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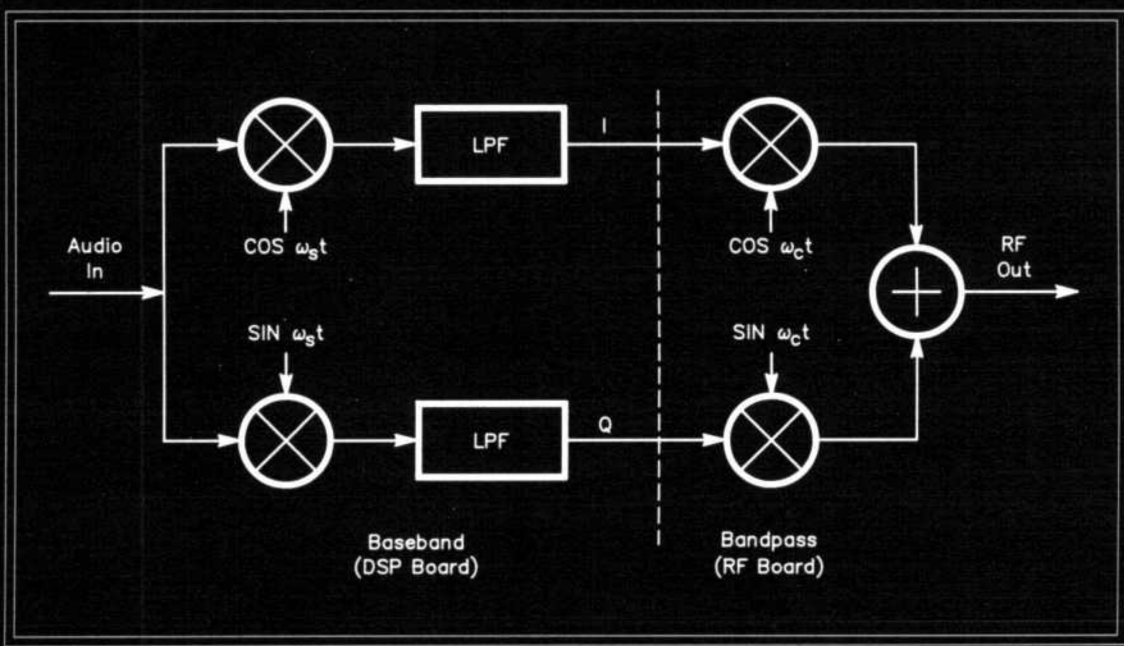
September 1993



Receiving and Transmitting SSB with the Weaver Method

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About the Cover:

A Weaver-method receiver and DSP-based SSB generation bring this old-fashioned technique right up to date.

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Features

Columns

September 1993 QEX Advertising Index

Empirically Speaking

Making PC Boards Easier

You may have noticed in the past few issues of *QEX* that we have made a change in the way we print printed-circuit board patterns. In the past, we always printed them "right reading," meaning that the image on the page is the image of the copper as you look at the side of the circuit board that contains the circuit. This results in, for example, any text etched into board reading correctly on the page. Recently, we began printing the patterns "wrong reading," such that the image on the page is the image you would get from a x-ray view of the board from the other side. In this case, text is reversed. (Actually, we print them both ways if space permits. When we don't, just send an SASE if you want the other image.)

Why the change? Principally, it's due to a change in the way home builders make boards. Previously, most builders used photographic techniques to make boards. They would first make a transparency from the published board pattern, using a photocopier or camera to do the job. This transparency would then be used to make a "contact" exposure of a copper-clad board coated with photosensitive resist material.

Given the resolutions needed (and possible) for these home building techniques, it really didn't matter which way we printed the PC pattern; you could always correct it by flipping the transparency over as needed. So, we chose to print patterns such that the text on the board was readable.

Recently, though, the development of iron-on PC techniques has led to a substantial use of that technique. Iron-on PC boards are made by photocopying the PC pattern on special material, such as TEK film or DynaArt material (see "Correspondence," in the June 1993 *QEX*). Using an iron, the toner on the material is then ironed on to the PC board, where it acts as a resist during etching. But for this to work, the copy must be "wrong reading," so that the toner, when applied to the board, gives the correct circuit, not a backwards copy of it.

With the old-style pattern, you would first have to make a reversed copy of the published pattern, *then* the iron-on copy. Now, one copy does the trick.

But copiers aren't particularly accurate reproduction devices. Often, copies are slightly larger or smaller than the original. This can be a serious problem on a large PC board, or for a microwave circuit. So an even better technique is to print the iron-on copy from a laser printer. To facilitate this, we are now making available, whenever possible, Postscript copies of PC layouts. We can't promise to do this every time; some boards aren't easily rendered in Postscript. But when we can, we will. If there are other changes we can make to help your home building activities, please, let us know.

By the way, other ARRL publications are now doing the same thing. When PC templates are printed or made available from the Technical Department, both right- and wrong-reading images will be made, whenever possible.

This Month in *QEX*

The Weaver method of SSB generation and detection is discussed in two articles this month, "A Different Weave of SSB Receiver," by Peter Traneus Anderson, KC1HR, and "A Weaver Method SSB Modulator Using DSP," by Carlos M. Puig, KJ6ST. Each of these articles is a follow-up to Anderson's August 1991 *QEX* article describing a Weaver-method exciter.

Need a counter prescaler that works in the 10-GHz band? Such is described by Angel Vilaseca, HB9SLV and Serge Rivière, F1JSR, in "A 12 GHz, 1/8 Prescaler."

Larry Brewster, NØRNX, and Dennis Bodson, W4PWF, present part 3 of our series on ALE, describing the continuing standards development effort, and comparing the federal and amateur worlds.

In his "RF" column, Zack Lau, KH6CP/1, reports on his visit to the Central States VHF Conference, and presents a 13-cm MMIC amplifier that produces +17 dBm out.—*KE3Z, email: jbloom@arrl.org (Internet)*

A Different Weave of SSB Receiver

Using the Weaver method in a receiver requires some care, but it's very doable.

Peter Traneus Anderson, KC1HR

This article describes an experimental 75-meter Weaver-method single-sideband (SSB) receiver built by combining parts of my previously described Weaver-method exciter with two of KK7B's R1 direct-conversion receiver boards.[1,2,3] Although a first-try receiver rather than an optimized design, the SSB receiver works well.

A Weaver-method SSB receiver offers advantages over filter or phasing-method receivers. Filter-type receivers tend to have broad skirts, due to limitations in mechanical or crystal filters. Weaver-method receivers have sharp skirts, due to the sharp audio-frequency low-pass filters used to define the passband.

Filter-type receivers require a minimum of two frequency conversions, one

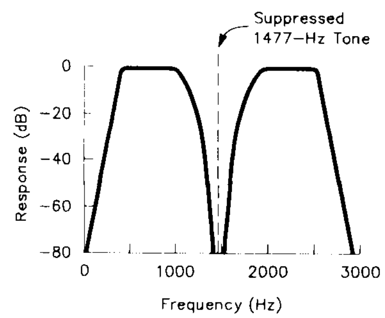

Fig 1—Spectrum of Weaver receiver audio response. The dashed line at 1477 Hz is the notched-out CW audio tone. The RF response is the same, with the local oscillator frequency replacing the 1477-Hz tone.

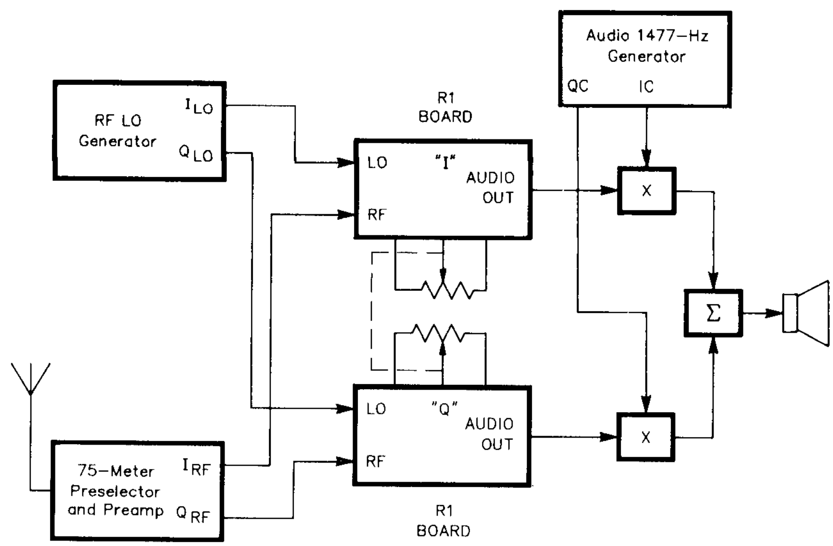

Fig 2—Block diagram of Weaver receiver. I and Q outputs of the LO and 1477-Hz generators are 90 degrees apart. I and Q outputs of the preamp are in phase with each other. The volume controls on the two R1 boards are ganged together.

[1]Notes appear on page 7.

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before the filter and one after the filter (this latter conversion, to audio frequencies, is often called product detection). Many filter-type receivers have two or more frequency conversions before the filter. Filter-type receivers also need additional filters or conversions to permit selecting upper and lower sidebands.

Weaver-method receivers require only two frequency conversions, one of which is performed entirely between dc and 3 kHz. A direct-conversion Weaver-method receiver is entirely practical.

Phasing-method receivers, which use a system related to the Weaver method, have limited rejection of the unwanted sideband on the wrong side of the local oscillator frequency due to phase and amplitude mismatches. Weaver-method receivers have similar mismatches which generate similar wrong-side responses. In the Weaver method, however, the local oscillator is in the center of the passband. Thus, wrong-side responses are inside the passband and are much less noticeable than out-of-passband responses.

A Weaver-method receiver suffers from one severe unwanted response: dc offsets in the low-pass filters or in the mixers generate a strong continuous tone in the center of the passband. I once thought this was an incurable problem.

A perusal of the *1981 Radio Amateur's Handbook* disclosed a description of Narrow-Band Voice Modulation (NBVM).[4] In NBVM, the middle part of the audio passband is omitted,

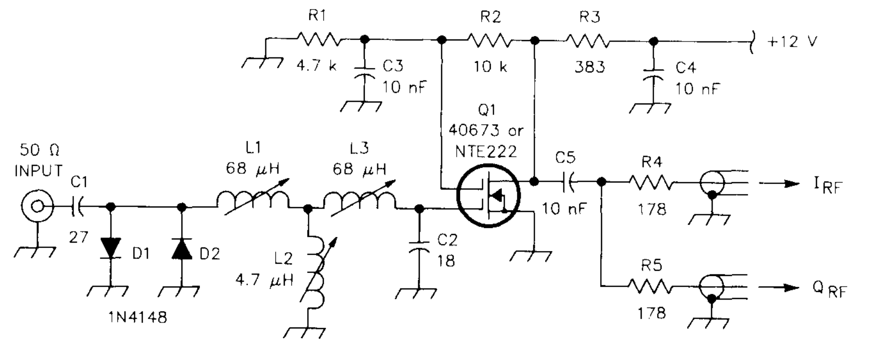

Fig 3—75-meter preselector and preamp. The NTE Semiconductor NTE222 is a replacement for the discontinued RCA 40673.

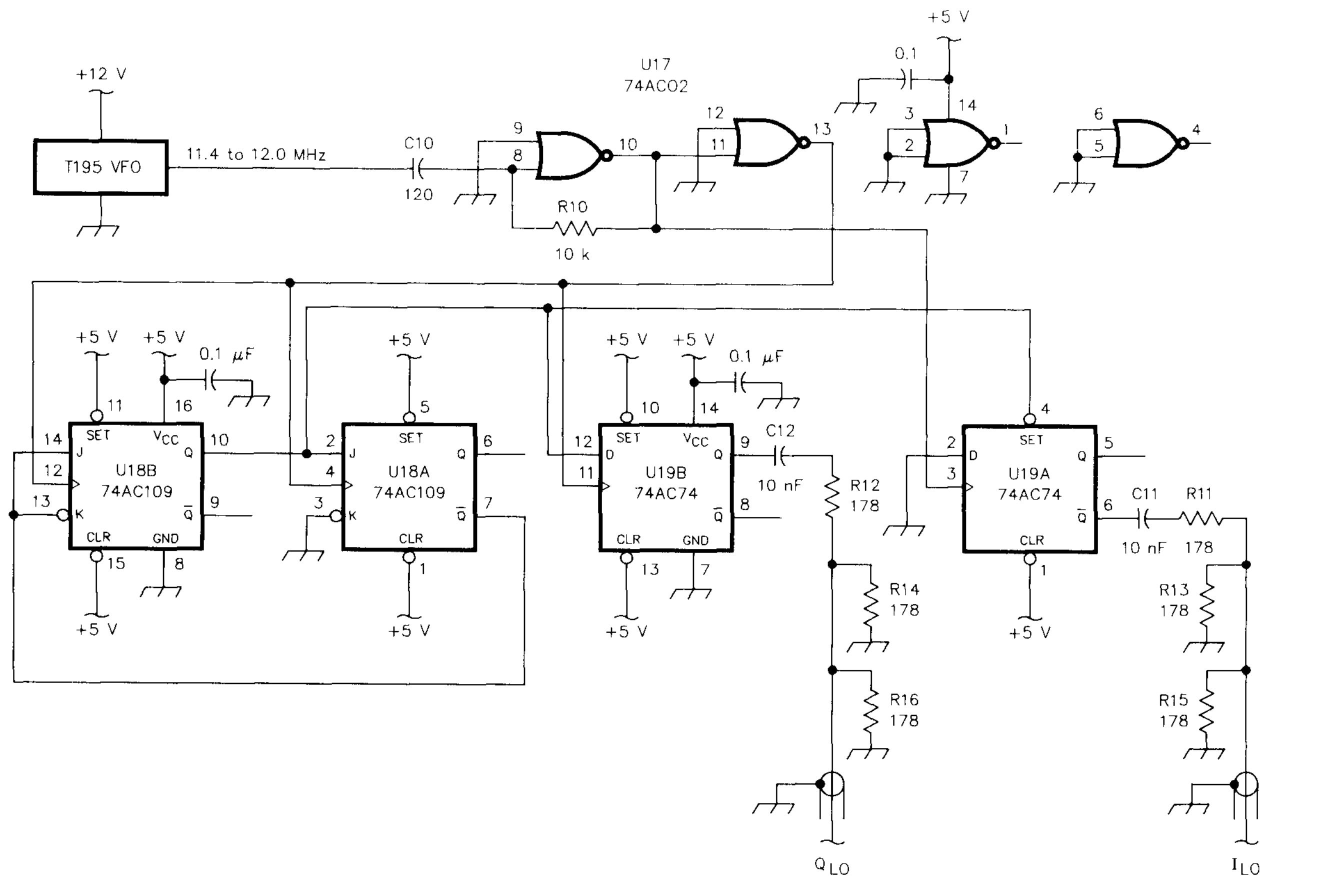

Fig 4—75-meter local-oscillator generator. The flip-flops divide by three. The two outputs are 90 degrees apart. The I output is a square wave. The Q output is low ⅓ of each cycle and high ⅔ of each cycle.

and the remaining parts are rearranged to be contiguous. The spectrum of speech shows that there are spectral gaps with little energy, so omitting the gap regions from the passband has little effect on speech intelligibility.

