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Chapter 13 —
CD-ROM Content

Supplemental Articles and Projects

- "Designing and Building Transistor Linear Power Amplifiers" Parts 1 and 2 by Rick Campbell, KK7B
- "A Fast TR Switch" by Jack Kuecken, KE2QJ
- "A Homebrew High Performance HF Transceiver — the HBR-2000" by Markus Hansen, VE7CA
- "The MicroT2 — A Compact Single-Band SSB Transmitter" by Rick Campbell, KK7B
- "The MkII — An Updated Universal QRP Transmitter" by Wes Hayward, W7ZOI
- "The Norcal Sierra: An 80-15 M CW Transceiver" by Wayne Burdick, N6KR (plus supporting files)
- "The Rockmite — A Simple Single-Band CW Transceiver" by Dave Benson, K1SWL (plus supporting files)
- "The TAK-40 SSB/CW Transceiver" by Jim Veatch, WA2EUJ
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- "The Tuna Tin 2 Today" by Ed Hare, W1RFI
- "VHF Open Sources" by Rick Campbell, KK7B (plus parts placement guides)

Chapter 13

Transmitters and Transceivers

Transmitters are the companion to the receivers discussed in the previous chapter. As with receiver design, the basic elements of transmitters such as oscillators and modulators are described in other chapters of this book. Transceivers—the combination of a transmitter and receiver in the same package — add switching and signal control circuitry.

Amplifiers for power levels above 100 W (at HF) are covered in the **RF Power Amplifiers** chapter. The **DSP and Software Radio Design** chapter has more information on digital techniques and architectures.

This chapter includes a trio of QRP transmitter projects, two transceiver projects, and supporting information and articles that can be found on the *Handbook* CD-ROM. Rick Campbell, KK7B, contributed a new section on design and construction of VHF signal sources.

Current Transceiver Overview

A supplemental article on this book's CD-ROM describes a range of commercial HF and VHF/UHF transceivers. With each subsequent edition, the overview is updated and the previous version moved to the ARRL website for future reference.

Transmitter technology has advanced in a parallel process similar to that of the technology of receivers. While transmitters are composed of many of the same named blocks as those used in receivers, it's important to keep in mind that there are significant differences. An RF amplifier in a receiver may deal with amplifying picowatts while one in a transmitter may output up to kilowatts. While the circuits may even look similar, the size of the components, especially cooling systems and power supplies, may differ significantly in scale. Still, many of the same principles apply.

Transceivers — the combination of a receiver and transmitter in a single physical piece of equipment — are the norm in Amateur Radio today. Separate receivers and transmitters are no longer offered by the major manufacturers, although many amateurs find this an easier approach when constructing homebuilt equipment. With receivers covered in their own chapter, this edition of the book combines the previously separate chapters on the closely related technology of transmitters and transceivers.

Transceivers achieve many economies by sharing receiver and transmitter elements such as high-performance components and circuits, power supplies and antenna switching circuits, as well as the physical enclosures and operating controls themselves. The sharing is facilitated by control and switching circuitry as discussed in the transceiver sections of this chapter.

Transmitters (and transceivers) may contain hazardous voltages, and at higher power levels RF exposure issues must be considered — review the **Safety** chapter for more information. Techniques for transmitter measurement are covered in the **Test Equipment and Measurements** chapter.

13.1 Transmitter Modulation Types and Methods

13.1.1 Data, Morse Code or Pulse Transmitters

The simplest transmitter consists of an oscillator generating a signal at the frequency we want to transmit. If the oscillator is connected to an antenna, the signal will propagate outward and be picked up by any receivers within range. Such a transmitter will carry little information, except perhaps for its location — it could serve as a rudimentary beacon for direction finding or radiolocation, although real beacons generally transmit identification data. It also indicates whether or not it is turned on, perhaps useful as part of an alarm system.

To actually transmit information, we must *modulate* the transmitter. The modulation process, covered in detail in the **Modulation** chapter, involves changing one or more of the signal parameters to apply the information content. This must be done in such a way that the information can be extracted at the receiver. As previously noted, the parameters available for modulation are:

Frequency — this is the number of cycles the signal makes per second.

Amplitude — although the amplitude, or strength, of a sinusoid is constantly changing with time, we can express the amplitude by the maximum value that it reaches.

Phase — the phase of a sinusoid is a measure of when a sinusoid starts compared to another sinusoid of the same frequency.

We could use any of the above parameters to modulate a simple transmitter with pulse type information, but the easiest to visualize is probably amplitude modulation. If we were to just turn the transmitter on and off, with it on for binary "one" and off for a "zero," we could surely send Morse code or other types of pulse-coded data. This type of modulation is called "On-Off-Keying" or "OOK" for short.

Some care is needed in how we implement such a function. Note that if we performed the obvious step of just removing and turning on the power supply, we might be surprised to find that it takes too much time for the voltage to rise sufficiently at the oscillator to actually turn it on at the time we make the connection. Similarly, we might be surprised to find that when we turn off the power we would still be transmitting for some time after the switch is turned. These finite intervals are referred to as *rise* and *fall times* and generally depend on the time constants of filter and switching circuits in the transmitter.

REAL WORLD CW KEYING

In the **Modulation** chapter, the importance of shaping the time envelope of the keying pulse of an on-off keyed transmitter is discussed. There are serious ramifications of not paying close attention to this design parameter. The optimum shape of a transmitter envelope should approach the form of a sinusoid raised to a power with a tradeoff between occupied bandwidth and overlap between the successive pulses. This can be accomplished either through filtering of the pulse waveform before modulation in a linear transmitter, or through direct generation of the pulse shape using DSP.

The differences between well-designed and poor pulse shaping can perhaps be best described by looking at some results. The following figures are from recent *QST* product reviews of commercial multimode 100 W HF transceivers. **Fig 13.1** shows the CW keying waveform of a transmitter with good spectrum control. The top trace is the key closure, with the start of the first contact closure on the left edge at 60 WPM using full break-in. Below it is the nicely rounded RF envelope. **Fig 13.2** shows the resultant signal spectrum. Note that the signal amplitude is about 80 dB down at a spacing of ±1 kHz, with a floor of –90 dB over the 10 kHz shown. **Figs 13.3** and **13.4** are similar data taken from a different manufacturer's transceiver. Note the sharp corners of the RF envelope, as well as the time it takes for the first "dit" to be developed. The resulting spectrum is not even down 40 dB at ±1 kHz and shows a floor that doesn't quite make –60 dB over the 10 kHz range. It's easy to see the problems that the latter transmitter will cause to receivers trying to listen to a weak signal near its operating frequency. The unwanted components of the signal are heard on adjacent channels as sharp clicks when the signal is turned on and off, called *key clicks*. Note that even the best-shaped keying waveform in a linear transmitter will become sharp with a wide spectrum if it is used to drive a stage such as an external power amplifier beyond its linear range. This generally results in clipping or limiting with subsequent removal of the rounded corners on the envelope. Trying to get the last few dB of power out of a transmitter can often result in this sort of unintended signal impairment.

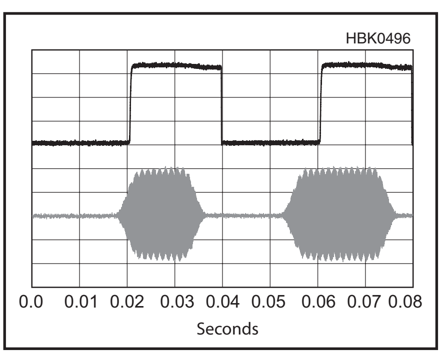


Fig 13.1 — The CW keying waveform of a transmitter with good spectrum control. The top trace is the key closure, with the start of the first contact closure on the left edge at 60 WPM using full break-in. Below that is the nicely rounded RF envelope.

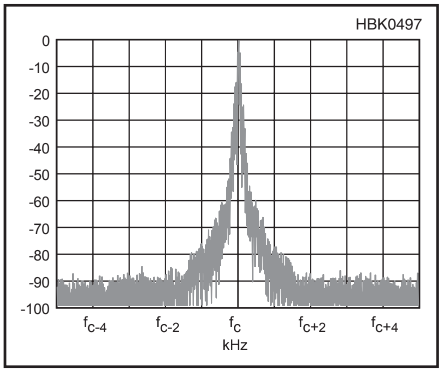


Fig 13.2 — The resultant signal spectrum from the keying shown in Fig 13.1. Note that the signal amplitude is about 80 dB down at a spacing of ±1 kHz, with a floor of –90 dB over the 10 kHz shown.

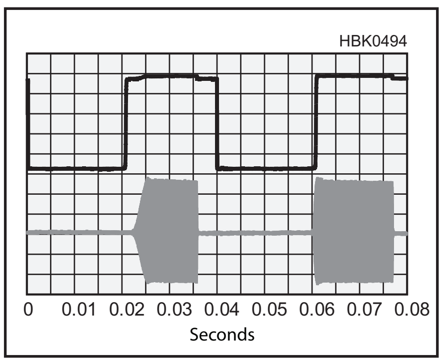


Fig 13.3 — The CW keying waveform of a transmitter with poor spectrum control. The top trace is the key closure, with the start of the first contact closure on the left edge at 60 WPM using full break-in. Note the sharp corners of the RF envelope that result in excessive bandwidth products.

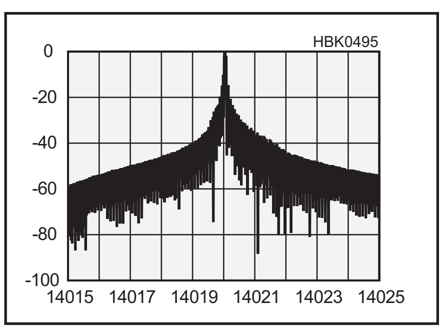


Fig 13.4 — The resultant signal spectrum from the keying shown in Fig 13.3. The resulting spectrum is not even down 40 dB at ±1 kHz and shows a floor that doesn't quite make 60 dB below the carrier across the 10 kHz.

13.1.2 *Project:* Simple CW Transmitters

The schematic for a very basic low power HF CW oscillator transmitter is shown in **Fig 13.5.** Using a crystal-controlled oscillator gains frequency accuracy and stability but gives up frequency agility.

In this design, the key essentially just turns power on and off. The value of the keying line 0.1 μF bypass capacitor was chosen so it would not create an excessive rise or fall time at reasonable keying speeds. This transmitter will generate a few mW of RF power at the crystal frequency, but it has a number of limitations in common with early vacuum tube transmitters that were such an improvement over the spark transmitters that preceded them.

First, the oscillator is dependent on the environment. Changes in the antenna from wind, for example, change the load and can cause the frequency to shift. Second, the oscillator generates signals at harmonics of the fundamental frequency. Third, every time the key is closed the oscillator must start up, taking a short but audible time for the output frequency to stabilize. This creates a distinctive change in output frequency referred to as "chirp," since the signal sounds like a bird's chirp in the receiver.

