performs all the baseband signal processing, producing an output suitable for driving the RF mixers directly. This module has the most complex circuit of the four, but is the easiest to build because the layout is not critical; I used a conventional punched board with sockets for the ICs and point-to-point wiring.

The microphone input is connected to a wideband gain stage (**fig. 3**) in order to bring the audio signal up to the nominal working level (2 to 3 volts p-p). The signal is then fed to a highpass filter in cascade with a low-pass filter. These filters are implemented as third-order Sallen and Key types with cutoff frequencies of 300 Hz and 3 kHz, respectively.

The signal is now split into two paths. Each path consists of a double balanced mixer, followed by a relatively sharp low-pass filter, followed by a buffer stage and 50-ohm pad designed to deliver approximately 0 dBm to the mixers.

The double balanced mixers are implemented with a series/shunt switch (1/2 of a CD4016 CMOS switch) and an op amp configured as an "invert/non-invert" stage. This type of mixer exhibits good linearity and balance at audio frequencies, but has strong spurious response at harmonics of the local oscillator frequency; this is why the microphone preamplifier is followed by a relatively sharp bandpass filter.

The signals from the mixers are then routed to the low-pass filters. The filter characteristic is important

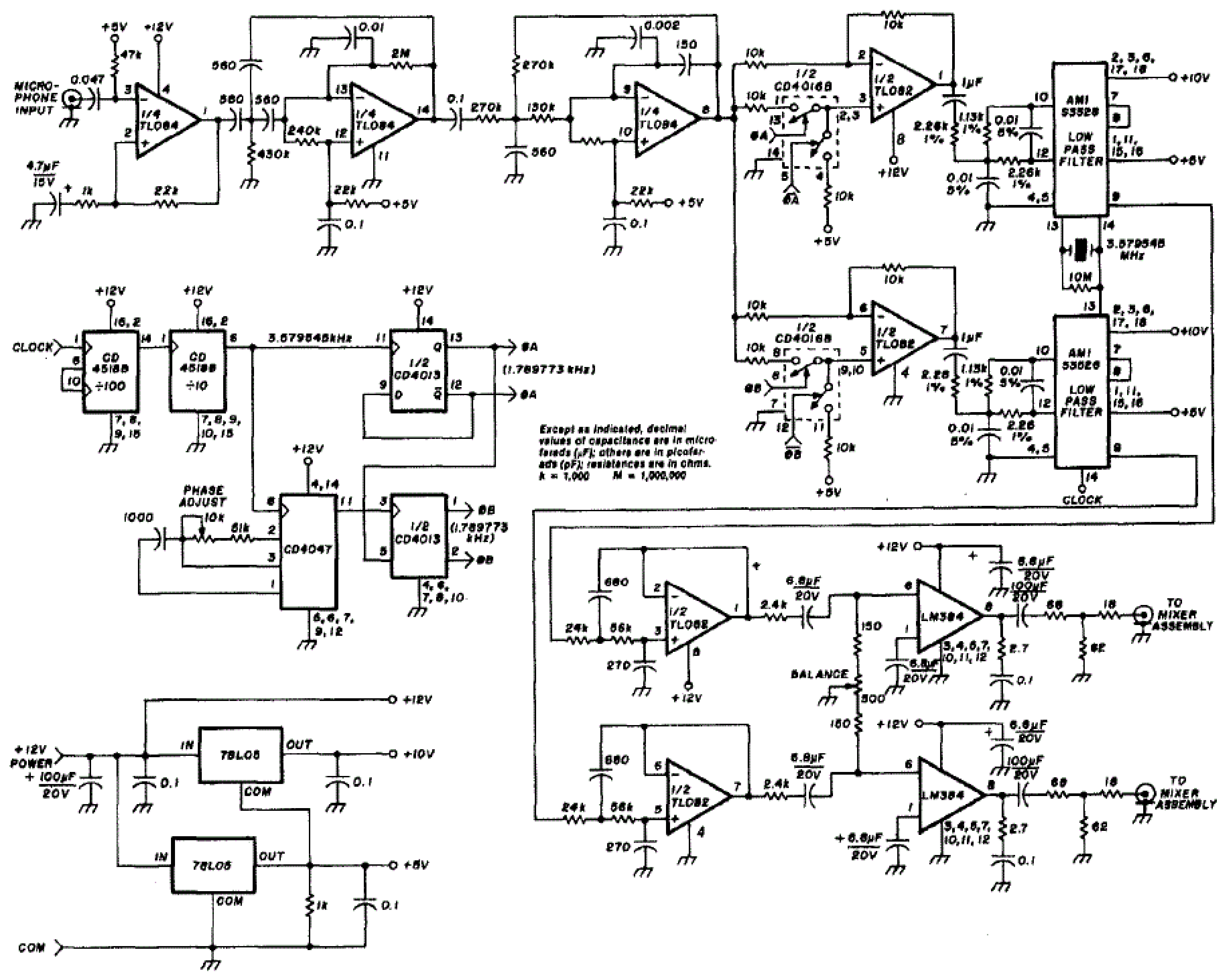
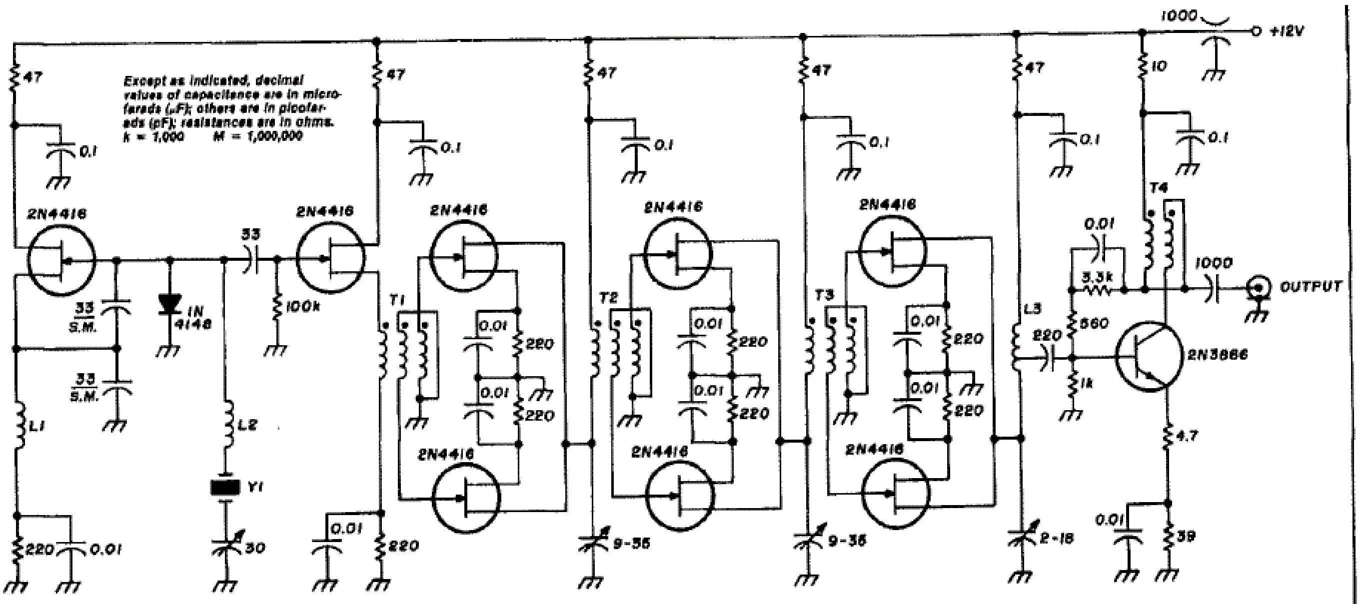


fig. 3. The audio modulator assembly consists of a microphone preamplifier and bandpass filter, a pair of double balanced mixers, and the switched capacitor filters. Also included in this assembly are the 1.8 kHz quadrature local oscillator and a pair of amplifiers for driving the RF mixers.