Thus, a notch or gap in the center of the passband eliminates the troublesome CW tone without impairing speech intelligibility.

One version of KK7B's R1 receiver has a passband from 300 Hz to 1000 Hz. The high side of the passband is set by a seventh-order elliptical low-pass filter which is 60 dB down at 1400 Hz, perfect for a Weaver receiver with an audio carrier at 1477 Hz.

The overall passband is shown in Fig 1. The passband is split into two subbands, 477 Hz to 1177 Hz and 1777 Hz to 2477 Hz, as explained in my previous article (see note 2). The gap between 1177 Hz and 1777 Hz is not very noticeable on speech. The gap is obvious when I tune a CW carrier across the passband.

The passband is 60 dB down at 77 Hz and 2877 Hz. Thus the shape factor is (2877-77)/(2477-477) = 2800/2000 = 1.4:1. These skirts are symmetrical in linear frequency, unlike those of conventional filter or phasing receivers, which are usually wide on the high-audio-frequency side of the passband.

The receiver block diagram is shown in Fig 2. The local oscillator and 1477-Hz generators are borrowed from my Weaver exciter (see note 2).

I found that a preselector and preamp, shown in Fig 3, are necessary ahead of the R1 boards. The two outputs of the preamp are in phase; the preamp outputs are labeled I and Q to refer to the two R1 boards which receive the preamp outputs.

The preselector is tuned by peaking L1 and L3 in the center of the 75-meter band. L2 need not be tuned.

The preamplifier uses an RCA 40673. This discontinued device can be replaced by a NTE Semiconductor device, NTE222. Isolation of the R1-board mixer inputs is provided by R4 and R5.

The local-oscillator generator, shown in Fig 4, starts with a 11.4- to 12-MHz VFO.[5] The first two flip-flops divide the VFO frequency by three, to give 3.75 to 4 MHz. The output of the third flip-flop is a ⅓-low, ⅔-high waveform. The fourth flip-flop provides a square wave 90 degrees out of phase with the output of the third flip-flop. R11 through R16

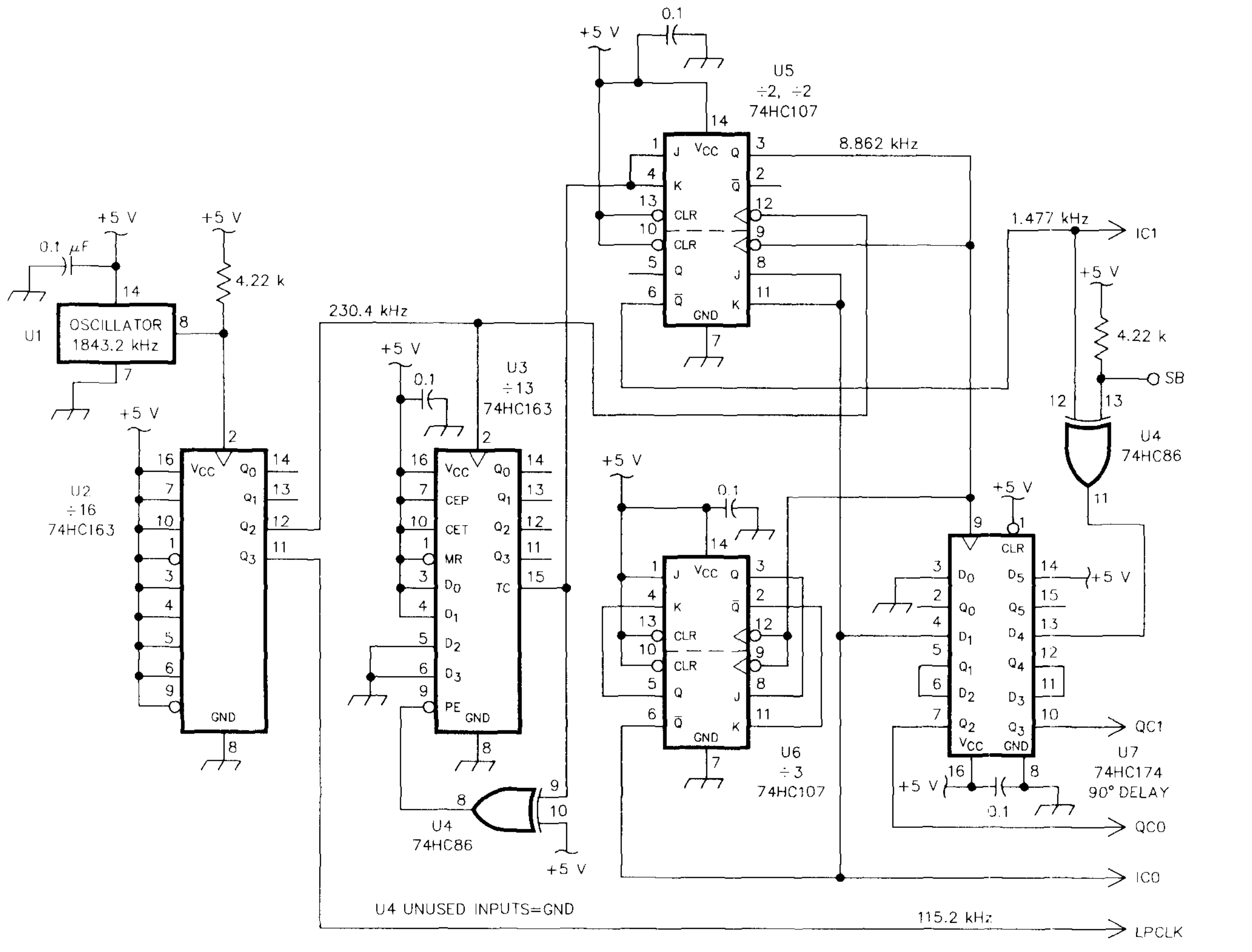

Fig 5—Audio 1477-Hz generator. This is identical to Fig 5 in note 2. The LPCLK output is not used in this receiver.

provide 50-Ω local-oscillator signals to the SBL-1 mixers on the R1 boards.

The asymmetrical Q output from the third flip-flop, has less fundamental-frequency component than has the symmetrical I output from the fourth flip-flop. This results in the SBL-1 in the Q-channel R1 board having more conversion loss than the SBL-1 in the I-channel R1 board. The Q-channel R1 board has higher gain in the U2B stage to compensate for the added conversion loss.

The two R1 boards of Fig 6 use KK7B's 1000-Hz seventh-order elliptical low-pass filters. The two boards have their 500-Ω volume controls ganged together. I could not obtain a dual 500-Ω control, so I used a Radio Shack dual 100-kΩ control, with a 500-Ω resistor across each section.

The audio power stages of the R1 boards, Q5 through Q9, are omitted, and R21 is connected directly to U2B's output (U2 pin 5). The values of R20 and R21 are increased to provide more gain. These values are larger on the Q board to provide the higher gain mentioned above.

Capacitors C30 and C31 ac-couple the R1 audio outputs to the audio mixers, U8 and U9, to eliminate the effects of dc offsets in U2 on the R1 boards. Germanium diodes D31, D32, D33, and D34 protect CMOS mixers U8 and U9 from voltage transients as power is turned on and off.

The clock generator of Fig 5 is identical to Fig 5 in my exciter article (note 2). This circuitry generates the 1477-Hz audio carriers for audio mixers U8 and U9 (Fig 6).

IC1 and IC0 comprise a two-bit digital local-oscillator signal to double-balanced mixer U8. QC1 and QC0 go to double-balanced mixer U9. These are two-bit signals rather than square waves, so the outputs of U8 and U9 will lack frequency components at the second, third, or fourth harmonics of the 1477-Hz audio carriers mixing with the

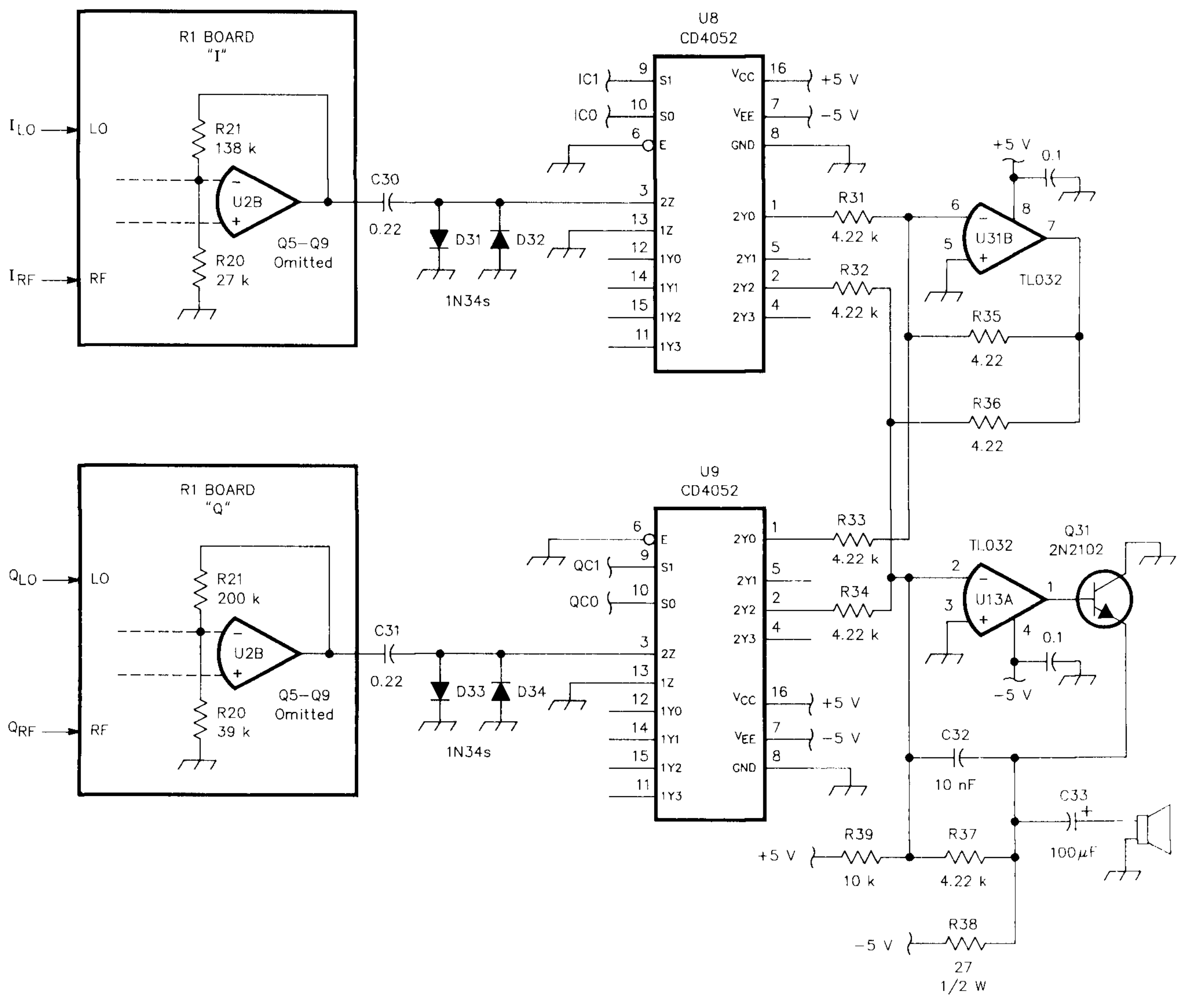

Fig 6—R1 boards, audio mixers, and audio stage of the receiver. The two R1 boards are wired for different gains to compensate for the different duty cycles of the I and Q outputs of the local oscillator generator.

outputs of the R1 boards. See note 2 for more details.

Grounding the node labeled "SB" in Fig 5 switches the receiver from lower sideband to upper sideband reception by changing the phase of ICO by 180 degrees. The output labeled "LPCLK" in Fig 5, is not used in the receiver. This output is used to clock active filters in the exciter in note 2.