By adding an amplifier stage to the oscillator transmitter, the oscillator can be isolated from changes in the environment to improve

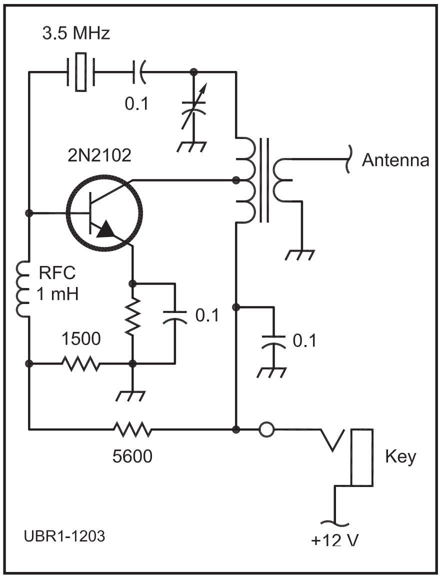


Fig 13.5 — A simple solid-state low-power HF crystal-controlled oscillator-transmitter. The unspecified tuned circuit values are resonant at the crystal frequency.

stability. Adding filtering at the output addresses the problem of harmonics.

The following project by Rev George Dobbs, G3RJV, was published in the Spring 2012 issue of *QRP Quarterly* and is reprinted courtesy of the QRP Amateur Radio Club (**www.qrparci.org**) and *Practical Wireless* (**www.pwpublishing.ltd.uk**). An article by Ed Hare, W1RFI, describing the Tuna-Tin 2 transmitter that was included in previous editions of the *ARRL Handbook* is available on this book's CD-ROM and listed in the References section of this chapter.

Fig 13.6 is the schematic of a much better, but still simple, CW transmitter. It is an update to the "Pebble Crusher" originally designed by Doug DeMaw, W1FB (SK), and described in his *QRP Notebook* (out of print). The transmitter uses an oscillator followed by a stage of amplification to about 1/2 W of output. Note that the output circuit includes a filter to reduce any harmonic output below the levels required by FCC rules.

The circuit is a two transistor transmitter using 2N2222A devices. These are small-signal devices so the transmitter output is only about ½ W. What might surprise the reader is the number of inductors (coils) in the design, but the circuit has good performance in impedance matching and harmonic reduction. A careful design ensures a clean output waveform, low in harmonic and other spurious content.

The oscillator is a variable frequency crystal oscillator (VXO) in which the 100 pF variable capacitor between the crystal and ground enables the oscillator frequency to be shifted. Depending upon individual crystals, that movement can be on the order of 5 kHz. Increasing the variable capacitance too much may cause the transistor to cease oscillating. The 150 pF capacitor in the emitter of the oscillator controls the oscillator feedback. This value is a compromise that appears to work well — but if the transistor fails to oscillate, try increasing this value a little.

The 4.7 μF electrolytic capacitor across the KEY connection minimizes key clicks by softening the keying waveform. Lowering this value gives a harder keying waveform and increasing it will further soften the waveform. A 10 Ω resistor is added to the base of the transistor to reduce the risk of high frequency parasitic oscillation.

A 22 μH RF choke (RFC1) provides the RF collector dc supply for the oscillator. A molded inductor was used in the original design. If this isn't available, about six turns of thin enameled wire wound through a small ferrite bead (type 43 mix) would give roughly the same inductance value.

The output from the collector goes to a single-element harmonic filter in a pi configuration. A 100 pF variable capacitor tunes the filter to resonance at the oscillator frequency. It should be adjusted for maximum output consistent with a clean CW signal.

If the intent is to use the oscillator stage as a transmitter without additional amplification, the filter is designed to have a 50 Ω output impedance for a matched 7 MHz antenna. The antenna can be connected in place of T1 in the schematic.

A 4:1 impedance ratio transformer (T1) matches the output of the oscillator to the

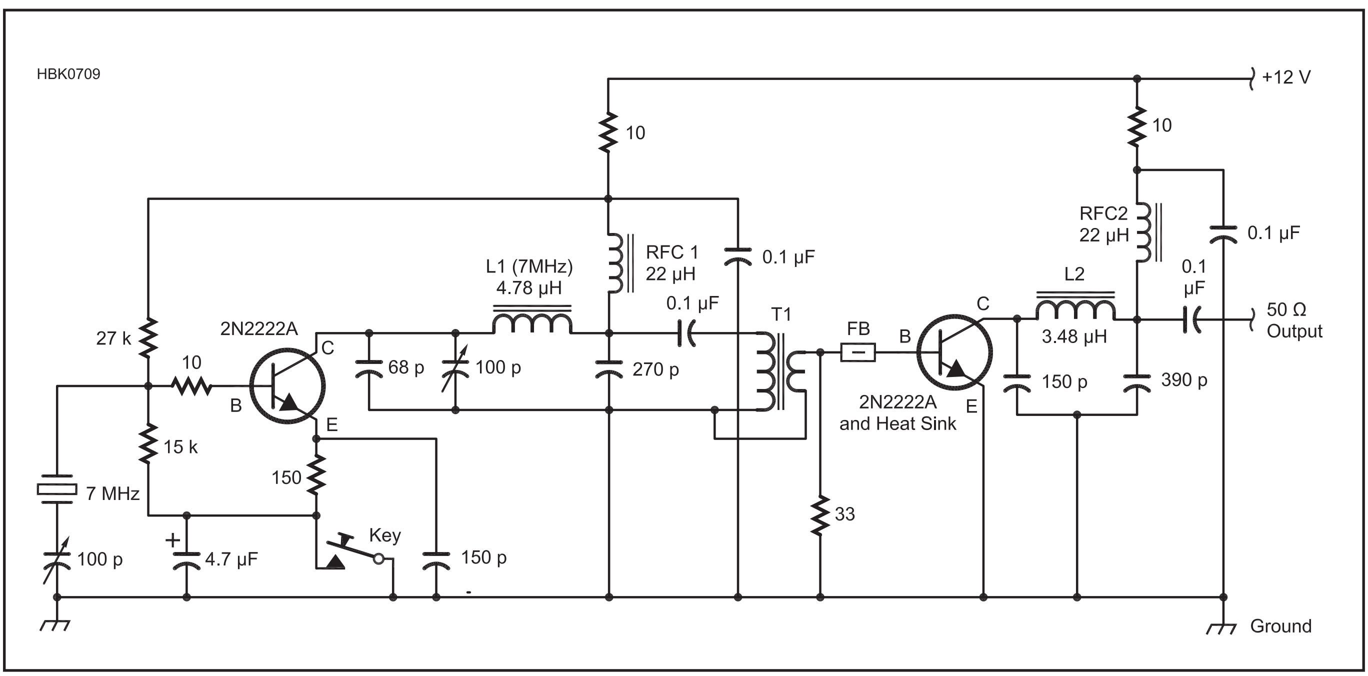


Fig 13.6 — The circuit of the Pebble Crusher transmitter. The oscillator section at left can be used as a standalone transmitter by replacing T1 with an antenna. The output of the amplifier circuit is approximately ½ W.

L1 — 32t #30 AWG on T50-6 toroid core
L2 — 28t #24 AWG on T50-6 toroid core
RFC1, RFC2 — 22 μH or 6t #30 AWG on ferrite bead (type 43 mix)
T1 — 12t #24 AWG (primary) and 6t #24 AWG (secondary) on FT37-43 toroid core (see text)
FB — ferrite bead (type 43 mix)

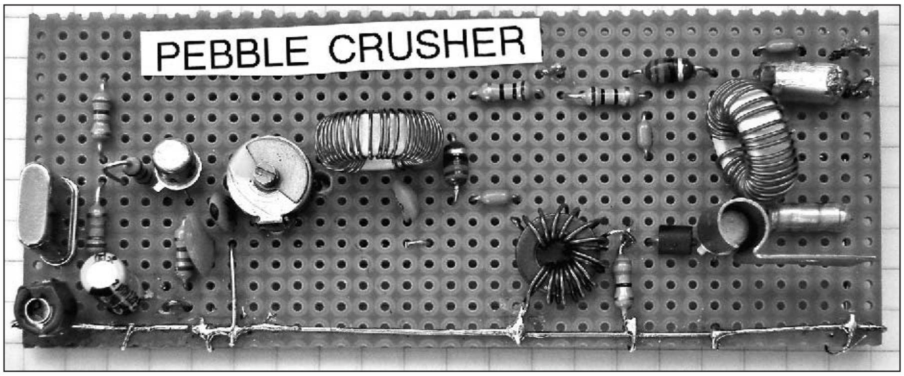


Fig 13.7 — G3RJV's "Anglicised" version of the Pebble Crusher is built on 0.1 inch spacing perfboard. The heat sink for the output transistor at lower right can be a commercial unit or created from metal strip or tubing (see text).

core and L2 has 28 turns of #24 AWG enameled copper wire also on a T50-6 core. The transformer (T1) is wound on a ferrite FT37-43 core. Wind the primary first with 12 turns of #24 AWG spread out over three-quarters of the core circumference. Then add the secondary winding which is made up of 6 turns of #24 AWG wire between turns of the primary winding at the grounded end.

Fig 13.7 shows 2N2222A transistors in metal TO-18 cases, but the plastic TO-92 version would also work. An advantage of using the metal TO-18 is that a small heat sink can be attached to the output transistor to dissipate surplus heat. If a commercial heat sink is not available, a small piece of aluminum or copper strip can be formed into a heat sink. The strip is wrapped around a drill bit the same diameter as the transistor case, one side overlapping where the ends of the metal meet. This is then squeezed to make a tight fit on the TO-18 case. A small piece of ¼ inch OD brass tubing slightly flattened and cemented to the transistor casing with epoxy will work as well.

13.1.3 Amplitude-Modulated Full-Carrier Voice Transmission

While the telegraph key in the transmitters of the previous section can be considered a *modulator* of sorts, we usually reserve that term for a somewhat more sophisticated system that adds information to the transmitted signal. As noted earlier, there are three signal parameters that can be used to modulate a radio signal and they all can be used in various ways to add voice (or other information) to a transmitted signal.

One way to add voice to a radio signal is to first convert the analog signal to digital data and then transmit it as "ones" and "zeros." This can be done even using the simple telegraph transmitters of the last section. This is a technique frequently employed for some applications using data applied to our "pulse" transmitter described earlier. Here we will talk about the more direct application of the analog voice signal to a radio signal.

A popular form of voice amplitude modulation is called *high-level amplitude modulation.* It is generated by mixing (or modulating) an RF carrier with an audio signal. **Fig 13.8** shows the conceptual view of this**. Fig 13.9** is a more detailed view of how such a voice transmitter would actually be implemented. The upper portion is the RF channel, and you can think of the previously described Tuna Tin Two transmitter as a transmitter that we could use, after shifting the frequency into the voice portion of the band. The lower portion is the audio frequency or AF channel, usually called the *modulator*, and is nothing more than an audio amplifier designed to be fed from a microphone and with an output designed to match the anode or collector impedance of the final RF amplifier stage.