L1	33 turns No. 24 enameled, FT50-61 core
L2	35 turns No. 24 enameled, T50-6 core
L3	4 turns No. 18 tinned, 1/4" ID x 1/2" long tap 1/2 turn from cold end
T1	12 turns No. 30 enameled, trifilar on FT50-61 core
T2	11 turns No. 30 enameled, trifilar on T37-6 core
T3	7 turns No. 30 enameled, trifilar on T37-6 core
T4	5 turns No. 24 enameled, bifilar on FT37-61 core
Y1	18.025 MHz, parallel resonant, 20 pF load capacitance

fig. 4. The local oscillator is a VXO operating at 18.025 MHz followed by three push-pull doublers and an amplifier. The output is 144.150 to 144.300 MHz at about +17 dBm.

because it affects the degree of unwanted sideband suppression and establishes the bandwidth of the transmitted signal in much the same way that the crystal filter does in a conventional rig. What is needed here is a high order elliptical low-pass filter with good stopband performance. One additional requirement is that the filters in each of the two signal paths have closely matched amplitude and phase characteristics; any significant mismatch here will affect the unwanted sideband suppression (which, in a Weaver modulator, results in a degradation of the audio quality).

While the filters could have been built using conventional LC techniques, I decided to use a pair of switched capacitor filter ICs. This device the S3528 from American Microsystems, Inc.,* is a seventh-order elliptical low-pass filter with a programmable cutoff frequency and better than 50 dB worth of stopband suppression. A pair of these devices is significantly smaller than corresponding passive LC filters and are "tweak free" — i.e., they require no adjustment whatsoever and are inherently well matched. A minor disadvantage to switched capacitor filters is that they require some additional filtering at the input and some filtering at the output to remove the residual clock component from the signal, but this was not difficult to accomplish.

After the filters, the two audio signals go to a balancing network followed by a pair of LM384 driver amplifiers. These amplifiers are power devices capable of driving the 50-ohm pads used to reduce the signal to approximately 0 dBm, a level appropriate for the IF ports of the RF mixers. The heavy attentuation also helps to insure that the mixers see a broadband resistive termination at their IF ports, which is important for proper mixer operation.

The switched capacitor filters contain their own oscillator, which is based on a standard 3.579545 MHz colorburst crystal. This clock signal is divided by 1000 and applied to a pair of flip flops, one of which is delayed by an adjustable one-shot to create a 90-degree phase lag. The output of these two flip-flops is an adjustable quadrature signal operating at 1.789773 kHz, which is close enough to the design value of 1.8 kHz for suitable operation. Feedback from the non-delayed flip-flop is employed to insure consistent phasing at startup; without such feedback, the

*American Microsystems, Inc., a division of Gould, Inc., 3800 Homestead Road, Santa Clara, California 95051.

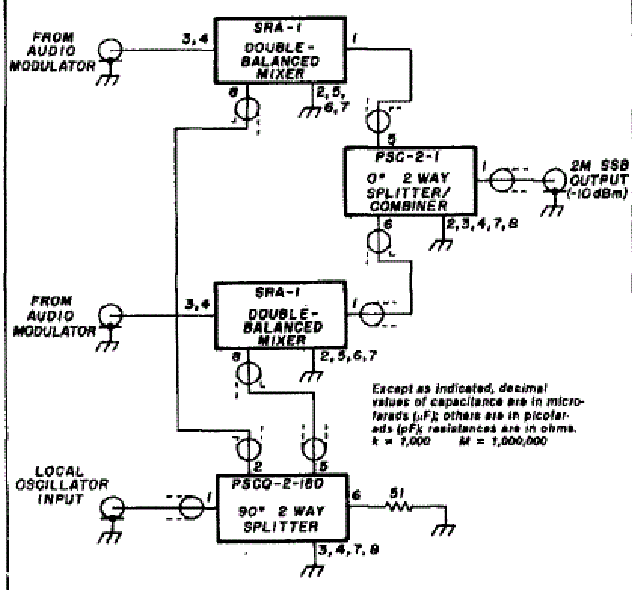


fig. 5. The mixer assembly contains two SRA-1 double balanced mixers, a 90-degree 2-way hybrid (which is used to create the quadrature local oscillator signals for the mixers), and 0-degree, 2-way hybrid, which sums the two mixer outputs into a single SSB signal.

transmitter would choose upper or lower sideband at random!

The AF local oscillator quadrature could have been generated with perfect accuracy through the use of flip-flops alone, but I used the one-shot delay in order to allow for some adjustment range; it can be shown mathematically that small phase errors in the RF mixer assembly can be cancelled by introducing an equal but opposite phase error into the system at the AF mixers.

The RF local oscillator (fig. 4) is a VXO running at a nominal frequency of 18.025 MHz, followed by three doubler stages and a buffer stage. This design is simple to build, adequately stable, and provides for enough tuning range to cover most of the portion of the 2-meter band commonly used for terrestrial communications. My version covers 144.150 MHz to 144.300 MHz; it is possible to obtain a wider coverage, but tuning ranges in excess of 0.1 percent of the nominal output frequency will result in reduced stability.