Mixers U8 and U9 are CMOS analog multiplexers. The digital local-oscillator signals drive the select inputs of U8 and U9. The 1477-Hz cycle is divided into six phases, numbered 1 through 6, in time order. I'll describe the operation of U8; U9 operates similarly.

During phases 1 and 2, multiplexer output 0 is selected, connecting U8 pin 3 to U8 pin 1 and leaving pin 2 open. During phases 4 and 5, multiplexer output 2 is selected, connecting U8 pin 3 to U8 pin 2 and leaving pin 1 open. During phases 3 and 6, pin 1 and pin 2 are both open.

The two op-amps of U31 sum the mixer outputs. Resistors R31, R32, R33, R34, R35, and R36 must be well matched as they determine the balance of the audio double-balanced mixers. I used 1% resistors here.

Op-amp U31B, with R35 and R36, forms a current mirror. The current from R36 into U31A pin 2 equals (minus) the current, from R31 and R33, into U31 pin 6.

Op-amp U31A forms an inverting unity-gain low-pass filter stage. Q31 provides a current boost so the output can drive low-impedance headphones or a small speaker. R39 provides a dc current into U31 pin 2 so Q31 runs class A and draws power only from the –5-V supply.

Noise from the +5-V supply will enter the audio signal through R39. My supply is quiet, so I have no problem, but a noisy supply may need filtering.

The audio signal at the emitter of Q31 equals the voltage on U8 pin 1, minus the voltage on U8 pin 2, plus the voltage on U9 pin 1, minus the voltage on U9 pin 2. This provides the double balance for the mixer action of U8 and U9.

All of the receiver's gain is before audio mixers U8 and U9, to minimize the audibility of the 1477-Hz tone from mismatches in mixers U8 and U9.

The receiver works well; clean signals are very intelligible. Some signals, of course, are transmitted distorted, and some are buried in noise or interference. Audio fidelity is not as good as that of a receiver with a wider passband.

The in-passband spurious responses show up as frequency-inverted speech. These are noticeable when listening to a CW carrier, but are hardly noticeable when listening to speech, as W1PNB reported for a Weaver exciter.[6] Tweaking the gain ratio on one R1 board versus the other would reduce the level of inverted speech. The 1477-Hz tone is barely audible at full gain. The skirts are excellent, and the passband sounds just wide enough for speech.

Notes

1 Weaver, D.K., "A Third Method of Generation and Detection of Single-Sideband Signals," *Proceedings of the IRE*, Dec 1956. Note that this whole issue was devoted to single sideband!

2 Anderson, P.T., "A Different Weave of SSB Exciter," *QEX*, Aug 1991, pp 3-9. Note error in Fig 7: a 4220-Ω resistor should be inserted in the wire between U11 pin 7 and U11 pin 2.

3 Campbell, R., "High-Performance Direct-Conversion Receivers," *QST*, Aug 1992, pp 19-28.

4 *The Radio Amateur's Handbook*, 1981, pg 14-33 (ARRL: Newington, CT).

5 Anderson, P.T., "Transistorizing Surplus VFOs," *QST*, Feb 1989, pp 45-46.

6 Wright, H.F., "The Third Method of SSB, How It Works in Theory and Practice," *QST*, Sept 1957, pp 11-15.

A Weaver Method SSB Modulator Using DSP

DSP eliminates the audio phase-shift and dc offset problems of analog Weaver-method modulators.

Carlos M. Puig, KJ6ST

Abstract

Single sideband (SSB) remains the primary Amateur Radio HF communications mode. Of the three known SSB generation methods, the Weaver method has received the least attention. A recent design showed that a Weaver modulator can be readily built with current technology; however, its prototype suffered from performance limitations arising from the characteristics of high-order switched-capacitor filters. (See Note 5.) The Weaver SSB modulator design presented in this paper overcomes these limitations by using digital signal processing (DSP) for critical baseband signal-processing steps. The modulator's measured performance meets applicable ARRL and FCC guidelines and compares favorably with that of commercial units.

Introduction

Single-sideband suppressed-carrier (SSB-SC, or just SSB) is the most commonly used modulation mode for Amateur Radio voice HF (3-30 MHz) communications. SSB is also the dominant mode for aeronautical, marine, and unencrypted military voice HF

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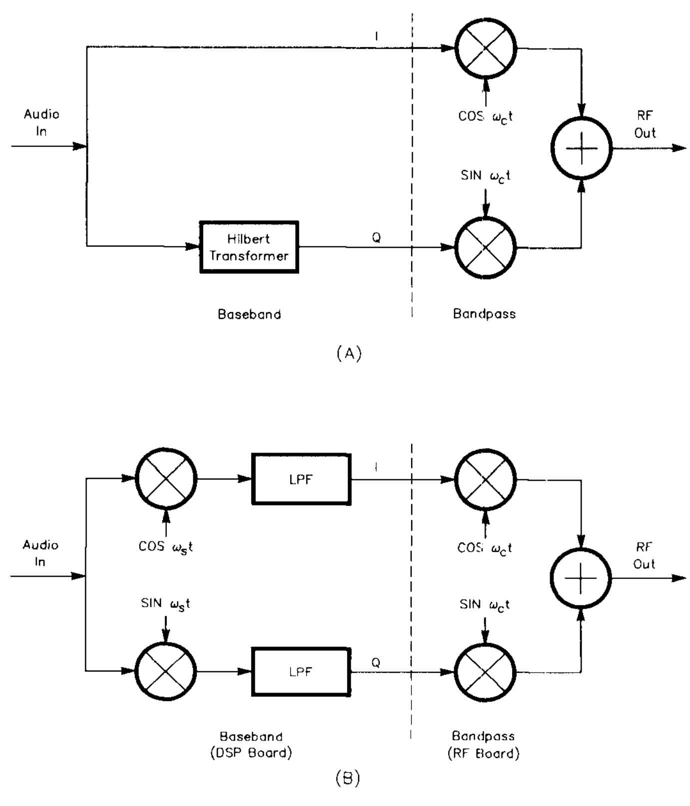

Fig 1—Phasing (A) and Weaver (B) SSB modulator block diagrams.

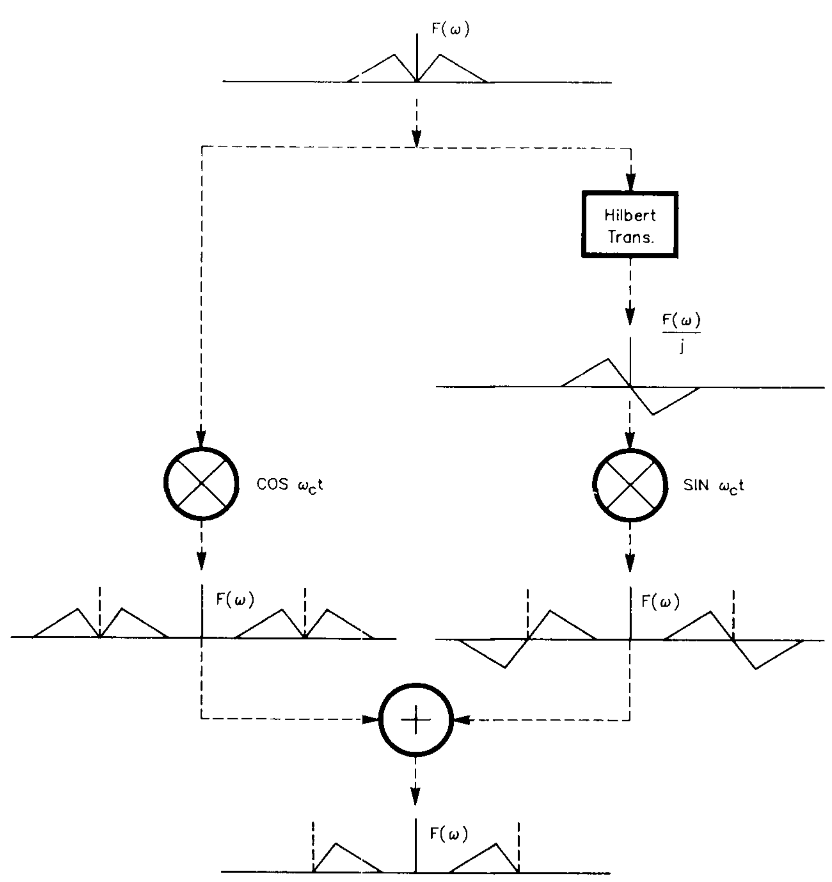

Fig 2—The phasing method in the frequency domain.

communications. SSB provides the best spectrum utilization and power efficiency of all analog communications modes and is less affected than AM by frequency-selective multipath fading.

Nearly all current SSB modulators use the *filter method*, in which an SSB signal is obtained from a double-sideband suppressed-carrier (DSB-SC) signal by use of a bandpass filter that selects only the desired sideband. Filter-method modulators offer good stability over temperature and time. However, they typically require the use of bandpass filters with eight or more accurately cut quartz crystals, and they are limited to operation at a single carrier frequency. Additionally, these crystal filters often suffer from poor phase characteristics. A second method of SSB generation, called the *phasing method*, can operate over a wide carrier frequency range. It is difficult to obtain good undesired sideband suppression (>40 dB) with the phasing method, and it is even more difficult to maintain that suppression level over time and temperature. Although the phasing method was popular briefly in the early days of SSB, it survives today primarily in textbooks (as an introduction to the Hilbert transform and analytic signals) and in digital signal processing (DSP) implementations, where aging and temperature stability aren't problems.

The third method of SSB generation was introduced by Weaver in 1956.[1] The Weaver method is rarely mentioned in communications textbooks, although it has appeared in more specialized technical publications.[2,3,4] Because it requires two closely matched, sharp-cutoff audio low-pass filters, it has historically not yielded good performance at reasonable cost. However, recent innovations in both analog and digital audio filtering have brought the Weaver method within the realm of practicality.

The project described in this report, which I did as my EE senior design project, was inspired by Anderson's 1991 revival of the Weaver method.[5] Anderson's design relies on the Linear Technology LT1064-1 8th-order elliptic switched-capacitor filter, which gives him the required 1.28-to-1 ratio between the stopband and passband frequencies. Although Anderson presented no sideband suppression measurements, his circuit appears to have achieved limited sideband suppression. Furthermore, Anderson reported only 40 dB of carrier and spurious-emission suppression, with higher carrier harmonic levels.

A review of Anderson's circuit suggests that Anderson's sideband suppression is limited by the phase mismatch between the two filters. The phase response of the two low-pass filters must be matched within 1 degree over the entire passband in order to achieve 40 dB of undesired sideband suppression, a standard signal-quality guideline for amateur SSB transmitters *(Handbook,* pp 18-4 and 18-8).[6] Anderson was forced to use both coarse and fine manual dc-offset trims to cancel the filters' large dc offsets, which would otherwise have appeared in the modulator's output as carrier power.

The primary object of this project was to obtain improved opposite sideband and carrier suppression in a Weaver method SSB modulator by using DSP for the required baseband signal processing. A traditional analog design is used for the RF (bandpass) signal processing. A secondary objective was to develop a simple DSP development board that can be reused in future communications projects.

Theory

Fig 1 shows a comparison of the phasing and Weaver methods of SSB modulation. In each case the audio input is a speech signal that is bandlimited to approximately 300-3000 Hz. The output is an SSB modulated signal with a suppressed carrier of angular frequency ω_c.

The phasing method requires a Hilbert transformer to obtain a –90° phase shift at unity gain over the entire speech band. The resulting I ("inphase") and Q ("quadrature") baseband signals are applied to a pair of doubly balanced modulators that act as four-quadrant analog multipliers. The local (carrier) oscillator signals applied to these modulators are also generated in quadrature. The desired SSB output is the sum of the two DSB-SC signals produced by these modulators.

The Weaver method uses exactly the

[1]Notes appear on page 13.

same bandpass processing as the phasing method. However, the method for generating the baseband I and Q signals differs significantly. Instead of a Hilbert transformer, the Weaver method uses a pair of doubly balanced modulators followed by low pass filters. The baseband modulators operate with quadrature audio subcarriers at a frequency (ω_s) of 1.5 kHz, in the middle of the speech band. The low-pass filters must have an essentially flat frequency response from 0-1.2 kHz, with high stopband attenuation at 1.5 kHz.

Fig 2 illustrates the operation of the phasing method in the frequency domain. For ease of discussion, the input baseband signal at the top is assumed to have a real spectrum. Note that the label $F(\omega)$ on each of the vertical axes does *not* imply that each graph represents the same signal; instead, these labels merely indicate a frequency spectrum representation of the signal at that stage. The output of the Hilbert transformer has a purely imaginary spectrum, so that the corresponding graph shows $F(\omega)/j$ rather than $F(\omega)$. The suppressed RF carrier is denoted by a dotted line. In this illustration, a lower sideband (LSB) signal is obtained at the output because the upper sideband (USB) components at the summer's inputs have equal magnitudes and are 180° out of phase.

Fig 3 shows the operation of the Weaver method in the frequency domain, with the same notational conventions as Fig 2. Note that the output of each subcarrier modulator shows a 50% overlap between the translated copies of the input spectrum. This overlap is a form of controlled aliasing that is an essential feature of the Weaver method. Signal components above the subcarrier frequency of 1.5 kHz are removed by the low-pass filters. The output modulators and summer operate as they did in Fig 2, but yield the USB.