The output power of the modulator is applied in series with the dc supply of the output stage (only) of the RF channel of the transmitter. The level of the voice peaks needs to

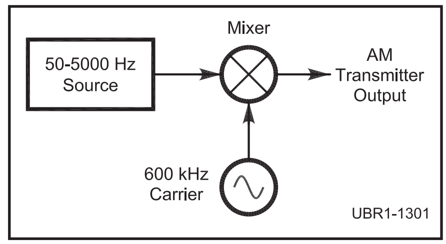


Fig 13.8 — Block diagram of a conceptual AM transmitter.

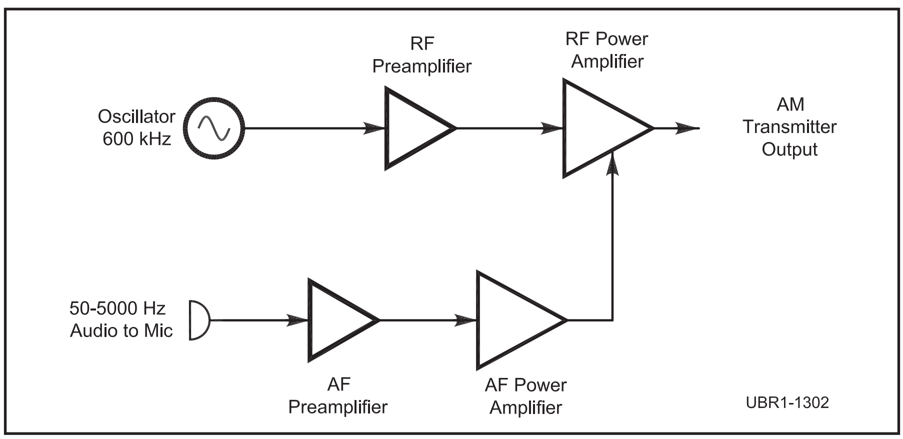


Fig 13.9 — Block diagram of a 600 kHz AM broadcast transmitter.

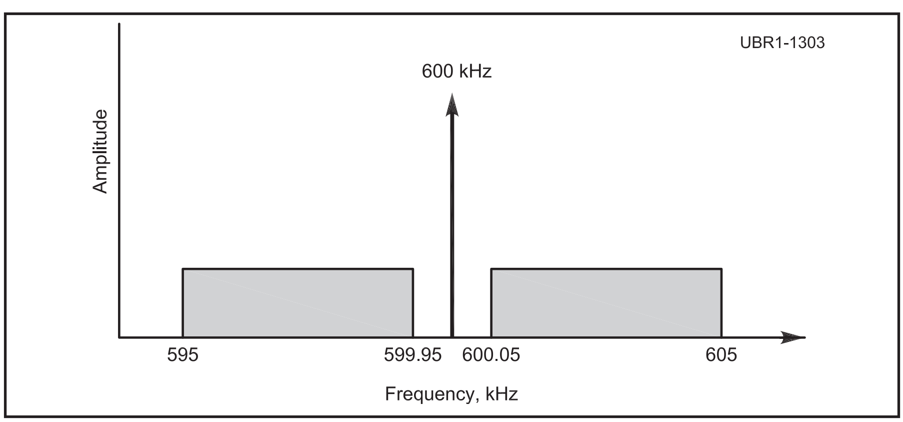


Fig 13.10 — The range of of spectrum used by a 600 kHz AM broadcast signal showing sidebands above and below a carrier at 600 kHz.

be just enough to vary the supply to the RF amplifier collector between zero volts, on negative peaks, and twice the normal supply voltage on positive voice peaks. This usually requires an AF amplifier with about half the average power output as the dc input power (product of dc collector or plate voltage times the current) of the final RF amplifier stage.

The output signal, called *full-carrier double-sideband AM*, occupies a frequency spectrum as shown in **Fig 13.10.** The spectrum shown would be that of a standard broadcast station with an audio passband from 50 Hz to 5 kHz. Note that the resulting channel width is twice the highest audio frequency transmitted. If the audio bandwidth were limited to typical "telephone quality speech" of 300 to 3300 Hz, the resulting bandwidth would be reduced to 6.6 kHz. Note also that while a perfect multiplication process would result in just two sidebands and no carrier, this implementation actually provides the sum of the carrier and the sidebands from the product terms. (See the **Receivers** chapter for the mathematical description of signal multiplication.)

Full-carrier double-sideband AM is used in fewer and fewer applications. The spectral and power efficiency are significantly lower than single sideband (SSB), and the equipment becomes quite costly as power is increased. The primary application is in broadcasting — largely because AM transmissions can be received on the simplest and least expensive of receivers. With a single transmitter and thousands of receivers, the overall system cost may be less and the audience larger than for systems that use more efficient modulation techniques. While the PEP output of an AM transmitter is four times the carrier power, none of the carrier power is necessary to carry the information, as we will discuss in the next section.

13.1.4 Single-Sideband Suppressed-Carrier Transmission

The two sidebands of a standard AM transmitter carry (reversed) copies of the same information, and the carrier carries essentially no information. We can more efficiently transmit the information with just one of the sidebands and no carrier. In so doing, we use somewhat less than half the bandwidth, a scarce resource, and also consume much less transmitter power by not transmitting the carrier and the second sideband.

SINGLE-SIDEBAND — THE FILTER METHOD

The block diagram of a simple single-sideband suppressed-carrier (SSB) transmitter is shown in **Fig 13.11**. This transmitter uses a balanced mixer as a *balanced modulator* to generate a double sideband suppressed carrier signal without a carrier. (See the **Mixers, Modulators, and Demodulators** chapter for more information on these circuits.) That signal is then sent through a filter designed to pass just one (either one, by agreement with the receiving station) of the sidebands.

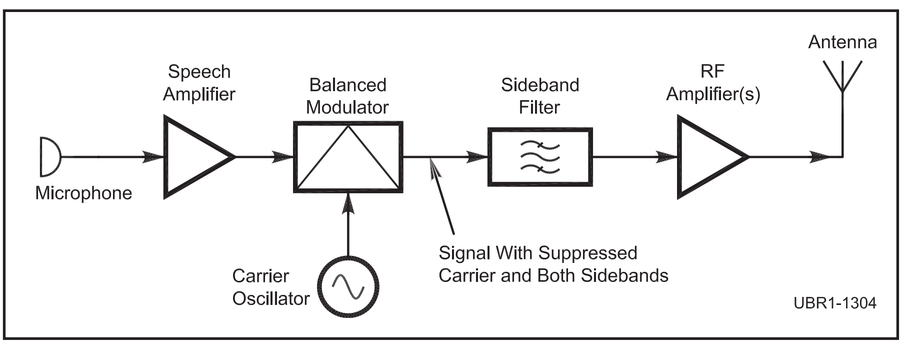


Fig 13.11 — Block diagram of a filter type single-sideband suppressed-carrier (SSB) transmitter.

Depending on whether the sideband above or below the carrier frequency is selected, the signal is called *upper sideband* (*USB*) or *lower sideband* (*LSB*), respectively. The resulting SSB signal is amplified to the desired power level and we have an SSB transmitter. Amateur practice is to use USB above 10 MHz and LSB on lower frequencies. The exception is 60 meter channels, on which amateurs are required to use USB for data and voice signals.

While a transmitter of the type in Fig 13.11 with all processing at the desired transmit frequency will work, the configuration is not often used. Instead, the carrier oscillator and sideband filter are often at an intermediate frequency that is heterodyned to the operating frequency as shown in **Fig 13.12**. The reason is that the sideband filter is a complex narrowband filter and most manufacturers would rather not have to supply a new filter design every time a transmitter is ordered for a new frequency. Many SSB transmitters can operate on different bands as well, so this avoids the cost of additional mixers, oscillators and expensive filters.

Note that the block diagram of our SSB transmitter bears a striking resemblance to the diagram of a superheterodyne receiver as shown in the **Receivers** chapter, except that the signal path is reversed to begin with information and produce an RF signal. The same kind of image rejection requirements for intermediate frequency selection that were design constraints for the superhet receiver applies here as well.

SINGLE-SIDEBAND — THE PHASING METHOD

Most current transmitters use the method

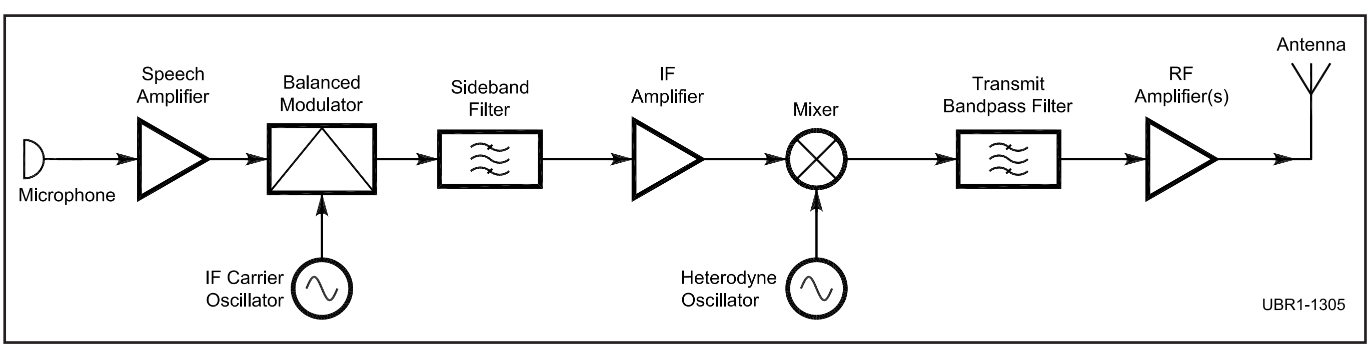


Fig 13.12 — Block diagram of a heterodyne filter-type SSB transmitter for multiple frequency operation.