Although no frequency indicator was constructed for this experimental rig, it would be relatively easy to build one because the oscillator runs at the transmitted frequency; there are no IF or BFO offsets to account for. A general-purpose frequency counter capable of operation at 2 meters can also be employed.

The RF mixer module (fig. 5) consists of a pair of SRA-1 mixers whose local oscillator inputs are driven in quadrature, and whose RF outputs are summed into a single output. The local oscillator drive is obtained from a commercial quadrature hybrid, in this case the PSCQ-2-180 from Mini-Circuits Labs.* The summation of the RF outputs is accomplished with a hybrid combiner (model PSC-2-1). The two audio drive signals are connected directly to the IF ports.

A post mixer amplifier is used to provide 30 dB gain to the −10 dBm 2-meter SSB signal output of the mixer assembly. This results in a signal of about 100 milliwatts, which is sufficient for on-the-air testing. This amplifier (**fig. 6**) is a three-stage device with a grounded gate FET followed by two broadband bipolar class A stages. Because of the relatively low power, I did not incorporate any further filtering of the signal; more would undoubtedly be incorporated, however, in a practical design.

test results

This experimental rig was tested on the air in order to get some subjective feedback on the audio quality. The estimated output power was 50 to 100 milliwatts, too small to be accurately measured on any of my test gear. My first QSO was with W1VDI in Providence, Rhode Island, about 30 miles from my QTH. I received a Q5 report.

Listeners generally reported that the audio quality was essentially equivalent to that of my regular 2-meter SSB rig (an ICOM 251A); minor differences in tone quality were attributed to the use of a different microphone. None of the test participants reported any trace of carrier leakage on the signal, which indicates that the carrier balance of the mixer assembly is adequate.

The only negative comment from the test participants was that there was a brief (2 to 3-second) period immediately after I keyed the transmitter each time, during which some traces of carrier could be heard; this effect "died out" within a few seconds. This was found to be caused by DC bias level settling in the audio modulator section; it can be avoided by maintaining continuous power to the audio modulator rather than attempting to switch the modulator power when the transmitter is keyed.

The transmitter was later examined with an RF spectrum analyzer and found to exhibit a carrier suppression of better than 45 dB, with the reverse sideband component down at least 30 dB from maximum output.

the Weaver technique as a receiver

The Weaver technique is bilateral. If all of the elements can be constructed to operate bilaterally (**fig. 1**), then the system can be used to demodulate SSB signals as well as generate them. Although I have not had a chance to experiment with receive applications, it seems to me that many of the advantages of this technique apply in a demodulation system as well. Images and spurs would be far less of a problem than

*Mini-Circuits Labs, Inc. 2625 East 14th Street, Brooklyn, New York 11235.