In both the phasing and Weaver methods, any amplitude or phase errors at the summer's inputs will result in an undesired sideband of finite power. A comparison of Figs 2 and 3 reveals two important differences between the phasing and the Weaver modulators:

(1) In the phasing method, the undesired sideband appears outside of the communications channel, on the other side of the suppressed carrier. Thus, the inevitable loss of modulator I/Q amplitude or phase balance that occurs with aging and temperature variations leads to spurious signals that may cause adjacent channel interference. This weakness is the main reason why the phasing method quickly lost ground to the filter method. In the Weaver method, the undesired sideband appears within the communications channel, as a frequency-inverted copy of the desired sideband, so that the undesired sideband is heard as low-level distortion of the desired signal.

(2) In the phasing method, a partially suppressed carrier appears just outside of the SSB signal. The partially suppressed carrier is rejected at the receiver and, therefore, is not heard. In the Weaver method, the partially suppressed carrier appears at the center of the SSB signal, so that this carrier is heard at the receiving station as a low-level 1500-Hz tone.

Thus, the phasing method's potential for harmful out-of-channel emissions is absent in the Weaver method, which exhibits only a potential for benign in-channel audio distortion products.

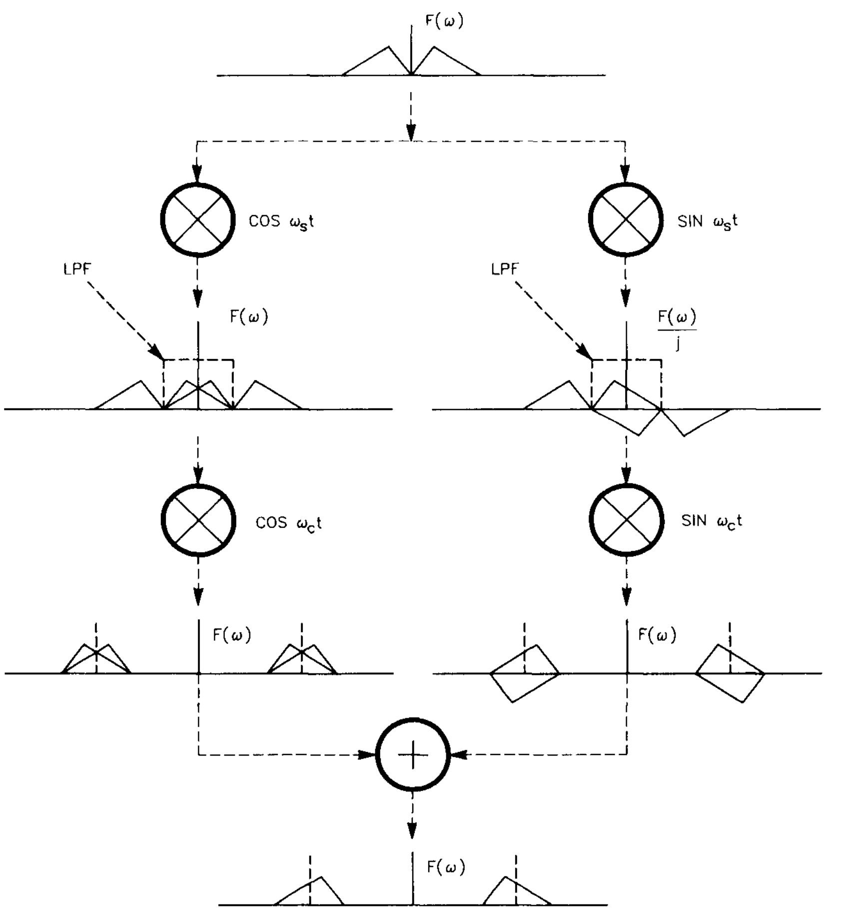

Fig 3—The Weaver method in the frequency domain.

Modulator Design and Operation

RF Board

Fig 4 shows a block diagram of the RF board, which implements the bandpass portion of the Weaver method, as designated in Fig 1(b). The circuit was built on a 4" × 5" prototyping board with a single ground plane, using a mixture of wire-wrap and point-to-point soldered wiring. The schematic is shown in Fig 5.

The two balanced modulators use the classic MC1496 in a circuit adapted from a Motorola data sheet.[7] The fixed-frequency quadrature local oscillators are implemented with a 28.322-MHz TTL oscillator module and a dual D flip-flop (74AS74), using the simple circuit described in the *Handbook* (p 18-10) and note 4. With this circuit, the carrier frequency (7.08 MHz) is one quarter of the oscillator frequency.

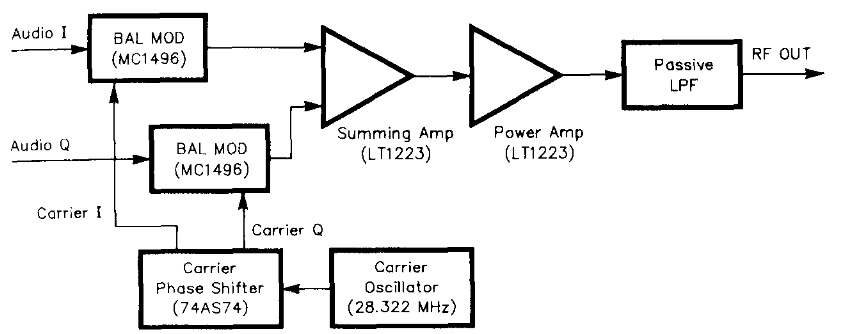

Fig 4—RF board block diagram.

The output amplifier uses two Linear Technology LT1223 100-MHz-bandwidth current-feedback op amps. The first stage operates as a unity-gain inverting amplifier. Higher gain cannot be obtained from this stage without unduly reducing the input impedance. The second stage provides a voltage gain of 7.

The output low-pass filter is a passive LC 5th-order Chebyshev design taken from the *Handbook* (p 2-44). The filter inductors were hand-wound on powdered-iron toroid cores. The inductance of each inductor was verified by measuring the frequency of parallel resonance with a 1% tolerance capacitor.

The RF board contains several trimpots: two for the MC1496 carrier balance (R5 and R15) and one for the overall amplitude balance at the summer's input (R29). The trimpots are adjusted for maximum carrier and undesired sideband suppression.

DSP Board

Fig 6 shows a block diagram of the DSP board, which implements the baseband portion of the Weaver method, as shown in Fig 1(b). This circuit was wire-wrapped on a 4" × 10" prototyping board with two ground planes. A schematic of the DSP board is available from ARRL HQ.[8]

The DSP board includes a single audio input channel, composed of a Maxim MAX291 8th-order Butterworth lowpass switched-capacitor anti-aliasing filter and a 12-bit MAX181 analog-to-digital converter. Two identical audio output channels, each comprising a MAX291 reconstruction filter and a 12-bit MAX507 digital-to-analog converter, supply the output baseband I and Q signals.

The TI TMS320C26, which serves as the digital signal processor, includes 1.5k 16-bit words of internal RAM, which is used for no wait-state execution of the Weaver method software. This processor executes a single FIR filter tap (multiply, accumulate, and data move) in 100 ns. The DSP software is initially loaded into the external 2k words of 100-ns CMOS static RAM by the PC interface and is then copied to the internal processor RAM as part of the reset initialization sequence. The processor runs with one wait state on external RAM and I/O accesses, at a basic instruction time of 200 ns.

The external RAM is based on the Dallas Semiconductor DS1220, which is a hybrid device that incorporates a CMOS SRAM, battery-backup controller, and 10-year lifetime lithium battery in a single package. Thus, any program stored in the external SRAM is retained after power is removed from the board. During normal operation, the SRAM is connected to the processor bus. On command from the host PC, the processor is put in a hold state and isolated from the SRAM, while a new program is loaded from the host PC via a parallel (printer) port interface. The program is loaded sequentially into the SRAM by means of a counter-based address generator. In addition to this byte-wide inbound path, the PC interface includes a 1-bit serial outbound path that can be used to transmit status information from the processor to the host PC. This outbound path was used for DSP program development and debugging.

All DSP software development was carried out with an inexpensive shareware TMS320C26 assembler (TDASM). Listings of the host PC control program and of the TMS320C26 DSP program are omitted from this paper, because of their length. They are available from the ARRL BBS (203 666-0578)and via Internet on ftp.cs.buffalo.edu.

DSP Algorithms

The DSP board runs at a relatively high external sampling rate of 24 kHz. This sampling rate allows the use of 6.25-kHz bandwidth Butterworth filters for anti-aliasing and reconstruction. In the DSP software, the input data stream is decimated 2:1, with the help of a 30th-order, 3.0-kHz bandwidth, low-pass FIR filter. The same filter design is used to interpolate 1:2 back to the 24-kHz external sampling rate. Keeping the 1.5-kHz bandwidth output baseband signals well below the output filters' cutoff frequency reduces any phase difference between the two output channels. The current design thus avoids the limitation on sideband suppression that Anderson encountered in his implementation.

The DSP software also implements the 1.5-kHz sine/cosine subcarrier generation, two multipliers, and the two sharp low-pass filters required by the Weaver method. These filters are 114th-order FIR designs with a passband frequency of 1.2 kHz and a stopband frequency of 1.5 kHz.

As shown in Fig 6, the MAX181 includes an input analog multiplexer, which is connected both to the anti-aliasing filter and to the two reconstruction filters. This arrangement allows the DSP board to measure the

Table I— Measured SSB Modulator Performance

Sideband	
Desired	0 dB (Ref. Level)
Opposite	<–40 dB
Intermodulation	
Distortion Products	Not Measured
Carrier and Spurious Products	
Fundamental	–60 dB
Second harmonic	–45 dB
Third harmonic	–57 dB
Other spurious	<–70 dB
Digital feedthrough	≈ –80 dB
Frequency Response	
0.30 - 2.50 kHz	±0.2 dB
0.22 and 2.80 kHz	–3 dB
3.00 kHz	<–55 dB
RF Output Power (50 Ω)	
1-V rms input level	+12 dBm
DAC clipping level	+16 dBm

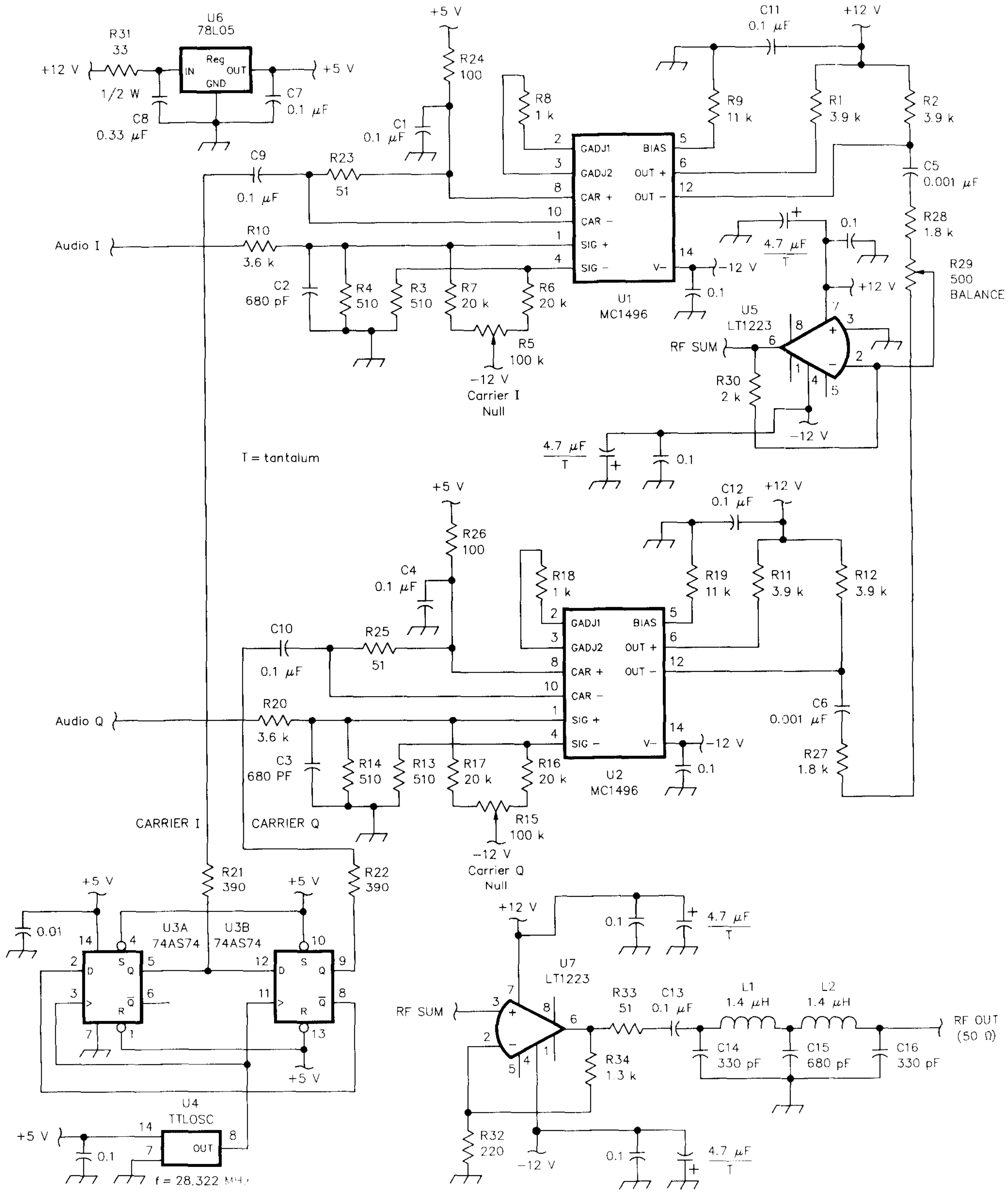

Fig 5—Schematic diagram of the RF board. C14, C15 and C16 are 5% silver-mica capacitors. L1 and L2 are 19 turns of no. 30 enamelled wire on a T-30-6 toroid core.