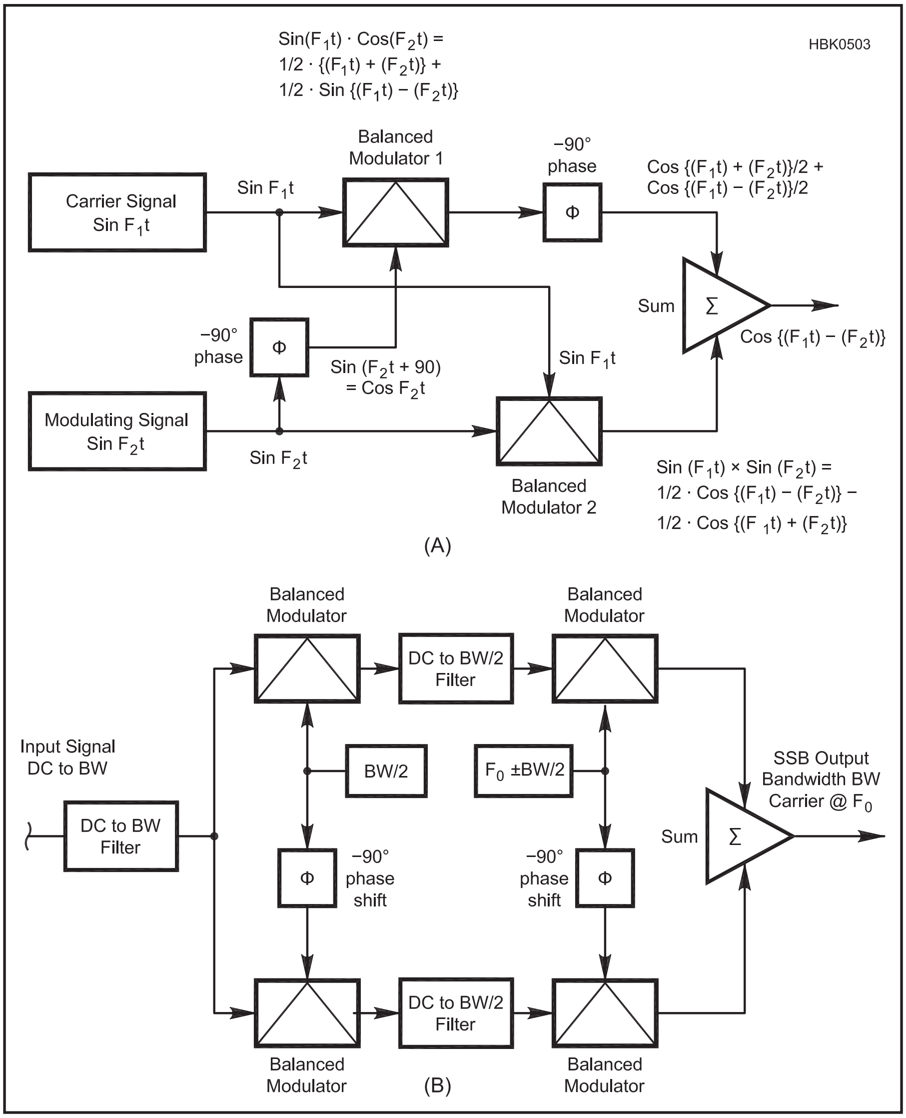


Fig 13.13 — Block diagrams of phasing type SSB transmitters for single frequency operation.

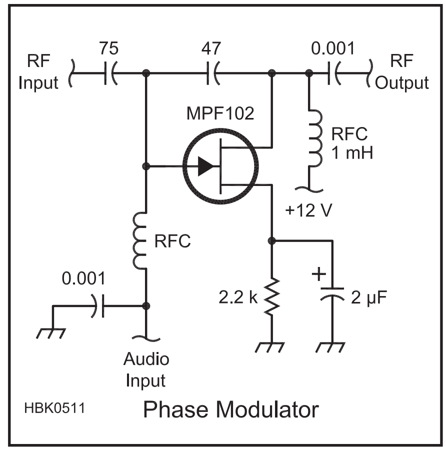


Fig 13.14 — Simple FET phase modulator circuit.

of SSB generation shown in Fig 13.11 and discussed in the previous section to generate the SSB signal. That is the *filter method*, but really occurs in two steps — first a balanced modulator is used to generate sidebands and eliminate the carrier, then a filter is used to eliminate the undesired sideband, and often to improve carrier suppression as well.

The *phasing method* of SSB generation is exactly the same as the image-rejecting mixer described in the **Receivers** chapter. This uses two balanced modulators and a phase-shift network for both the audio and RF carrier signals to produce the upper sideband signal as shown in **Fig 13.13A**. By a shift in the sign of either of the phase-shift networks, the opposite sideband can be generated. This method trades a few phase-shift networks and an extra balanced modulator for the sharp sideband filter of the filter method. While it looks deceptively simple, a limitation is in the construction of a phase-shift network that will have a constant 90° phase shift over the whole audio range. Errors in phase shift result in less than full carrier and sideband suppression. Nonetheless, there have been some successful examples offered over the years.

SINGLE-SIDEBAND — THE WEAVER METHOD

Taking the phasing method one step further, the Weaver method solves the problem of requiring phase-shift networks that must be aligned across the entire audio range. Instead, the Weaver method, shown in Fig 13.13B, first mixes one copy of the message (shown with a bandwidth of dc to BW Hz) with an in-band signal at BW/2 Hz and another copy with a signal at BW/2 Hz that is phase-shifted by –90°. Instead of phase-shifting the message, only the signal at BW/2 Hz must be phase-shifted — a much simpler task!

The output of each balanced modulator is filtered, leaving only components from dc to BW/2. These signals are then input to a second pair of balanced modulators with a more conventional LO signal at the carrier frequency, f_0, offset by +BW/2 for USB and –BW/2 for LSB. The output of the balanced modulators is summed to produce the final SSB signal.

The Weaver method is difficult to implement in analog circuitry, but is well-suited to digital signal processing systems. The Weaver method has become common in DSP-based equipment that generates the SSB signal digitally.

13.1.5 Angle-Modulated Transmitters

Transmitters using frequency modulation (FM) or phase modulation (PM) are generally grouped into the category of *angle modulation* since the resulting signals are often indistinguishable. An instantaneous change in either frequency or phase can create identical signals, even though the method of modulating the signal is somewhat different. To generate an FM signal, we need an oscillator whose frequency can be changed by the modulating signal.

We can make use of an oscillator whose frequency can be changed by a "tuning voltage." If we apply a voice signal to the TUNING VOLTAGE connection point, we will change the frequency with the amplitude and frequency of the applied modulating signal, resulting in an FM signal.

The phase of a signal can be varied by changing the values of an R-C phase-shift network. One way to accomplish phase modulation is to have an active element shift the phase and generate a PM signal. In **Fig 13.14**, the current through the field-effect transistor is varied with the applied modulating signal,

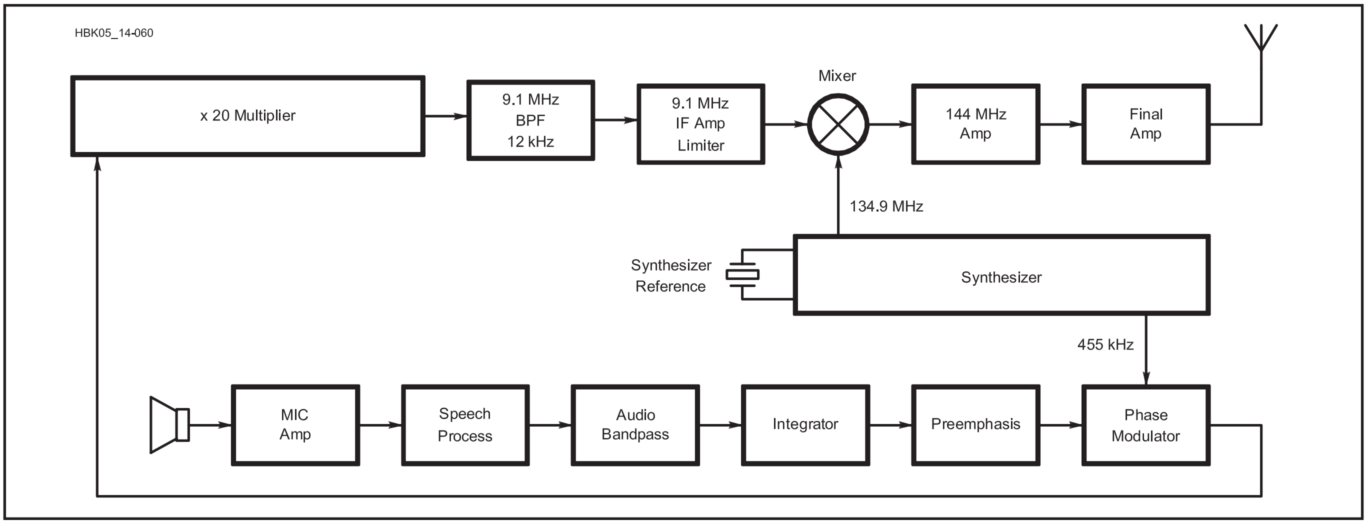


Fig 13.15 — Block diagram of a VHF/UHF NBFM transmitter using the indirect FM (phase modulation) method.

varying the phase shift at the stage's output. Because the effective load on the stage is changed, the carrier is also amplitude-modulated and must be run through an FM receiver-type limiter in order to remove the amplitude variations.

FREQUENCY MODULATION TRANSMITTER DESIGN

Frequency modulation is widely used as the voice mode on VHF for repeater and other point-to-point communications. **Fig 13.15** shows the phase-modulation method, also known as *indirect FM*, as used in many FM transmitters. It is the most widely used approach to FM. Phase modulation is performed at a low frequency, say 455 kHz. Prior to the phase modulator, speech filtering and processing perform four functions:

1. Convert phase modulation to frequency modulation (see below).
2. Apply pre-emphasis (high-pass filtering) to the speech audio higher speech frequencies for improved signal-to-noise ratio after de-emphasis (low-pass filtering) of the received audio.
3. Perform speech processing to emphasize the weaker speech components.
4. Compensate for the microphone's frequency response and possibly also the operator's voice characteristics.

Multiplier stages then move the signal to some desired higher IF and also multiply the frequency deviation to the desired final value. If the FM deviation generated in the 455 kHz modulator is 250 Hz, the deviation at 9.1 MHz is 20 × 250, or 5 kHz.

Frequency Multipliers

Frequency multipliers are frequently used in FM transmitters as a way to increase the deviation along with the carrier frequency. They are composed of devices that exhibit high levels of harmonic distortion, usually an undesired output product. In this case the desired harmonic is selected and enhanced through filtering. The following examples show the way this can be done, both with amplifiers and with passive diode circuits.

A passive multiplier using diodes is shown in **Fig 13.16A**. The full-wave rectifier circuit can be recognized, except that the dc component is shorted to ground. If the fundamental frequency ac input is 1.0 V_{RMS}, the second harmonic is 0.42 V_{RMS} or 8 dB below the input, including some small diode losses. This value is found by calculating the Fourier series coefficients for the full-wave-rectified sine wave, as shown in many textbooks.

Transistor and vacuum-tube frequency multipliers operate on the following principle: if a sine wave input causes the plate/collector/drain current to be distorted (not a sine wave) then harmonics of the input are generated. If an output resonant circuit is tuned to a harmonic, the output at the harmonic is emphasized and other frequencies are attenuated. For a particular harmonic the current pulse should be distorted in a way that maximizes that harmonic. For example, for a doubler the current pulse should look like a half-wave rectified sine wave (180° of conduction). A transistor with Class B bias would be a good choice. For a tripler, use 120° of conduction (Class C).

An FET, biased at a certain point, is very nearly a *square-law* device as described in the **Analog Basics** chapter. That is, the drain-current change is proportional to the square of the gate-voltage change. It is then an efficient frequency doubler that also de-emphasizes the fundamental.

A push-push doubler is shown in Fig 13.16B. The FETs are biased in the square-law region and the BALANCE potentiometer minimizes the fundamental frequency. Note that the gates are in push-pull and the drains are in parallel. This causes second harmonics to add in-phase at the output and fundamental components to cancel.