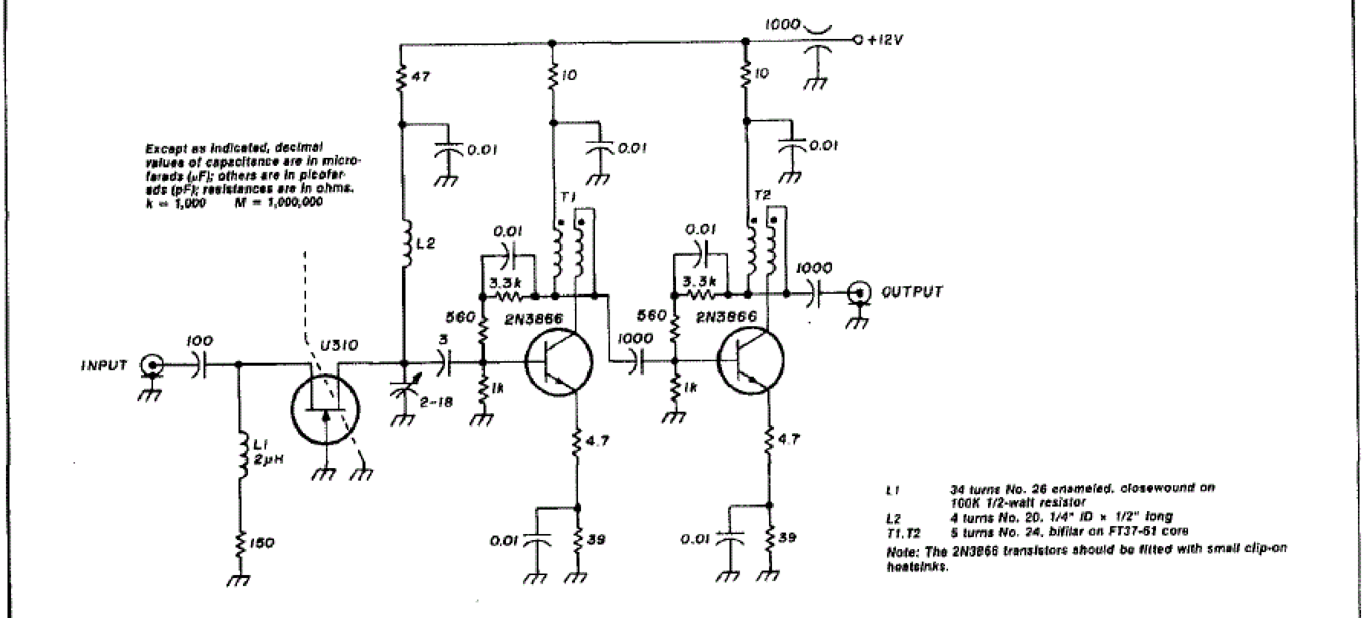


fig. 6. The post mixer amplifier provides about +30 dB of gain at an output of approximately 100 milliwatts.

in conventional heterodyne architectures. Dynamic range should be quite good, since conversion and demodulation occur in the first stage without the need for IF amplifiers, which can overload. All of the gain (with the exception, perhaps, of some RF amplification before the first mixer pair) would be accomplished at audio frequencies, where recent advances in IC processing techniques make low noise audio amplification relatively easy.

low cost variations

It should be possible to reduce the cost of this design by substituting components in the RF mixer assembly and the audio filters. The hybrid combiner and 90-degree splitter could perhaps be replaced by Wilkinson dividers (made from two 1/4-wavelength sections of 75-ohm cable, joined at one end) and a 1/4-wavelength section of 50-ohm cable for the phase delay. The cable scheme would probably have enough bandwidth and accuracy for 2-meter SSB operation, especially in view of the relatively narrow bandwidth popularly used on 2-meter SSB. Precise measurement of the cable lengths would not be necessary, since small amounts of phase error can be "tuned out" with the phase adjustment in the audio modulator section.

The audio filters need not be quite as sophisticated as the ones used in the prototype design; the switched capacitor filters could be replaced with equivalent LC designs. The differential phase performance of the two filters is important, however, for good reverse sideband suppression; it will therefore be necessary to measure the component tolerances of the Ls and Cs quite carefully.

special consideration

Any practical application of the Weaver modulator will require some special design consideration. For example, when using the prototype transmitter in conjunction with a conventional "filter type" receiver, leakage from the transmitter's local oscillator would overload the receiver front end during reception. One way to minimize this problem would be to disable the VXO multiplier stages during receive. In the interests of stability, however, it would not be advisable to key the VXO itself.

acknowledgements

I would like to thank W1VDI, K1TOZ, and WA9WTK/1, all of whom participated in the on-the-air testing phase of this project. I would also like to thank Ed Wetherhold, W3NQN, for his advice on the subject of audio filters.

references

1. Donald K. Weaver, "A Third Method of Generation and Detection of Single-Sideband Signals," *Proceedings of the Institute of Radio Engineers*, December, 1956, pages 1703-1705. (This issue was a landmark for SSB development, as it contains a number of now-famous articles, including the well known Norgaard articles on the phasing technique of SSB generation and detection.)
2. Howard F. Wright, "The Third Method of SSB," *QST*, September, 1957, pages 11-15.
3. J.F.H. Aspinwall, "The Third Method — A New System of SSB Generation," *Wireless World*, January, 1959, pages 39-43.
4. Herbert Krauss, et al, *Solid State Radio Engineering*, John Wiley and Sons, 1980, pages 233-234.
5. Joseph Sansone, "Get High-Q in Active Bandpass Filters with a Quadrature Modulation Scheme," *Electronic Design*, November 8, 1978, pages 124-127.

ham radio