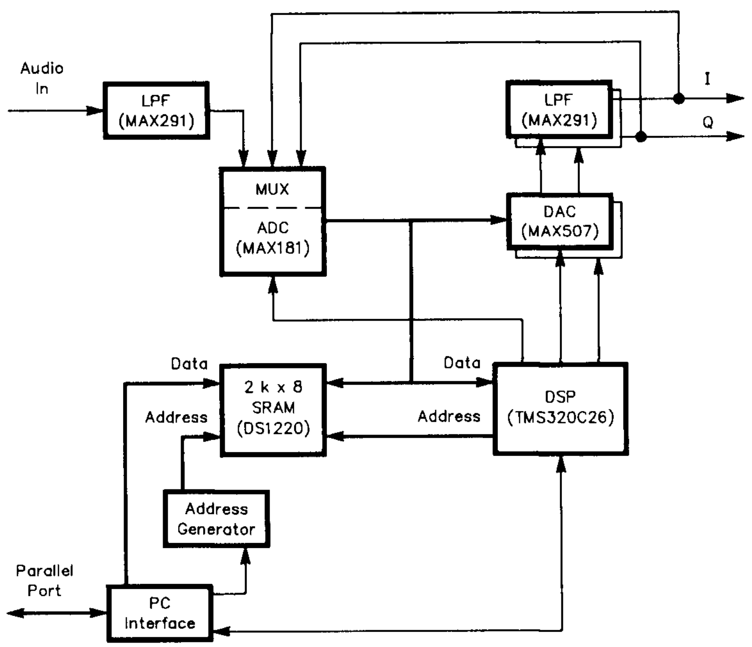

Fig 6—DSP board block diagram.

actual output filter dc offsets, and thus to cancel these offsets by adding or subtracting the appropriate corrections in software. Thus, the current design eliminates completely the need for manual filter dc offset trims.

Measured Performance

Table I summarizes the measured performance of the modulator. All measurements were made with an HP 8590A spectrum analyzer. The desired sideband level is the reference (0 dB) for all other relative power (dB) measurements. The opposite (undesired) sideband was not visible, even when the HP 8590A was set to its narrowest resolution bandwidth of 300 Hz. Thus, the undesired sideband level shown in Table I is an upper limit to the circuit's actual performance. Intermodulation distortion products could not be measured because of the wide skirts of the analyzer's 300-Hz filter.

A broad noise floor arising from digital feedthrough was visible at –80 dB over the range of 0-10.5 MHz, with a few spurs reaching –70 dB. This low-level noise did not change when the RF board's I and Q inputs were grounded, so the noise was due either to the power supply or, more likely, the unfiltered digital carrier generation circuit on the RF board.

Conclusions

The Weaver method SSB modulator complies with the ARRL *Handbook* guidelines (p 18-4) for SSB modulators. The out-of-channel harmonic and spurious signal levels comply with FCC requirements for output power levels as high as 1 kW (*Handbook* p 40-8). The SSB modulator has a performance comparable to that of commercially manufactured units.

Because of the trimpots used on the RF board, this design may exhibit a gradual loss of carrier and sideband suppression with age. With a Weaver method modulator, such aging effects will not generate co-channel interference and will interfere only minimally with the transmitted speech intelligibility (see note 5). The low-level 1.5-kHz tone caused by the partially suppressed carrier is an esthetic drawback that does not affect communications effectiveness.

The use of digital signal processing for the baseband processing allowed the elimination of the two problems Anderson encountered: the phase differential and large dc offsets in the 1.2-kHz bandwidth low-pass filters. Thus, DSP resulted in a superior implementation of the Weaver method.

The DSP approach has two disadvantages: a small amount of digital noise feedthrough, and high complexity, cost, and power consumption. Although the measured digital feed-through was quite small, it could be further reduced by additional filtering. The design's complexity and cost could be justified in most modern transmitters, where the DSP can be used for other functions, such as speech compression and digital modulation.

Notes

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8 Request the schematic of the DSP board by sending an SASE to the ARRL Technical Department. Request the "9/93 QEX Weaver DSP Package." □□

The Growing Family of Federal Standards for HF Radio Automatic Link Establishment (ALE)

Part III: Where are the Federal Standards for HF ALE Radio Networking Going?

With protocols for basic radio linking established, protocols for networking and higher-level functions are next in line.

Larry Brewster, NØRNX and Dennis Bodson, W4PWF

Introduction

Several recent articles have presented an overview of Federal and Military High Frequency (HF) Automatic Link Establishment (ALE) radios to the amateur community.[1,2,3] Other articles describe the standards that must be in place so that government agencies can purchase radios from different vendors and still be assured that they all interoperate.[3] Another recent article describes what may well be the first attempt for amateurs to use HF ALE equipment on amateur frequencies.[4] And a recent series of articles in *QEX* presented you with a view of the "current state and possible future of digital communications in the highfrequency (HP) radio band (3-30 MHz)."[5,6,7,8] To round out the discussion on the present state-of-the-art in HF radio networking, this article presents the Federal HF ALE program prospective. Comparisons will be made between the federal and amateur programs whenever possible.

Amateur Packet Radio

Efforts in the amateur packet-radio community are probably best described in terms of the International Standards Organization (ISO) Open Systems Interconnect (OSI) model.

In general, we look at the communication task and relate it to the OSI model. A representation can be made between the functions of the radio controller and the OSI Reference Model. For the functions associated with the data link layer, almost all amateur packet-radio work is being done with what is commonly known as the AX.25 Data Link Layer Protocol. This protocol has been described in numerous amateur publications over the last few years.

Also described in great detail in numerous amateur publications are the efforts of amateurs to standardize the OSI network and transport layers. An article in a *QST* feature column "Packet Perspective" recently brought us up to date on what is happening in packet networking within the amateur community.[9] This article does an excellent job of summarizing many of the ideas presently being considered in amateur packet radio. The article describes the five most common ideas/attempts at an HF ALE radio solution. Each of these schemes has a set of followers, each attempting to develop the ideal networking solution.

Federal Packet Radio

For many years, the federal community has also been exploring the application of networking technology using packet radio networks. Since 1983, US Government communications groups have been using a packet radio controller to provide data transmission to multiple users on a single radio channel on HF, VHF, and UHF frequencies. This network uses a terminal node controller (TNC), any two-way radio, and the AX.25 protocol.[10]

Not unlike those of the amateur committee, the federal network system requirements are:

1. uncomplicated operator interface and ease of use,
2. rapid access to the network, even

[1]Notes appear on page 19.

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during jamming or poor propagation,
3. continuous operation for all network configurations and propagation conditions,
4. automated routing so that the message gets through even during conditions of very poor connectivity,
5. minimum networking overhead so maximum throughput is obtained,
6. message transmission at all common data rates,
7. connectivity to a wide range of media.

To answer at least the first three of the above requirements, FED-STD-1045 HF ALE radios were developed.[11] FED-STD-1045 radios use a very robust and redundant waveform and have traded data throughput for error control. Items 4 and 5 (ie, advanced networking) are being developed in the continuation standards, proposed FED-STD-1046 and proposed FED-STD-1047. Item 7 is being addressed in proposed FED-STD-1048, which will address multiple-media issues.

Federal HF ALE Radio

To bring you up to speed on the federal effort in HF ALE radio, a recap is necessary. With the recent integration of microprocessor control with state-of-the-art HF radio technology, radios are now being produced that are capable of providing automatic routine and emergency communications while maintaining interoperability between various vendors. Recent federal (see note 11) and military standards are encouraging vendors to produce interoperable (ie, between manufacturers) HF ALE radio systems.[12,13] These radios have features such as selective calling, automatic handshaking, frequency scanning, link quality analysis, and common address group and net calls.

ALE Controller Function Description

A typical ALE radio consists of a HF single-sideband (SSB) transceiver that is connected to a controller to function as an automated HF station. The controller can contain microprocessors or can be an addition to a conventional microcomputer. The present FED-STD-1045 type controller is considered to be a data link-layer controller in terms of the OSI model. The data link controller is roughly equivalent to the amateur TNC. Presently under development are network-layer and transport-layer controllers which will add features of advanced networking to the basic FED-STD-1045 radio. These networking controllers will be standardized by proposed FED-STD-1046 and proposed FED-STD-1047.

ALE radios scan over several channels (frequencies), at two to ten channels per second, looking for traffic. Upon detecting traffic, an ALE radio stops scanning to interrogate for traffic addressed to it; if none is found, it continues scanning. For transmitting, the system has previously determined by Link Quality Analysis (LQA) techniques which channel(s) will do the best job of propagating the message to the distant station. This system assumes that the other members of the network with which you wish to communicate can choose among several frequencies, and the choice may be based on the atmospheric conditions. Through a system of monitoring traffic, sounding, and LQA reporting it has been predetermined which channel(s) in the group will best support a transmission. The transmission is in the form of a link request and contains the address of the originator and the receptor, and can also contain an orderwire message. The actual message can be contained within the orderwire, or it can be indicated to the receptor that a link is requested for voice or modem traffic. After the link-layer controller establishes a link with another station, communication begins. If the message is transferred as part of the link request (as an order-wire mes-

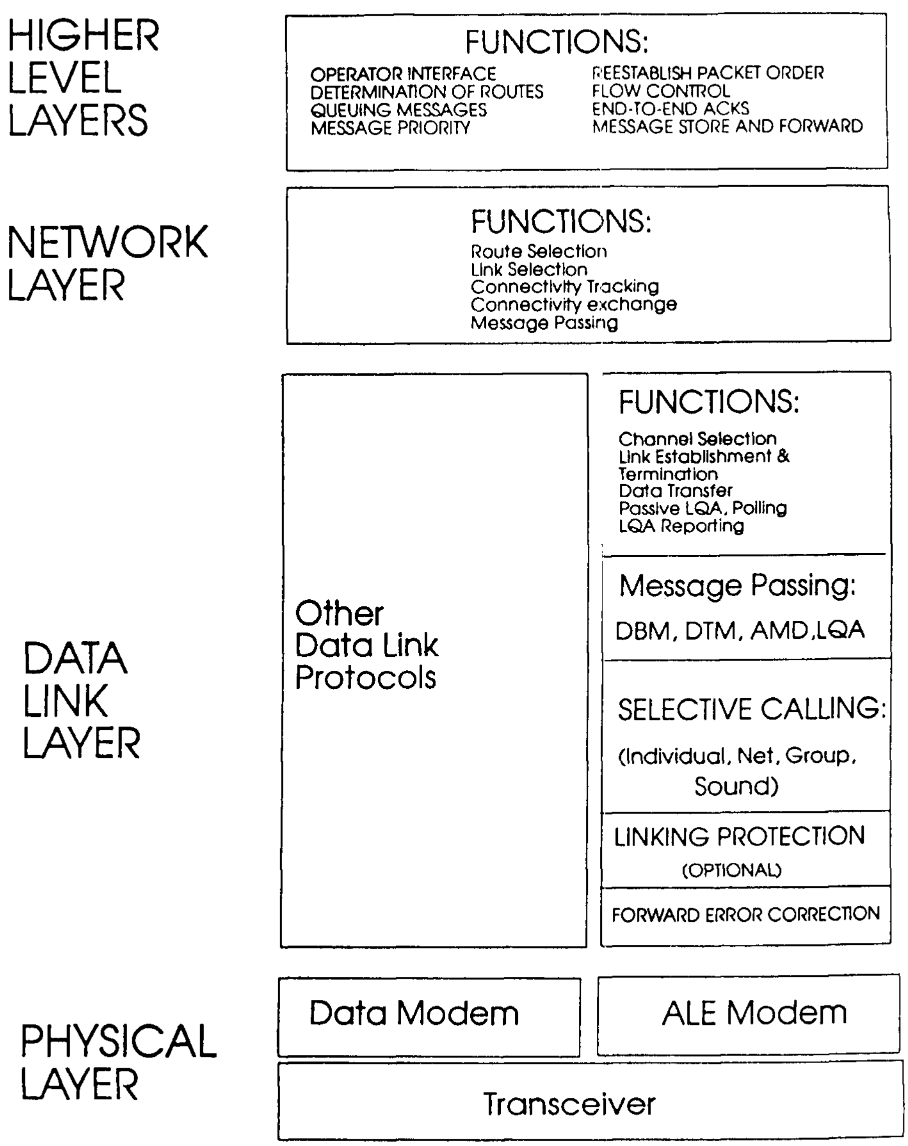

Fig 1—Functional heirarchy of an automated HF station.

sage) or as a data message, it uses a small interval of air-time compared with the time required to send the same message by voice. Therefore, as more and more concepts for networking are developed, the system network throughput will increase.

The principal functions of the station, and the association with the reference model, are schematically shown in Fig 1. When we start adding features such as indirect route selection, connectivity tracking, connectivity (data) exchange (CONEX), automatic message exchange (AME), and automatic message exchange with store-and-forward message exchange (AME w/S&F) we extend into the network and higher-level functions.

Physical Layer Functions

The major function of the controller that relates to the physical layer is that of transceiver and modem (data modem or ALE modem). The data modem used is generally of a MIL-STD-188-11OA type, but other possibilities exist. The ALE modem employs 8-ary frequency shift keying (FSK) with 8-millisecond tones, where 3-bit symbols are sent at a rate of 125 per second, giving a raw data rate of 375 bits per second (BPS).