Fig 13.16C shows an example of a single-ended doubler using a bipolar transistor. The efficiency of a doubler of this type is typically 50%, that of a tripler 33%, and of a quadrupler 25%. Harmonics other than the one to which the output tank is tuned will appear in the output unless effective band-pass filtering is applied. The collector tap on L1 is placed at the point that offers the best compromise between power output and spectral purity.

A push-pull tripler is shown in Fig 13.16D. The input and output are both push-pull. The balance potentiometer minimizes even harmonics. Note that the transistors have no bias voltage in the base circuit; this places the transistors in Class C for efficient third-harmonic production. Choose an input drive level that maximizes harmonic output.

The step recovery diode (SRD) shown in **Fig 13.17A** is an excellent device for harmonic generation, especially at microwave frequencies. The basic idea of the SRD is as follows: When the diode is forward conducting, a charge is stored in the diode's diffusion capacitance, and if the diode is quickly reverse-biased, the stored charge is very suddenly released into an LC harmonic-tuned circuit. The circuit is also called a "comb generator" because of the large number of harmonics that are generated. (The spectral display looks like a comb.) A phase-locked loop (PLL) can then lock onto the desired harmonic.

A varactor diode can also be used as a multiplier. Fig 13.17B shows an example.

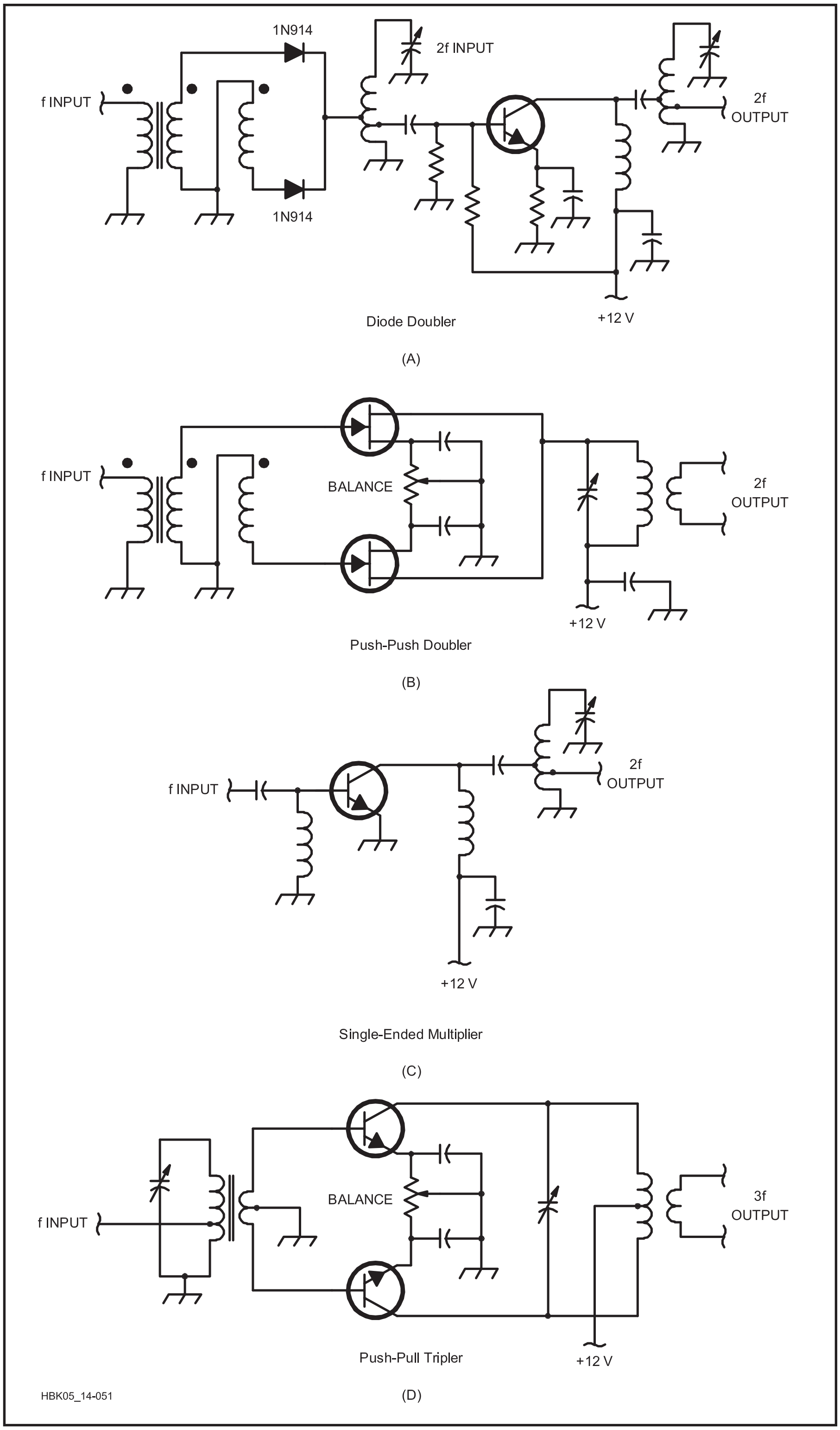


Fig 13.16 — A: diode doubler. B: push-push doubler using JFETS. C: single-ended multiplier using a BJT. D: push-pull tripler using BJTs.

This circuit depends on the fact that the capacitance of a varactor changes with the instantaneous value of the voltage across it, in this case the RF excitation voltage. This is a nonlinear process that generates harmonic currents through the diode. Power levels up to 25 W can be generated in this manner.

Following frequency multiplication, a second conversion to the final output frequency is performed. Prior to this final translation, IF band-pass filtering is performed in order to minimize adjacent channel interference that might be caused by excessive frequency deviation. This filter needs good phase linearity to assure that the FM sidebands maintain the correct phase relationships. If this is not done, an AM component is introduced to the signal, which can cause nonlinear distortion problems in the PA stages. The final frequency translation retains a constant value of FM deviation for any value of the output signal frequency.

The IF/RF amplifiers can be nonlinear Class C amplifiers because the signal in each amplifier contains, at any one instant, only a single value of instantaneous frequency and not multiple simultaneous frequencies whose relationship must be preserved as in SSB. These amplifiers are not sources of IMD, so they need not be "linear." The sidebands that appear in the output are a result only of the FM process. (The spectrum of an FM signal is described by Bessel functions.)

In phase modulation, the frequency deviation is directly proportional to the frequency of the audio signal. (In FM, the deviation is proportional to the audio signal's amplitude.) To make deviation independent of the audio frequency, an audio-frequency response that rolls off at 6 dB per octave is needed. An op-amp low-pass circuit in the audio amplifier accomplishes this function. This process converts phase modulation to frequency modulation.

In addition, audio speech processing helps to maintain a constant value of speech amplitude, and therefore constant IF deviation, with respect to audio speech levels. Preemphasis of speech frequencies (a 6 dB per octave high-pass response from 300 to 3000 Hz) is commonly used to improve the signal-to-noise ratio at the receive end. Analysis shows that this is especially effective in FM systems when the corresponding de-emphasis (complementary low-pass response) is used at the receiver. (See reference for Schwartz.) By increasing the amplitude of the higher audio frequencies before transmission and then reducing them in the receiver, high-frequency audio noise from the demodulation process is also reduced, resulting in a "flat" audio response with lower hiss and high-frequency noise.

An IF limiter stage may be used to ensure that any amplitude changes that are created during the modulation process are removed. The indirect-FM method allows complete frequency synthesis to be used in all the transmitter local oscillators (LOs), so that the channelization of the output frequency is very accurate. The IF and RF amplifier stages are operated in a highly efficient Class-C mode, which is helpful in portable equipment operating on small internal batteries.

FM is more tolerant of frequency misalignments between the transmitter and receiver than is SSB. In commercial SSB commu-

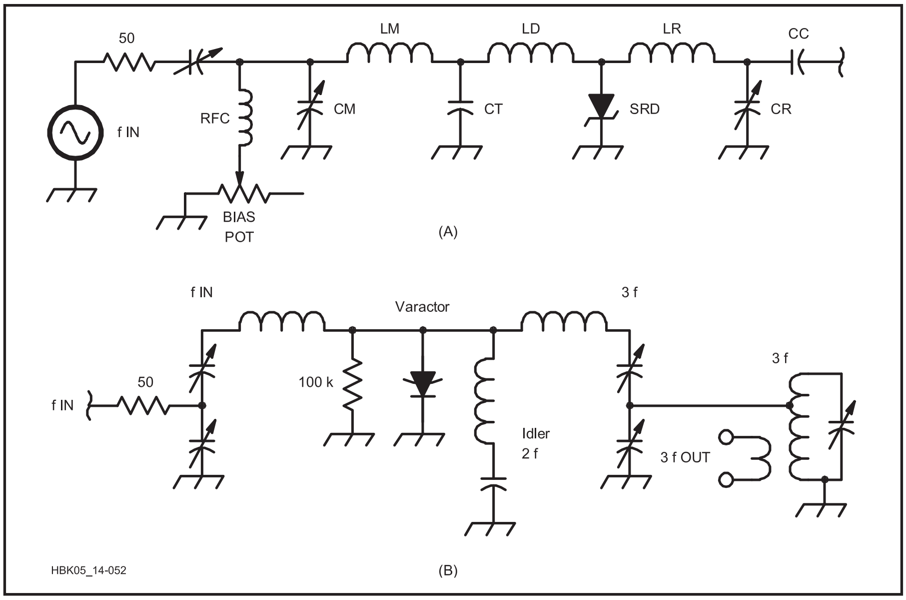


Fig 13.17 — Diode frequency multipliers. A: step-recovery diode multiplier. B: varactor diode multiplier.

nication systems, this problem is solved by transmitting a *pilot carrier* with an amplitude 10 or 12 dB below the full PEP output level. The receiver is then phase-locked to this pilot carrier. The pilot carrier is also used for squelch and AGC purposes. A short-duration "memory" feature in the receiver bridges across brief pilot-carrier dropouts, caused by multipath nulls.

In a "direct FM" transmitter, a high-frequency (say, 9 MHz or so) crystal oscillator is frequency-modulated by varying the voltage on a varactor diode. The audio is pre-emphasized and processed ahead of the frequency modulator as for indirect-FM.

13.1.6 The Superhet SSB/CW Transmitter

The modern linear transmitting chain makes use of the concepts presented previously. We will now go through them in more detail. The same kind of mixing schemes, IF frequencies and IF filters that are used for superhet receivers can be, and very often are, used for a transmitter. In the following discussion, "in-band" refers to signal frequencies within the bandwidth of the desired signal. For example, for an upper sideband voice signal with a carrier frequency of 14.200 MHz, frequencies of approximately 14.2003 to 14.203 would be considered in-band. Out-of-band refers to frequencies outside this range, such as on adjacent channels.