Data Link Layer (DLL) Function

Referring to Fig 1, we see that at the data-link layer, parallel paths exist where the system/user has the option of using a data modem or ALE modem.

a) *Data-Link Layer/Data Modem*—The data-link layer for the data modem is basically a pass-through function, where functions of the network layer act directly on the physical layer function.
b) *Data-Link Layer/ALE Modem*—The lowest layer of the DLL ALE Modem is the Forward Error Correction (FEC) function. Other sublayers add linking protection (optional), selective calling, an orderwire message section, channel selection, link establishment and termination, LQA collection and reporting, and polling functions.

- *Forward Error Correction (FEC)*—This sublayer adds forward error correction, using a Golay code, applied to the ALE words, and each word is sent three times to allow redundancy and majority voting.
- *Linking Protection (optional)*—Linking protection (LP) is a technique that protects the linking functions from unintentional or malicious interference by scrambling the ALE signaling exchanged among protected stations.[14]
- *Selective Calling*—Calls can be made using individual, net, or group station addressing or sounding. The fundamental address element in the ALE system is the individual station address. A net is a prearranged collection of stations called with a net call. A group is non-prearranged collection of stations where little or nothing is known about them except their individual addresses and scanned channels. Sounding is the ability to empirically test selected channels (and propagation paths) by providing a very brief, beacon-like, identifying broadcast which may be used by other stations to evaluate connectivity, propagation, and availability; and to select known working channels for possible later use for communications or calling.
- *Message Passing*—The link-layer message passing by an HF ALE radio, between communicators, might be in an automatic message display (AMD) message, data text message (DTM), data block message (DBM), or LQA mode. The AMD message appears on an 80-character front-panel display device, the DTM is a standard-speed message mode, and the DBM mode message is high-speed (relative to AMD and DTM) with deep interleaving to penetrate HF channel long fades and large noise bursts. LQA concerns the automatic measurement of the quality of the ALE signal on link(s) between stations. The resultant LQA data is used to score the channels and to support selection of a "best" (or an acceptable) channel for calling and communication.
- *Channel Selection*—The primary function of the link-layer HF ALE controller is to monitor receiver channels, choose the best transmit channel, and link with the called station to carry out some function. Linking functions include linking for voice traffic, linking for the transmission of orderwire, and linking to allow transmission of data by an external modem.
- *Link Establishment and Termination*—The data-link function initiates a required response and acknowledgement to complete a three way handshake.
- *Data Transfer*—The data transfer function transfers data such as passive LQA data. Passive LQA data is data obtained from monitoring normal ALE traffic as well as soundings.
- *Passive LQA*—Evaluation of channel quality by measuring the characteristics of received signals is termed passive LQA because the local radio does not transmit a request for this data. Passive LQA data is obtained by listening to normal ALE traffic and soundings (see note 14).
- *Polling*—The polling data link functions are used to acquire current link quality data by using handshaking and exchanges with one or more stations.
- *LQA Reporting*—LQA reporting is the broadcast of data for purposes of updating the memory in other stations in the network.

Amateur Use of Data Link Layer ALE radios

A recent article in *QST*'s "Packet Perspective" column written by Stan Horzepa, WA1LOU, describes what may well be the first amateur band experiment using FED-STD-1045 ALE equipment. "The experiments, which began in June, 1992, were conducted to see how well ALE performs in the noise and interference typical on the amateur bands." (See note 4.) The article gives a brief synopsis of ALE as it exists today, as well as the results of these amateur band tests.

Federal HF ALE Networking

The federal program for advancing HF ALE radio into networking is presently in the definition stage. Protocols are being written that will extend the basic capabilities of the ALE controller into functions associated with the network and transport layers of the OSI model. As an example of the extension of networking to the functions of the ALE controller, we can look at the protocol necessary for providing star group networking functions for prearranged networks that have direct connectivity. The scanning call signal structure of a star group (one station to many) is illustrated in Fig 2 (see note 12).

The Federal HF ALE networking program is presently defining a series of controllers that extends from a very simple networking controller capable of only slave routing to a full-featured HF controller with the ability to hold messages until connectivity (direct or indirect) to the destination is achieved. In general, these controllers can be diagramed as shown in Fig 3.[15]

The FED-STD-1045 HF ALE radio system is not normally thought of as a packet-radio system. Packets are generally associated with internal packet-network operation and are not necessarily visible to host computers attached to the network. As the functions of the basic ALE controller are extended to higher and higher levels of the OSI model, these functions begin to take on packet-radio qualities. Message passing between layers will be accomplished as shown in Fig 4 where an example shows a message being passed down through various layers with headers added to eventually be broadcast as a packet in a DTM message. The reverse procedure is used with a received message as each layer strips off the header overhead associated with that layer.

The station originating the message is interested in the transmission of the message from the source station to the destination station. Transmission of the message may be a direct single-path transfer, an indirect multisegment transfer occurring in near real-time, or an indirect multisegment transfer requiring an intermediate station to store the message temporarily until a future path is available. Networking concepts are concerned with end-to-end protocols, networking interface, segmenting and reassembly, sequence numbers, creation and deletion of headers, store and forward functions, routing, local control and lockups, network-wide functions and flow control, topology, and network performance.[16]

Networking Layer Control Functions

The functions performed at the network layer may be grouped into two broad categories: routing functions and data management functions. Routing functions select paths through the network for voice and data traffic using link- and path-quality data. Data management functions acquire and communicate that information to a higher level (see note 14). The data that must be maintained is in the form of routing tables and connectivity matrices.

a) Route Selection Function

Routing is an essential element for correct operation of these communication systems. Adaptive routing in particular is required to optimize the use of system resources. When a station cannot be directly linked with a desired destination, other stations may be employed to assist in getting the message through. The simplest option, termed indirect calling, establishes a link with a station other than the desired destination for message passing purposes (see note 14). This indirect calling is the process of sending a message to a station that may have connectivity with the desired station for the purpose of forwarding the message. This may be accomplished either manually (indirect calling) or automatically (relaying),

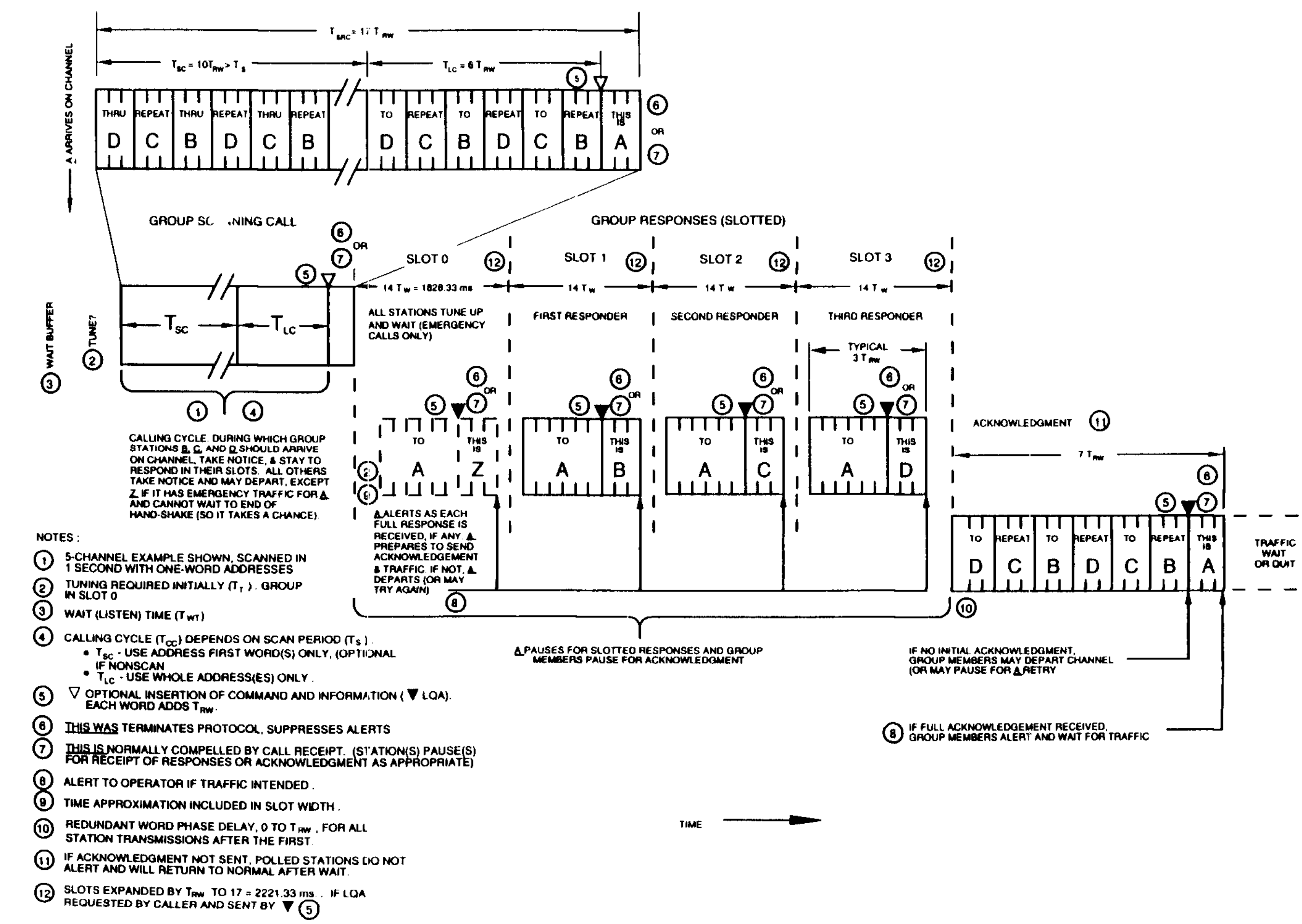

Fig 2—Structure of a star group scanning calling signal.

and may include store-and-forward buffering at intermediate stations.

b) Link Selection Function

The network layer routes the message through the data link controller in the desired manner. One choice may be through a data modem if the message is data and if a data modem is connected. Another choice may be to use the ALE controller and ALE modem. Also, if using the ALE modem, the link selection function must choose to use a sounding, AMD, DTM, or DBM message mode.

c) Connectivity Tracking Function

This function establishes a path quality table to keep track of connectivity. The data stored in a path quality table within the network layer function is obtained from known connectivity, periodic broadcasts, or by request of connectivity exchange (CONEX) data. The data stored in the path quality table is shared with the transport layer function. This layer has responsibility for path determination.

d) Connectivity Exchange (CONEX)

CONEX data can be distributed to other stations on a periodic transmission basis or can be requested/distributed on a demand basis.

e) Message Passing Function

The network layer function must choose to use an orderwire AMD, DTM, or DBM message mode. This may be as an operator real-time message or an Automatic Message Exchange (AME). Since the network function layer does not buffer messages, store-and-forward messages must originate in the higher function layers.

The AME network layer function is defined as the automatic exchange of messages between stations that have direct connectivity with each other. It is the automatic transfer of ALE orderwire data and messages directly from ALE station A to ALE station(s) B (and others), and destined for B (and others), over one of their links with no routing needed.[17]

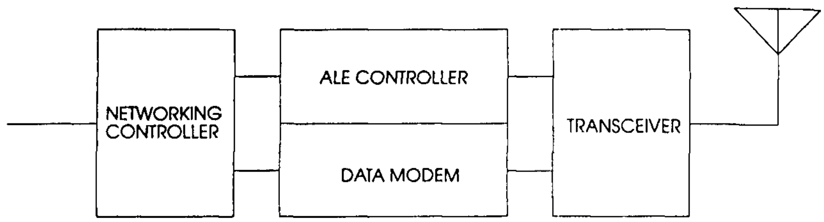

Fig 3—Functional block diagram of an automated station.

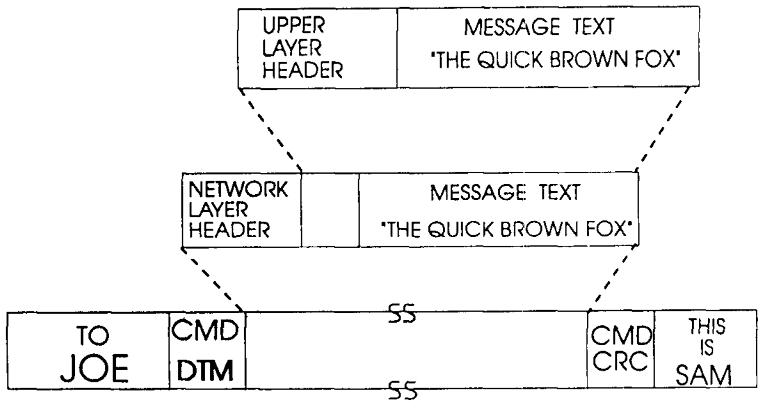

Fig 4—Mapping of upper layers to data link layer DTM.

Transport and Higher-Level Layer Functions

It is the functions of the transport layer and layers above to:

- handshake with the operator
- pass messages down to the network layer
- use the information in the routing and connectivity matrices to determine the direct or indirect route
- instruct the lower layers of the preferred message type (ie, AMD, DTM, or DBM)
- queue messages in case no immediate path(s) is available
- establish message priorities
- re-establish out-of-order packets
- maintain flow control
- keep track of end-to-end acknowledgements, and
- perform the duties of AME (w/S&F) (see below).

Also, there are many other upper-layer functions presently anticipated to be added at a later date.