For this example we will use the commonly-encountered dual-conversion scheme with the SSB generation at 455 kHz, a conversion to a 70 MHz IF, and then a final conversion to the HF operating frequency. **Fig 13.18** is a block diagram of the system

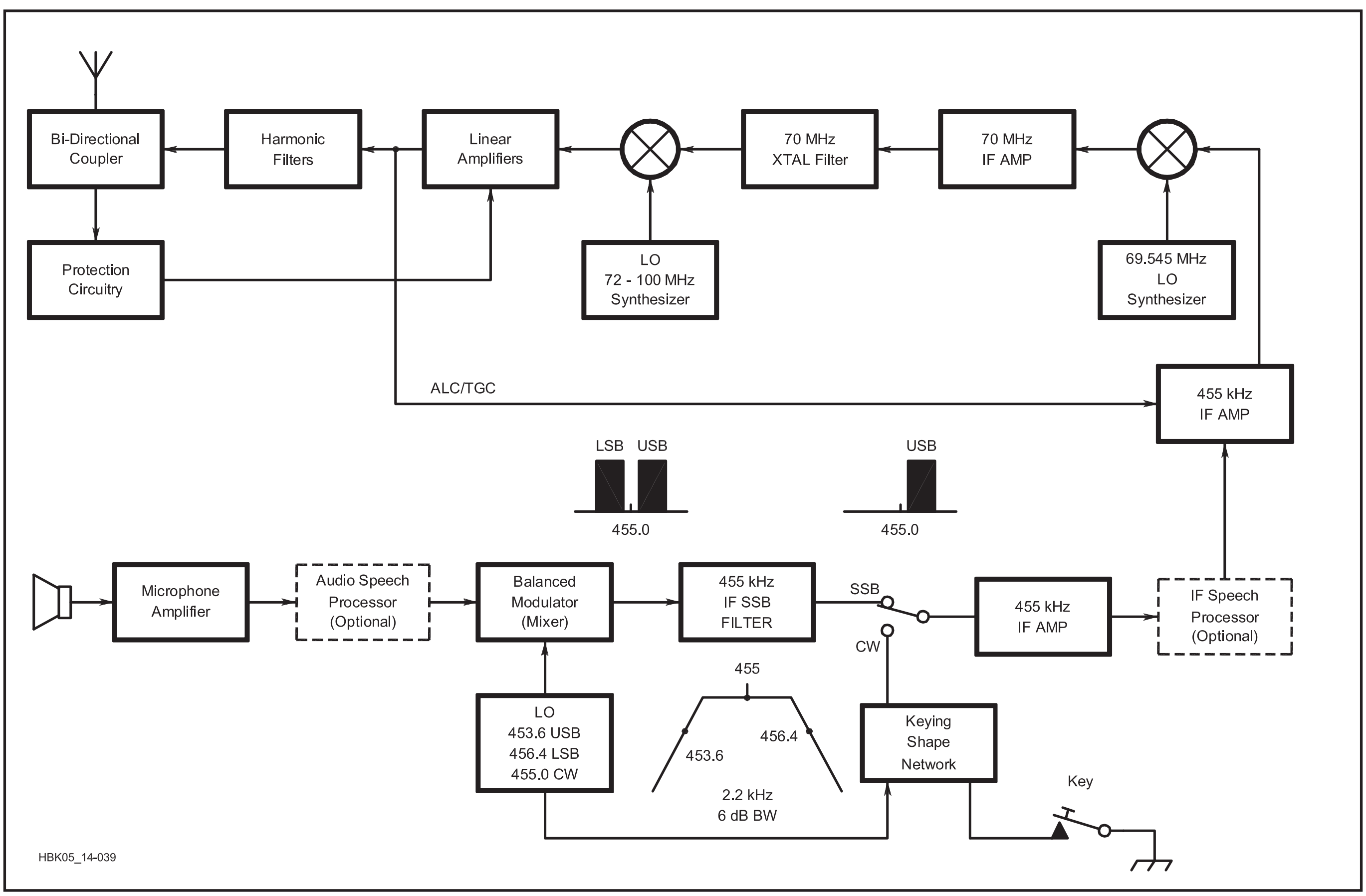


Fig 13.18 — Block diagram of an upconversion SSB/CW transmitter.

under consideration. Let's discuss the various elements in detail, starting at the microphone.

MICROPHONES

A microphone (mic) is a transducer that converts sound waves into electrical signals. For communications quality speech, its frequency response should be as flat as possible from around 200 to above 3500 Hz. Response peaks in the microphone can increase the peak to average ratio of speech, which then degrades (increases) the peak to average ratio of the transmitted signal. Some transmitters use "speech processing," which is essentially a specialized form of speech amplification either at audio or IF/RF. Since most microphones pick up a lot of background ambient noises, the output of the transmitter due to background noise pickup in the absence of speech may be as much as 20 dB greater than without speech processing. A *noise canceling* microphone is recommended to reduce this background pickup if there is much background noise. Microphone output levels vary, depending on the microphone type. Typical amateur microphones produce about 10 to 100 mV output levels.

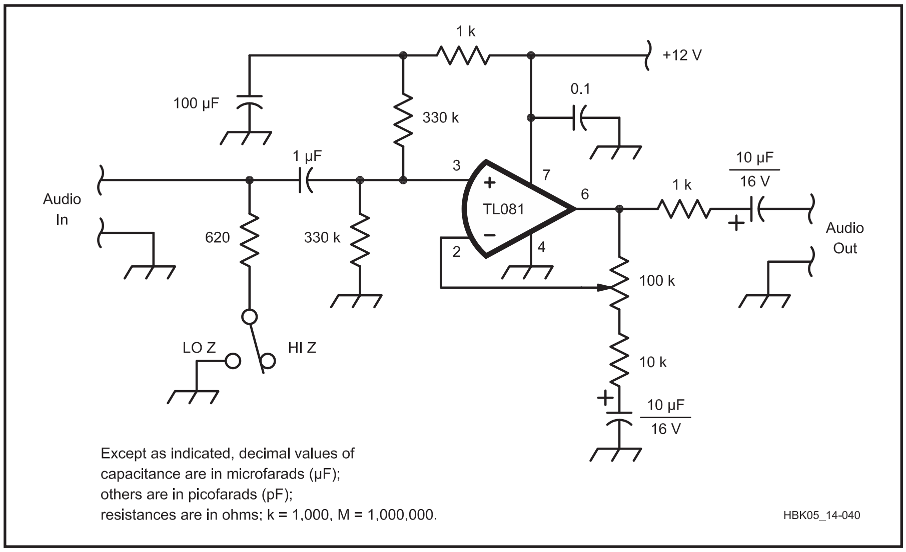


Fig 13.19 — Schematic diagram of a microphone amplifier suitable for high or low impedance microphones.

Ceramic

Ceramic mics have high output impedances but low level outputs. They require a high-resistance load (usually about 50 kΩ) for flat frequency response and lose low-frequency response as this resistance is reduced (electrically, the mic "looks like" a small capacitor). These mics vary widely in quality, so a cheap mic is not a good bargain because of its effect on the transmitted power level and generally poor speech quality.

Dynamic

A dynamic mic resembles a small loudspeaker, with an impedance of about 600 Ω and an output of about 12 mV on voice peaks. Early dynamic mics (designed for vacuum tube transmitters) included a built-in transformer to transform the impedance to 100 kΩ suitable for high input impedance speech amplifiers. Currently available dynamic mics provide the output directly, although the transformers are available for connection outside the mic. Dynamic mics are widely used by amateurs.

Electret

Electret mics use a piece of special insulator material, such as Teflon, that contains a "trapped" polarization charge (Q) at its surfaces to create a capacitance (C). Sound waves modulate the capacitance of the material and cause a voltage change according to $\Delta V/V = -\Delta C/C$. For small changes in capacitance the change in voltage is almost linear. These mics have been greatly improved in recent years and are used in most cellular handsets and computer headsets.

The output level of the electret is fairly low, and an integrated preamp is generally included in the mic cartridge. A voltage of about 4 V dc is required to power the preamp, and some commercial transceivers provide this voltage at the mic connector. The dc voltage must be blocked by a coupling capacitor if a dynamic mic element is to be used with a transceiver that supplies power to the microphone. The dynamic mic is unlikely to be damaged by the applied voltage, but the usual symptom is very low audio output with a muffled sound.

In recent years, micro electro-mechanical systems (MEMS) technology has been applied to microphones. The sound pressure is applied to a small transducer integrated on a silicon chip, which creates a mechanical motion in response to the force of the sound waves. This motion is converted into an electrical signal, usually by the varying capacitance of the transducer. Mass production of MEMS microphones is being driven by the cellular handset market.

MICROPHONE AMPLIFIERS

The balanced modulator and (or) the audio speech processor need a certain optimum level, which can be in the range of 0.3 to 0.6 V ac into perhaps 1 kΩ to 10 kΩ. Excess noise generated within the microphone amplifier should be minimized, especially if speech processing is used. The circuit in **Fig 13.19** uses a low-noise BiFET op amp. The 620 Ω resistor is selected for a low impedance microphone, and switched out of the circuit for high-impedance mics. The amplifier gain is set by the 100 kΩ potentiometer.

It is also a good idea to experiment with the low-and high-frequency responses of the mic amplifier to compensate for the frequency response of the mic and the voice of the operator.

SPEECH PROCESSING

The output of the speech amplifier can be applied directly to the balanced modulator. The resulting signal will be reproducible with the maximum fidelity and dynamic range available for the bandwidth provided. The communications efficiency of the system will depend on the characteristics of the particular voice used. The usual peak-to-average ratio is such that the average transmitted power is on the order of 5% of the peak power.

For communications use, it is generally beneficial to increase the average-to-peak ratio by distorting the speech waveform in a measured way. If one were to merely increase the amplifier gain, a higher average power would be obtained. The problem is that clipping, heavy distortion and likely spurious signals beyond legal limits would be generated. By carefully modifying the speech waveform before application to the balanced modulator, a synthetic waveform with a higher average-to-peak waveform can be generated that will retain most of the individual voice characteristics and avoid spurious signals. We will discuss two such techniques of *speech processing* below and another as we move into the RF circuitry.

Note that speech processing should not be used with most digital mode transmissions if the digital signal is generated by audio tones applied to the transmitter's microphone input. The modulation of these signals requires linear processing and amplification to preserve the waveform shape and minimize distortion products. (The same caution applies to the ALC function as discussed in that section.)

Audio Speech Clipping

If the audio signal from the microphone amplifier is further amplified, say by as much as 12 dB, and then if the peaks are clipped (sometimes called *slicing* or *limiting*) by 12 dB by a speech clipper, the output peak value is the same as before the clipper, but the average value is increased considerably. The resulting signal contains harmonics and IMD but the speech intelligibility, especially in a white-noise background, is improved by 5 or 6 dB.

The clipped waveform frequently tends to have a square-wave appearance, especially on voice peaks. It is then band-pass filtered to remove frequencies below 300 and above 3000 Hz. The filtering of this signal can create a "repeaking" effect. That is, the peak value tends to increase noticeably above its clipped value.

An SSB generator responds poorly to a square-wave audio signal, creating significant peaks in the RF envelope. (This is described mathematically as the Hilbert Transform effect.) These peaks cause out-of-band splatter in the transmitter's linear output power amplifier unless Automatic Level Control (ALC, to be discussed later) cuts back on the RF gain. The peaks increase the peak-to-average ratio and the ALC reduces the average SSB power output, thereby reducing some of the benefit of the speech processing. The square-wave effect is also reduced by band-pass filtering (300 to 3000 Hz) the input to the clipper as well as the output.

Fig 13.20 is a circuit for a simple audio speech clipper. A CLIP LEVEL potentiometer before the clipper controls the amount of clipping and an OUTPUT LEVEL potentiometer controls the drive level to the balanced modulator. The correct adjustment of these potentiometers is done with a two-tone audio input or by talking into the microphone, rather than driving with a single tone, because single tones don't exhibit the repeaking effect.

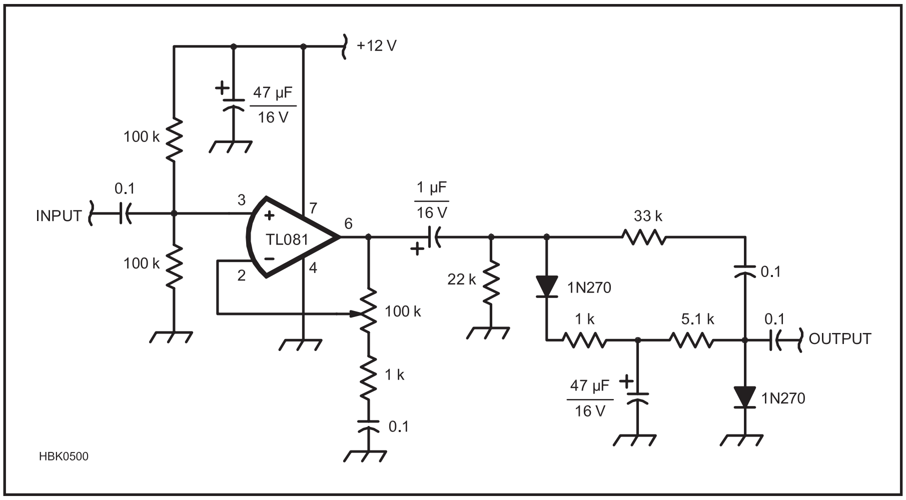


Fig 13.21 — A forward acting speech compressor circuit.

Audio Speech Compression

Although it is desirable to keep the voice level as high as possible, it is difficult to maintain constant voice intensity when speaking into the microphone. To overcome this variable output level, it is possible to use an automatic gain control that follows the average variations in speech amplitude. This can be done by rectifying and filtering some of the audio output and applying the resultant dc to a control terminal in an early stage of the amplifier.

If an audio AGC circuit derives control voltage from the output signal, the system is a closed loop. If short attack time is necessary, the rectifier-filter bandwidth must be opened up to allow syllabic modulation of the control voltage. This allows some of the voice frequency signal to enter the control terminal, causing distortion and instability. Because the syllabic frequency and speech-tone frequencies have relatively small separation, the simpler feedback AGC systems compromise fidelity for fast response.

Problems with loop dynamics in audio AGC can be side-stepped by eliminating the loop and using a forward-acting system. The control voltage is derived from the input of the amplifier, rather than from the output. Eliminating the feedback loop allows unconditional stability, but the trade-off between response time and fidelity remains. Care must be taken to avoid excessive gain between the signal input and the control voltage output. Otherwise the transfer characteristic can reverse; that is, an increase in input level can cause a decrease in output. A simple forward-acting compressor is shown in **Fig 13.21**.

BALANCED MODULATORS

A balanced modulator is a mixer. A more complete discussion of balanced modulator design was provided in the **Mixers, Modulators, and Demodulators** chapter. Briefly, the IF frequency LO (455 kHz in the example of Fig 13.18) translates the audio frequencies up to a pair of IF frequencies, the LO plus the audio frequency and the LO minus the audio frequency. The balance from the LO port to the IF output causes the LO frequency to be suppressed by 30 to 40 dB. Adjustments are provided to improve the LO null.

The filter method of SSB generation uses an IF band-pass filter to pass one of the sidebands and block the other. In Fig 13.18 the filter is centered at 455.0 kHz. The LO is offset to 453.6 kHz or 456.4 kHz so that the upper sideband or the lower sideband (respectively) can pass through the filter. This creates a problem for the other LOs in the radio, because they must now be properly offset so that the final transmit output's carrier (suppressed) frequency coincides with the frequency readout on the front panel of the radio.

Various schemes have been used to create the necessary LO offsets. One method uses two crystals for the 69.545 MHz LO that

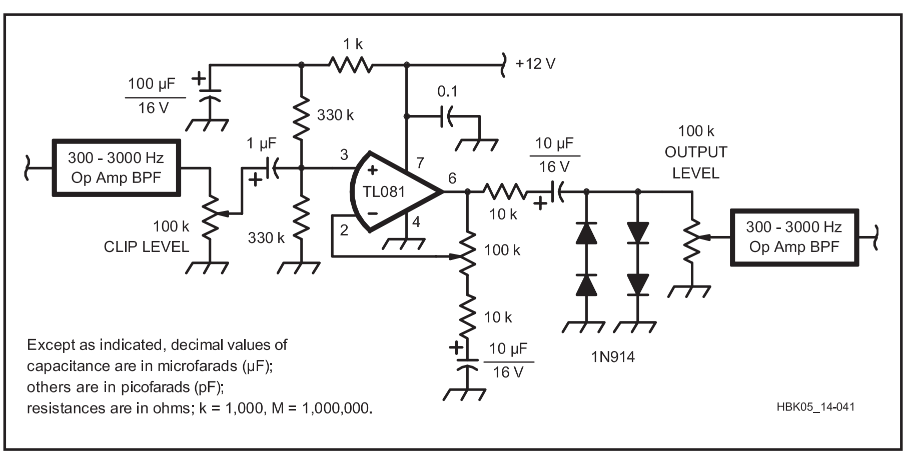


Fig 13.20 — Schematic diagram a simple audio speech clipper.

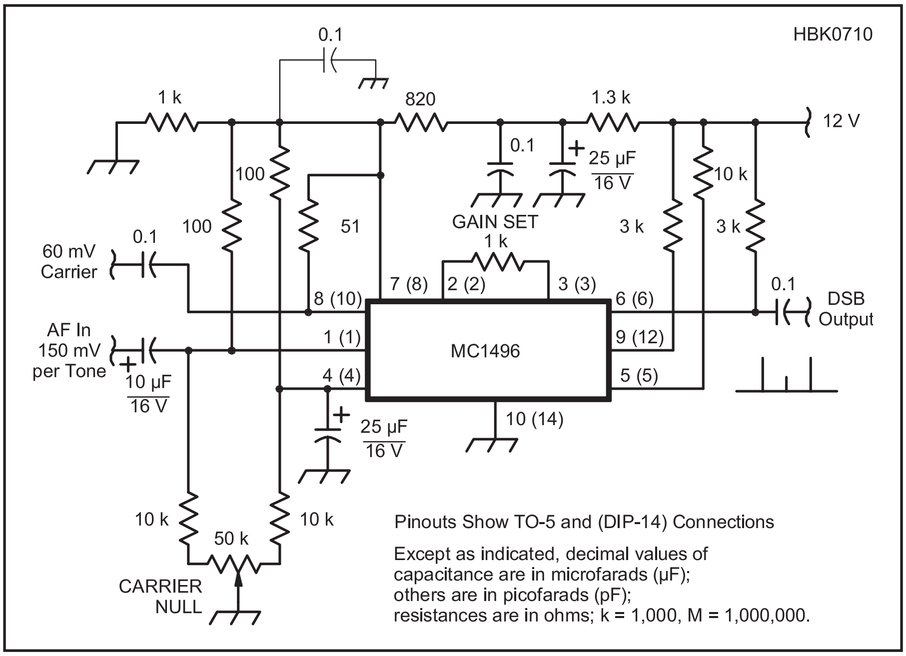


Fig 13.22 — An IC balanced modulator circuit using the MC1496 IC. The resistor between pins 2 and 3 sets the subsystem gain.

can be selected. In synthesized radios the programming of the microprocessor controls the various LOs. Some synthesized radios use two IF filters at two different frequencies, one for USB and one for LSB, and a 455.0 kHz LO, as shown in Fig 13.18. These radios can be designed to transmit two independent sidebands (ISB) resulting in two separate channels in the spectrum space of the usual AM channel.

In times past, balanced modulators using diodes, balancing potentiometers and numerous components were used. These days it doesn't make sense to use this approach. ICs and packaged diode mixers do a much better job and are less expensive. The most widely known balanced modulator IC, the MC1496, has been around for more than 25 years and is still one of the best and least expensive. **Fig 13.22** is a typical balanced modulator circuit using the MC1496.

The data sheets for balanced modulators and mixers specify the maximum level of audio for a given LO level. Higher audio levels create excessive IMD. The IF filter following the modulator removes higher-order IMD products that are outside its passband but the in-band IMD products should be at least 40 dB below each of two equal test tones. Speech clipping (AF or IF) can degrade this to 10 dB or so, but in the absence of speech processing the signal should be clean, in-band.

IF FILTERS

The desired IF filter response is shown in **Fig 13.23A**. The reduction of the carrier frequency is augmented by the filter response. It is common to specify that the filter response be down 20 dB at the carrier frequency. Rejection of the opposite sideband should (hopefully) be 60 dB, starting at 300 Hz below the carrier frequency, which is the 300-Hz point on the opposite sideband. The ultimate attenuation should be at least 70 dB. This would represent a very good specification for a high quality transmitter. The filter passband should be as flat as possible (with passband ripple less than 1 dB or so).