Automatic Message Exchange with Store-and-Forward (AME w/S&F) is the capability that enables a station to specify a message's routing via indirect relay through another (store-and-forward) station to a destination station.[18] It is the automatic transfer of ALE orderwire data and messages indirectly from ALE station A to ALE station(s) B (and others), through relay(s) C (and others) employing routes and routing as available and based on routing protocols (see note 17). This automatic indirect routing is a transport layer function with decisions based on the use of connectivity exchange information supplied from the network layer function. AME (w/S&F) also includes the capability for a station to automatically accept and store another station's message for direct (or indirect) relay (see note 18).

It is also the function of the higher layers to characterize the system configuration, mission, and topology to meet changing network needs. System configurations affect the routing and networking.

The physical topology is one of the characteristics of the network affected by the network mission. The topology may be a star where the key station might be a national command center. Another choice of topology would be to have a distributed topology where each station has intelligence equal to any other. The location of network stations may change literally minute to minute, as in the case of a system that contains mobile units. It is also possible that the system would only contain fixed-location systems. The network definition tables, output power, and perhaps antenna orientation might be affected by the fixed/mobile nature of the network.

Conclusion

The federal and military communities

have been developing HF ALE radios for many years. There are many fielded radio systems across various government agencies, and the technology is maturing. It is in the area of automatic networking that development work is still being pursued. The architects of these systems are attempting to solve many of the same problems that developers of the Amateur Radio networking effort are experiencing. The functions and features of these ALE radios as well as the interoperability between them has been tested.

Work is ongoing to characterize the performance of HF ALE on actual networks. This work is progressing in network layer simulation studies to determine the effects on message throughput when adding the features of sounding, polling, and networking features.[19]

The federal program for advancing HF ALE radio can be thought of as consisting of a series of hierarchical building blocks for interoperability and operation (see notes 1, 2 and 3). The basic FED-STD-1045 radios are used as building blocks with advanced features such as polling, linking protection, connectivity exchange, automatic message exchange, message store and forward, network coordination, and networking to multiple media added as needed.

Many parallels can be drawn between the effort of federal HF ALE developers and developers of the amateur packet program. These two development efforts are presently struggling with network and transport layer functions while trying to maintain interoperability among the users. The development of these programs using the concepts of OSI has gone a long way in encouraging interoperability. While many of the system constraints are slightly different for the two efforts, in general, they have a lot in common. By diagnosing the efforts of each of these developments, benefits will be received by both.

Many proponents are suggesting a form of ALE for the Amateur Radio community. Work is presently going on to define the networking requirements.

Acknowledgments

This work was supported by the National Communications System (NCS) and the National Telecommunications and Information Administration/Institute for Telecommunication Sciences (NTIA/ITS).

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A 12 GHz, 1/8 Prescaler

Need to count microwave frequencies? Here's a prescaler IC that will work up through the 3-cm band.

Angel Vilaseca, HB9SLV and Serge Rivière, F1JSR

This article describes two applications—a frequency counter and a phase-locked loop—that make use of the Fujitsu FMM110HG, a 0.6-to-12-GHz prescaler IC that divides the input frequency by 8.

Fig 1 shows the circuit used with the IC, and Fig 2 gives an etching pattern used to build the prescaler on a PC board.

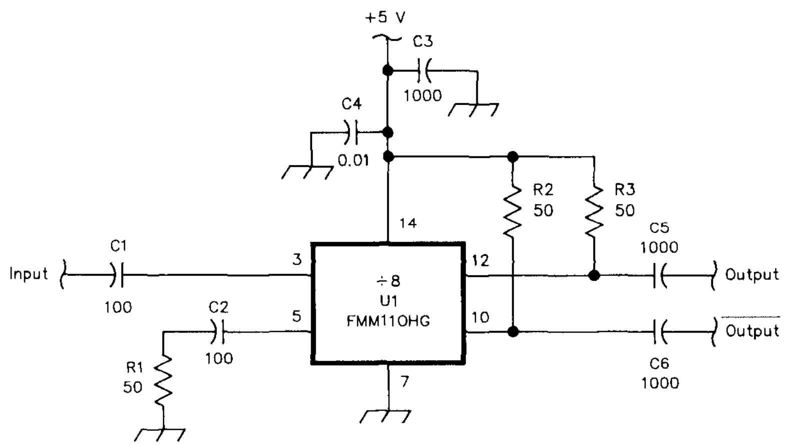


Fig 1—Schematic diagram of the prescaler circuit.

C1,C2—100-pF, single-layer (if possible) SMD capacitors.
C3,C5,C6—1000-pF, single-layer (if possible) SMD capacitors.
C4—0.01-µF SMD capacitor.
R1,R2,R3—50-Ω chip resistors.
U1—Fujitsu FMM110HG prescaler IC.

Angel Vilaseca, HB9SLV
8, Place du Rondeau
1227 Carouge
Switzerland

Serge Rivière, F1JSR
"Chez Viollet"
Draillant
74550 Perrignier
France

The circuit is based on a Fujitsu FMM110HG. It is a monolithic GaAs integrated circuit in a surface-mount package.

The IC's main features, taken from the data sheet, are:

Input frequency:	0.6 to 12 GHz
Input level:	0 dBm to +10 dBm
Output level:	+4 dBm typical
Maximum input level:	+13 dBm
Power supply:	+5 V at 120 mA

The PCB is made of normal epoxy (yes!) and was not particularly designed to have 50-Ω input microstrip conductors. This would be rather pointless because the IC's input impedance is far from constant over a broad frequency range. The epoxy PCB was not found to be especially lossy even at 10 GHz. Indeed, operation was flawless with less than one milliwatt at the input, well within the IC's specification.

Mylar copies of the PC pattern are available from F1JSR.

The 50-Ω resistors at the input and outputs, are "garden variety" and are not critical as long as they are small SMDs to fit the board.

The input capacitors are a little more critical. We recommend you use single-layer capacitors, well suited for SHF. But they are not all *that* critical. In the first experiments, while waiting for delivery of the 100-pF single-layer caps, a leadless disc-ceramic 470-pF

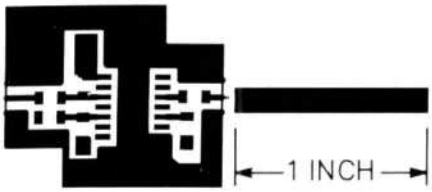


Fig 2—Printed circuit etching pattern for the prescaler circuit. The components are surface-mounted on the circuit side of the board, with the other side of the board left unetched. Use epoxy board material.

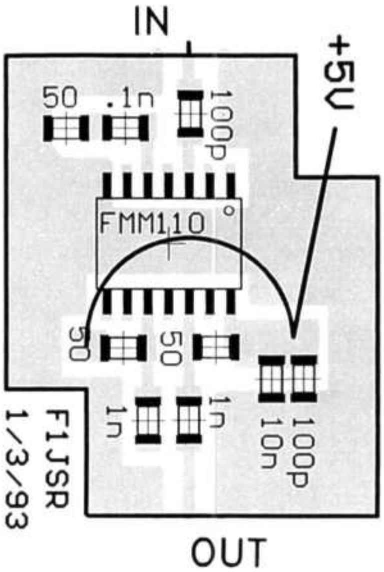


Fig 3—Component placement diagram for the prescaler.

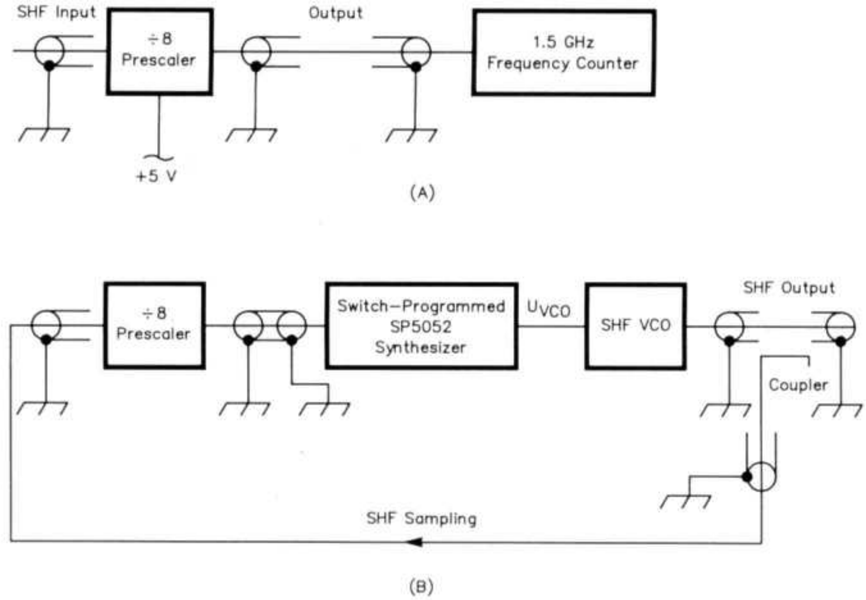


Fig 4—Block diagrams of the prescaler applications. At A, use with a 1.5-GHz frequency counter. B shows an SHF synthesizer using the prescaler.

capacitor was split in four with a pair of pliers, thus making four 100-pF capacitors for the price of one...and it worked! The SMD parts must be soldered as close to the IC as possible.

The bottom of the IC is soldered to ground, mainly to improve heatsinking. This was done as follows: Turn the IC upside down. Melt two small dots of solder on the IC's gold-plated belly. Pre-tin the ground pad on the PCB's component side where the IC will be located. Carefully position the IC on the PCB, making positively sure it is the right way around! With each pin on the right pad (SMD components are *small*), heat the grounded pad until the solder melts and neatly grounds the IC's body. Do not use too much solder. Finally solder the IC's pins to their tracks.

The PCB is mounted in a small Proteam box, which is sold already fitted with two connectors and a bypassed power-supply input. The PCB is soldered on both sides to the inside of the box to make a good ground contact. (This kind of box may be quite hard to find.) Although we didn't try it, a normal tin-plated box would probably be good enough.

Using the Prescaler with a Frequency Counter

The displayed frequency read from an unmodified counter must be multiplied by 8 to obtain the real measurement. To obtain a direct display of the actual frequency being measured, some counters could be preset when used with an external prescaler. Such a preset is generally a power of two to allow for the most popular prescaler modules.

Another approach would be to add a divide-by-125 divider chain after the prescaler so that the total division would be 1000. In that case, the frequency counter need only be a 12-MHz model, and the displayed frequency can be read directly in GHz instead of MHz. Of course, the divide-by-125 chain would have to operate at ⅛ the highest measured frequency.

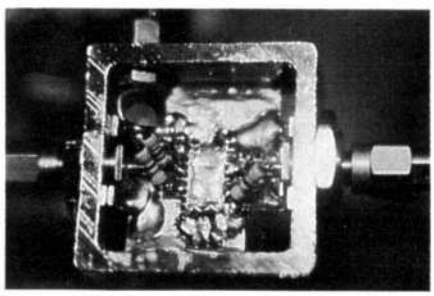
A third possibility would be to modify the counter's timebase to increase the count enable period to 8 times its normal value. But this could result in a very slow frequency display update.

Using the Prescaler in a Synthesizer

Two modules are needed in this case: a synthesizer with an input frequency of 1.5 GHz or more and a VCO suited for the frequency band of interest.

We successfully used an SP5052 from Plessey up to 2.5 GHz. Such a synthesizer has been described in several publications lately.

A Gunn oscillator fitted with a Varicap was used as a VCO, as was a DRO designed by F6IWF and published in *VHF Communications* in February 1992. It should also be possible to use a VCO on a lower frequency, followed by frequency multiplying stages.



Directly sampling the SHF frequency can help keep synthesizer steps small.

Some tests were made between F1JSR and HB9AFO across Geneva Lake on a 30-km path, using both NBFM and FM ATV. The results were very interesting, particularly with the Gunn oscillator. The notorious drift of this kind of oscillator was so much reduced as to make an NBFM QSO possible! We didn't try SSB, though.

The prescaler has two outputs, which may come in handy in the SHF synthesizer application. The first output goes to the SP5052 synthesizer, the second to a frequency counter for control purposes.

Operation

As with most prescalers, without an input signal the IC may oscillate at about 650 MHz. The oscillation stops when a signal is fed to the input.

It might be useful to adjust the power-supply voltage between 4.5 and 5.5 volts to optimize operation. This is the only tweaking needed.

Don't worry if the 50-Ω output resistors and the IC become a bit hot; this is entirely normal.

And we saved the worst for last. The price for the whole unit is about 1700 French francs, or about $300. But that makes it still much cheaper than off-the-shelf professional 12-GHz counters. □□

RF

Zack Lau, KH6CP/1

Central States VHF Conference Comments

Central States was a great conference, as usual, though the weather was far too warm for the antenna measuring session. (I think the construction workers on the roof quit before we did, though!) A rough prototype of the log feed idea I mentioned in a previous column did quite well, even with a small dish with a usable diameter of only 22.5 inches. The feed was much closer to the dish than I had expected. As a result, I erroneously concluded that the focal point was 9, rather than 11 inches from the dish, so the scalar feeds for 10 and 6 GHz I brought weren't even close to the proper focus, even though they were varied a little. The 6-GHz feed appeared to want to be even closer to the dish! Careful testing with the sun (don't burn yourself or your feed!) got the feed close to the focal point for 24-GHz testing. But by then it was hot and late enough to just measure once

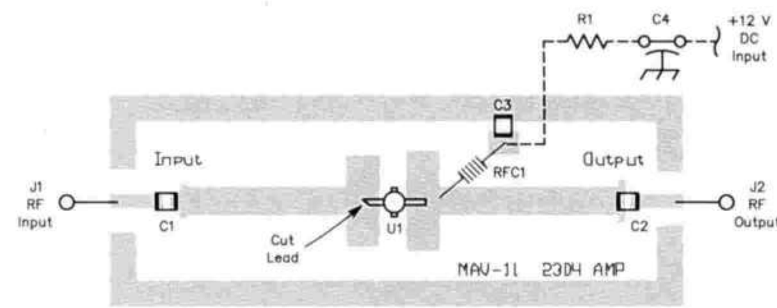


Fig 1—Component placement diagram for the 13-cm amplifier.

C1, C2—10-pF chip capacitors. Cheap NPO types used.
C3—1000-pF chip capacitor
C4—100-pF to 0.01 µF feedthrough capacitor soldered to brass wall.
J1,J2—Female SMA connectors.
R1—120-ohm, ½ watt carbon film resistor.
RFC1—6 turns #24 enam wire close wound. Use a 3/16-inch drill bit to form the turns. (Parameters aren't critical—this was pulled out of the scrap bin.)
U1—MAV-11 or MSA-1104 MMIC.

Table 1—2304 MHz power output test

P_{in} (dBm)	P_{out} (dBm)
0.0	7.17
5.0	12.17
6.0	13.17
7.0	13.83
8.0	14.83
9.0	15.67
10.0	16.33
11.0	17.00
12.0	17.67
13.0	18.17

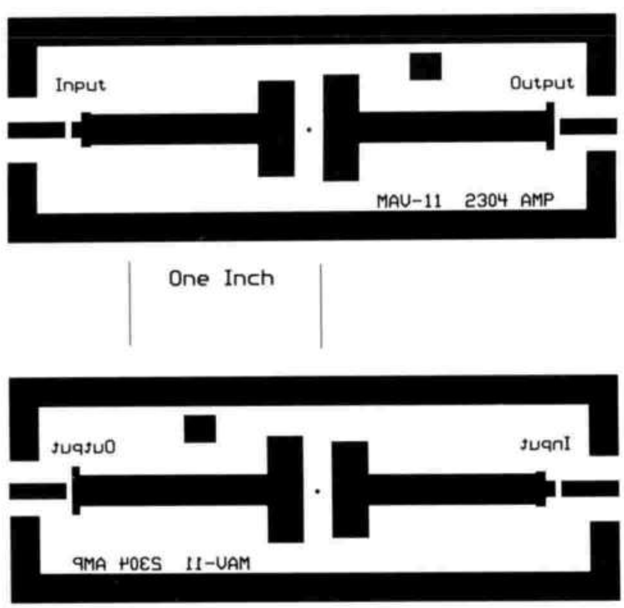


Fig 2—Etching pattern for the 13-cm amplifier.

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Newington, CT 06111
email: zlau@arrl.org (Internet)

```
* MSA1104 at 2304 MHz
* Tune input and output for
* good return loss/SWR
* file \mh\1104amp.ckt

W50:85MIL
WIS:?50.8866MIL?
PIS:?49.3106MIL?
WIN:?161.11MIL?
PIN:?872.762MIL?
WIS2:?190.974MIL?
PIS2:?170.914MIL?
WOS1:?187.988MIL?
POS1:?197.671MIL?
WOS2:?41.7189MIL?
POS2:?46.5488MIL?
WOL:?162.82MIL?
POL:?980.178MIL?

blk

trl 1 2 w=w50 p=300mil sub1
slc 2 3 l=.3nh c=10pf q=15 f=2.3ghz
trl 3 4 w=w50 p=50mil sub1
cross 4 5 6 7 w1=w50 w2=wis w3=win w4=wis sub1
trl   6 8     w=win p=pin sub1
cross 8 9 10 11 w1=win w2=wis2 w3=30mil w4=wis2 sub1
ost   5    w=wis p=pis sub1
ost   7    w=wis p=pis sub1

ost  9    w=wis2 p=pis2 sub1 ost 11    w=wis2 p=pis2 sub1

two 10 20 110 MSA11
wrap 110 0 a:30mil sub1
wrap 110 0 a:30mil sub1


cross 20 25 30 35  w1=30mil w2=wos1 w3=wol w4=wos1 sub1

ost   25  w=wos1 p=pos1 sub1

ost   35  w=wos1 p=pos1 sub1

trl  30 40  w=wol p=pol sub1

cross 40 45 50 55 w1=wol w2=wos2 w3=w50 w4=wos2 sub1

ost 45 w=wos2 p=pos2 sub1
ost 55 w=wos2 p=pos2 sub1

slc 50 65 l=0.3nh c=10pf q=15 f=2.3ghz

trl   65 70 w=w50 p=300mil sub1

amp: 2por 1 70

end
freq
2.0ghz 2.3ghz 2.5ghz
end
opt
amp ms11 .1 lt
amp ms22 .1 lt
end

data
msa11:s
2ghz .37 149 2.39 52 .224 -5 .43 140
2.5ghz .45 133 2.02 33 .231 -10 .47 125
sub1:ms h=31mil er=2.55 met1=cu 1.4mil rgh=100uin tand=0.002
*1/32 inch teflon board
end
```

Fig 3—Microwave Harmonica source file.

Table 2—Frequency response test

f (MHz)	P_{in} (dBm)	P_{out} (dBm)	*Gain* (dB)
432	–0.33	8.17	8.50
903	–0.17	6.17	6.34
1152	–0.17	6.17	6.34
1300	0.17	7.33	7.50
2160	–0.33	7.17	7.34
2300	–0.67	7.00	7.67
2700	–0.67	2.67	3.34

and pack up. The loss in gain due to the moving phase center is under 2 dB with this dish, despite the feed covering 903 to 5760 MHz. A bigger dish would have made this even less, but might have overly complicated airline travel!

I made a carrying case out of 1-inch-square aluminum tubing, plywood, a clear plastic sheet and cardboard which did an excellent job of protecting the dish. Why a plastic sheet? Since the side panels were attached with 8-32 cap head screws, I thought it might be a good idea to be able to show airport security what was inside the box without spending 10 minutes taking the screws out. Besides, the next step involves making the box into a fully functional tripod with a compass rose. The plastic should protect the compass rose from damage and wet weather. A cast aluminum handle was screwed to the aluminum frame, and cardboard was wrapped around the box for added protection. I cut a hole in the cardboard to let the handle stick out.

Perhaps after the VHF/microwave contest and conference season I'll come up with a better log feed design that doesn't have such a high SWR on 13 cm. That should noticeably improve the measured gain. I'll be at the Microwave Update in Atlanta, September 24 and 25th. Hopefully I'll have recovered by then from my 6m-to-24-GHz effort on Mt Equinox in the September VHF contest (FN33)—and from the 10-GHz contest.

The noise figure contest at Central States was also quite popular, with many 2-meter and 432-MHz preamps measured. There was also a second session for the fanatics, who got to puzzle over comparisons using three different noise sources with "calibrated" preamps. We were hoping to do a further comparision with a hot/cold noise source, but despite Tommy's (WD5AGO), best efforts we didn't get to. The contest went quite quickly, since there weren't

```
MICROWAVE HARMONICA PC V5.5   File: \mh\1104amp.ckt    31-JUL-93
23:03:04

 Freq      MS11     MS21     MS22      K
 GHz        dB       dB       dB

           AMP      AMP      AMP      AMP

 2.000   -18.932    8.46   -21.660    1.05
 2.300   -24.963    7.81   -19.926    1.07
 2.500   -20.156    7.34   -25.525    1.08
```

Fig 4—Microwave Harmonica output.

any battles over the last meaningless 0.01 dB. For instance, how does one compare a Fujitsu FHX 16 with a waveguide input on 10 GHz with a SMA connectored NEC NE32684A? Al's (WB5LUA) solution was just to make the waveguide-to-SMA-transition loss 0.11 dB, the difference in noise figures according to the readout of the Hewlett Packard 8970.

A 50-mW, 13-cm amplifier using the MAV-11/MSA-1104 MMIC

While it is possible to get 25 mW from an MAR-8 MMIC amplifier on this band, many people don't like this device due to stability problems, though it is possible to put an MAR-7 in front of it and stabilize it without losing output power. The 1104 MMIC is rated at 17.5 dBm for 1-dB compression at 0.5 GHz, but is under 16 dBm at 2 GHz on a rapidly falling curve (which looks like it would be around 14 dBm with extrapolation to 2.3 GHz). It is obvious that there is a pretty bad mismatch at both the input and output ports if you look at the S parameters. Thus, I expected to restore much of its performance just by tuning the input and output lines. As Table 1 indicates, this is indeed the case.

Construction

I made an open box by soldering 0.025-inch-thick brass walls to the top and bottom ground foils. I used SMA connectors on the input and output to get a repeatable interface. These are attached to the 1/2-inch-high brass walls with 2-56 screws. Very short leads with Teflon coax will work on 2304, but the results aren't as consistent. The ground leads of the MMICs are bent against the body of the device and soldered to the ground plane. The prototype didn't have a pad for the RF choke—it just went to a chip capacitor soldered to the top ground foil. The bias resistor is shielded by the ground plane. It is hooked up to the choke though a small hole with the ground foil cleared away. The board material is 1/32-inch Teflon with a dielectric constant of 2.55. □□

Upcoming Technical Conferences

1993 ARRL Conference on Digital Communications

• September 10-11, 1993, Holiday Inn Airport in Tampa, Florida

• Contact: Brian Lantz, KO4KS, TPALAN, 6403 N Paddock Avenue, Tampa, FL 33614, Tel 813-877-1469, Internet brianlantz@delphi.com.

• Events: Friday, registration and round-table discussions; Saturday, registration, technical talks, discussions on "standards."

• Misc: several "distractions" are available on site and within a short drive.

• Hotel reservations: call the hotel direct at 813-879-4800 and mention the conference. Special rate of $52 per night is available from 3 days before through 3 days after.

Microwave Update '93

• September 23-26, 1993, Northwest Atlanta Hilton, Atlanta, Georgia

• Contact: Jim Davey, WA8NLC, 4664 Jefferson Township Place, Marietta, GA 30066, tel [W] 404 333-2136 [H] 404-998-6971.

• Events: Thursday evening, registration and socializing; Friday, technical sessions, flea market and noise figure contest; Saturday, technical sessions and banquet; Sunday, a short meeting is planned.

• Hotel reservations: call Kim Gilliam (at the Hilton) at 800-234-9304. An airport shuttle bus is available at a round-trip cost of $25. Call 800-237-0709 for reservations.

Mid Atlantic VHF Conference

• October 2, 1993, Days Inn, Horsham, Pennsylvania

• Contact: John Sorter, KB3XG, PO Box 451, Montgomeryville, PA 18936, tel 215 270-3185.

• Events: The conference will feature diverse and challenging talks in the VHF and microwave areas.

• Hotel reservations: call the Days Inn at 215 674-2500, or fax 215 674-0145, and mention "The Packrats" for special room rates. *Otto's*, a German/American restaurant, is located across the parking lot from the Days Inn; meals run between $5-10.

AMSAT-NA 1993 Annual Meeting and Space Symposium

• October 7-10, La Quinta Inn, Arlington, Texas

• Contact: AMSAT, 850 Sligo Avenue #600, Silver Spring, MD 20910, tel 301 589-6062, fax 301 604-3410.

• Events: Thursday 9 AM, leave from the La Quinta with an expert guide and visit various electronic surplus stores in the Fort Worth/Dallas area; Friday morning, an antenna test range will be set up near the hotel—bring your antenna design and test it out; Friday afternoon, presentation of AMSAT technical papers begins and/or attend the *ARRL Amateur Radio Educational Workshop*, hosted by Rosalie White, WA1STO, ARRL Educational Activities Department Manager. Papers of interest to educators and amateurs will be featured; Friday evening, gather with friends and sample the local Texas BBQ and Tex-Mex cuisine, or go next door to the stadium and see the Texas Rangers in action; Saturday, technical talks, topics include Phase 3D, Pacsats, DSP modems and more; Saturday evening, a traditional "attitude adjustment" followed by a banquet dinner. As usual, a keynote speaker will be present to impressively entertain you; Sunday, a series of technical session on various topics is planned, including the traditional *Beginners Forum*. The open board of Director's meeting will be held in the afternoon.

• Hotel reservations: call the La Quinta Inn at 800 531-5900. Make reservations before September 24 and receive the special AMSAT rate of $50 for a single and $55 for a double room. Shuttle bus service is available between Dallas/Fort Worth Airport and the La Quinta Inn.

Radio Amateurs of/du Canada—First National Convention

• July 29-31, 1994, Marlborough Inn Convention Centre, Calgary, Alberta, Canada.

• Contact: Gerry Shand, VE6BLI, 55 - 51551 Range Road 212A, Sherwood Park, AB, T8B 1B2 Canada, tel 403 922-2099, fax 403 438-4398.

• **Call for Papers:** Papers concerning both the technical an practical aspects of ham radio are welcome in regards to HF, VHF, UHF communication methods and techniques, packet, AMTOR, RTTY, AMSAT and EME. These papers will center on a 30-45 minute presentation with a 15 minute question and answer period from the audience. Papers will be printed in the *Technical Proceedings of the First National Convention of RAC.* Submission schedule: (1) author's name, paper's title, introduction and abstract are due no later than October 15, 1993 (2) first editorial draft of paper will be due no later than march 1, 1994 (3) final copy for printing to be due no later than May 30, 1994 (4) presentation dates and times will be confirmed later.

• Events: Special event station CI6RAC; exhibitions, dealer displays, seminars, ladies' program, children's program, RAC general meeting, QCWA meeting, QSL bureau, banquet with keynote speaker. Contact Gerry Shand, VE6BLI, for more information.

(*Have an upcoming technical event? Drop us a note with the all the details and we'll include it in* Upcoming Technical Conferences.) □□