Special filters, designated as USB or LSB, are designed with a steeper roll-off on the

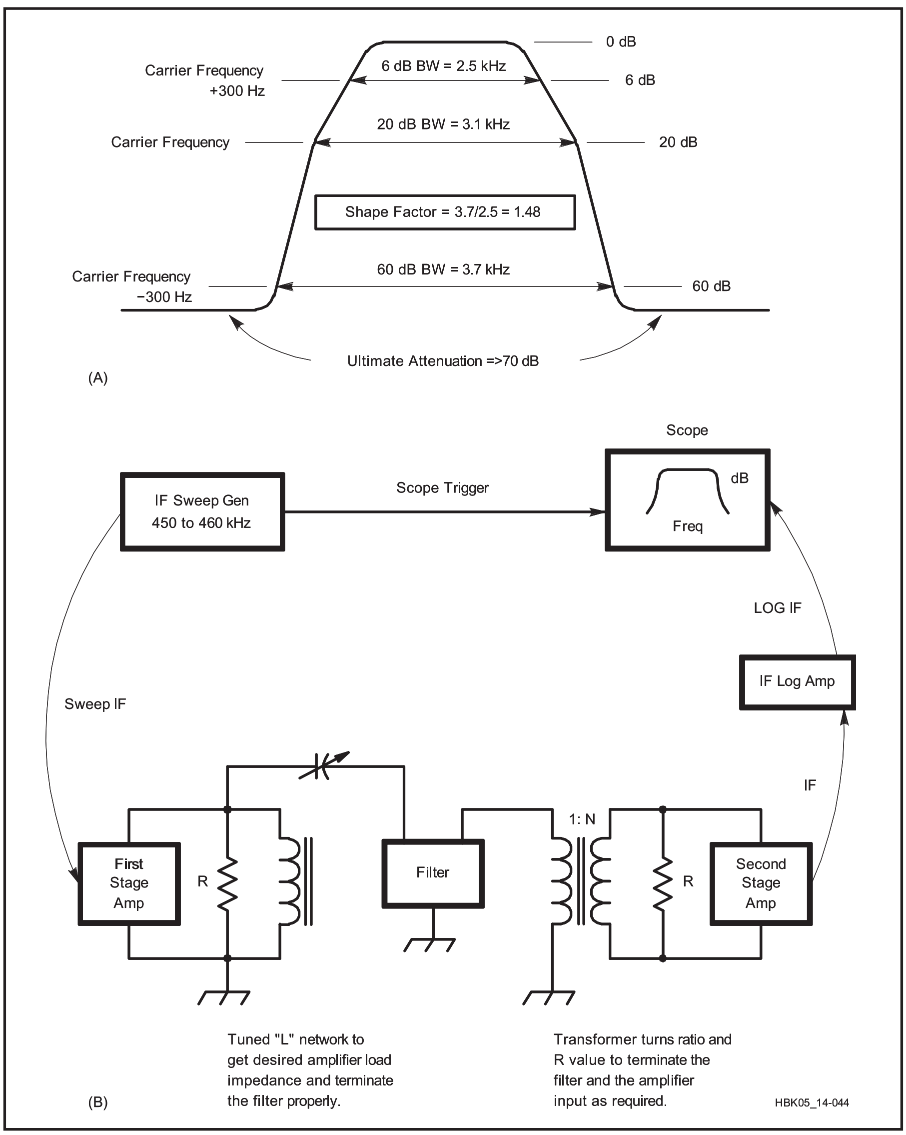


Fig 13.23 — At (A), desired response of a SSB IF filter. At (B), one method of terminating a mechanical filter that allows easy and accurate tuning adjustment and also a possible test setup for performing the adjustments.

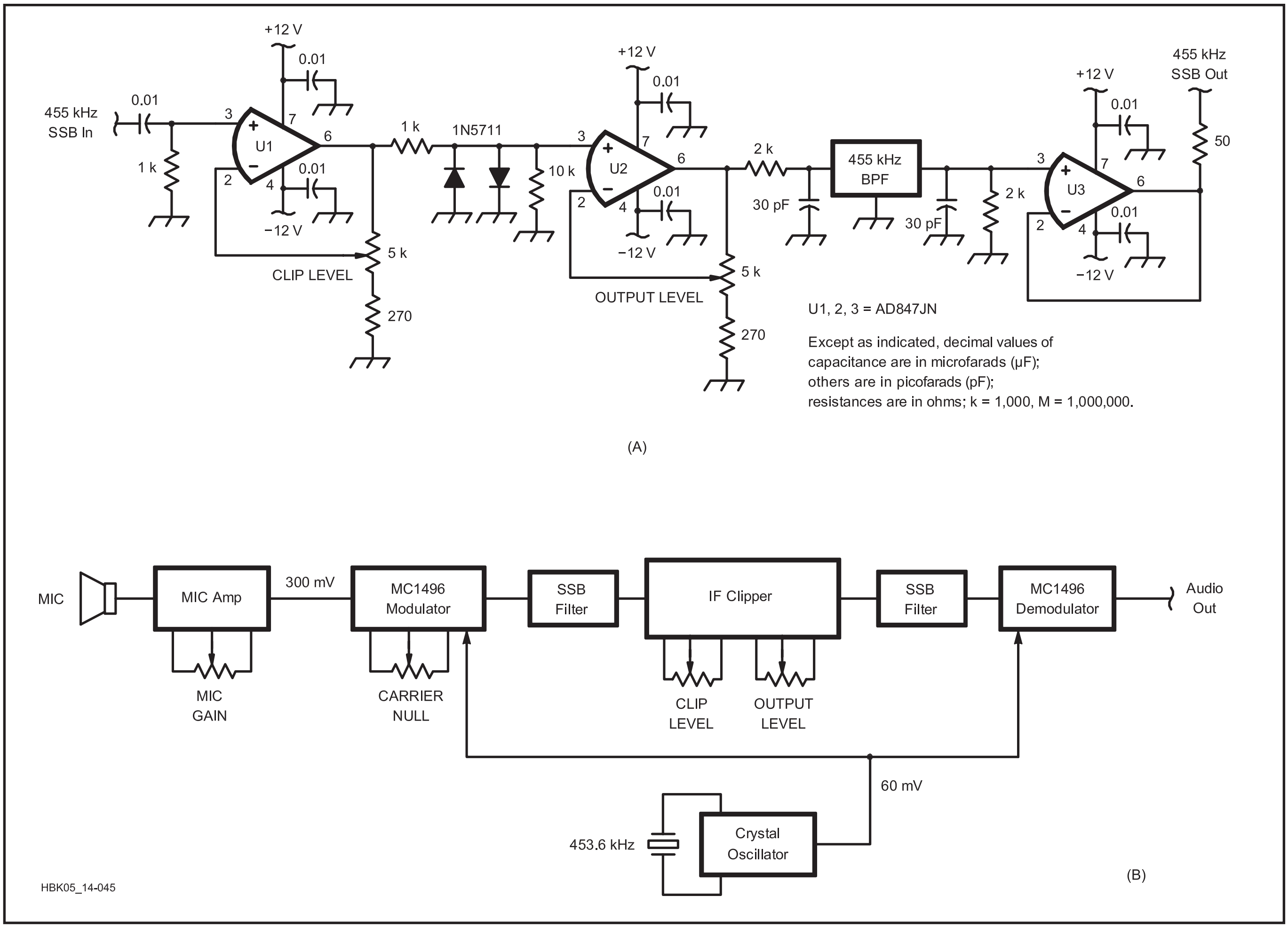


Fig 13.24 — IF speech clipping. At (A), schematic diagram of a 455 kHz IF clipper using high-frequency op amps. At (B) block diagram of an adaptation of the above system to an audio in-audio out configuration.

carrier frequency side, in order to improve rejection of the carrier and opposite sideband. Mechanical filters are available that do this. Crystal-ladder filters (see the **RF and AF Filters** chapter) are frequently called "single-sideband" filters because they also have this property. The steep skirt can be on the low side or the high side, depending on whether the crystals are across the signal path or in series with the signal path, respectively.

Filters require special attention to their terminations. The networks that interface the filter with surrounding circuits should be accurate and stable over temperature. They should be easy to adjust. One very good way to adjust them is to build a narrow-band sweep generator and look at the output IF envelope with a logarithmic amplifier, as indicated in Fig 13.23B. There are three goals:

- The driver stage must see the desired load impedance.
- The stage after the filter must see the desired source (generator) impedance.
- The filter must be properly terminated at both ends.

Lack of any of these conditions will result in loss of specified filter response. Fig 13.23B shows two typical approaches. This kind of setup is a very good way to make sure the filters and other circuitry are working properly.

Finally, overdriven filters (such as crystal or mechanical filters) can become nonlinear and generate distortion. Thus it is necessary to stay within the manufacturer's specifications. Magnetic core materials used in the tuning networks must be sufficiently linear at the signal levels encountered. They should be tested for IMD separately.

IF SPEECH CLIPPER

Audio clipper speech processors generate a considerable amount of in-band harmonics and IMD (involving different simultaneously occurring speech frequencies). The total distortion detracts somewhat from speech intelligibility. Other problems were mentioned in the earlier section on speech processing. IF clippers overcome most of these problems, especially the Hilbert Transform problem. (See Sabin and Schoenike in the References section.)

Fig 13.24A is a schematic diagram of a 455 kHz IF clipper using high-frequency op-amps. 20 dB of gain precedes the diode clippers. A second amplifier establishes the desired output level. The clipping produces a wide band of IMD products close to the IF frequency. Harmonics of the IF frequency are easily rejected by subsequent selectivity. "Close-in" IMD distortion products are band-limited by the 2.5 kHz wide IF filter so that out-of-band splatter is eliminated. The in-band IMD products are at least 10 dB below the speech tones.

Fig 13.24B shows a block diagram of an adaptation of the above system to an audio in-audio out configuration that can be inserted into the mic input of any transmitter to provide the benefits of RF speech processing. These are sometimes offered as aftermarket accessories.

Fig 13.25 shows oscilloscope pictures of an IF clipped two-tone signal at various levels of clipping. The level of clipping in a radio can be estimated by comparing with these photos.

Listening tests verify that the IMD does not sound nearly as bad as harmonic distortion. In fact, processed speech sounds relatively clean and crisp. Tests also verify that speech intelligibility in a noise background is improved by 8 dB. (See the article on RF clippers by Sabin in the References section.)

The repeaking effect from band-pass filtering the clipped IF signal occurs, and must be accounted for when adjusting the output level. A two-tone audio test signal or a speech signal should be used. The ALC circuitry (discussed later) will reduce the IF gain to prevent splattering in the power amplifiers. If the IF filter is of high quality and if subsequent amplifiers are clean, the transmitted signal is of very high quality and is very effective in noisy situations and often also in pile-ups.

The extra IF gain implies that the IF signal entering the clipper must be free of noise, hum and spurious products. The cleanup filter also helps reduce the carrier frequency, which is outside the passband.

An electrically identical approach to the IF clipper can be achieved at audio frequencies. If the audio signal is translated to, say 455 kHz, processed as described and translated back to audio, all the desirable effects of IF clipping are retained. This output then plugs into the transmitter's microphone jack. Fig 13.24B shows the basic method. The mic amplifier and the MC1496 circuits have been previously shown and the clipper circuit can be the same as in Fig 13.24A.

The interesting operating principle in all of these examples is that the characteristics of the IF clipped (or equivalent) speech signal do not change during frequency translation, even if translated down to audio and then back up to IF in a balanced modulator.

IF Linearity and Noise

Fig 13.18 indicates that after the last SSB filter, whether it is just after the SSB modulator or after the IF clipper, subsequent BPFs are considerably wider. For example, the 70 MHz crystal filter may be 15 to 30 kHz wide. This means that there is a "gray region" in the transmitter in which out-of-band IMD that is generated in the IF amplifiers and mixers can cause adjacent-channel interference.

A possible exception, not shown in Fig 13.18, is that there may be an intermediate IF in the 10 MHz region that also contains a narrow filter.

The implication is that special attention must be paid to the linearity of these circuits. It's the designer's job to make sure that distortion in this gray area is much less than distortion generated by the PA and also less than the phase noise generated by the final mixer. Recall also that the total IMD generated in the exciter stages is the result of several amplifier and mixer stages in cascade; therefore, each element in the chain must have at least 40 to 50 dB IMD quality. The various drive levels should be chosen to guarantee this. This requirement for multistage linearity is one of the main technical and cost burdens of the SSB mode.

Of interest also in the gray region are additive white, thermal and excess noises origi-

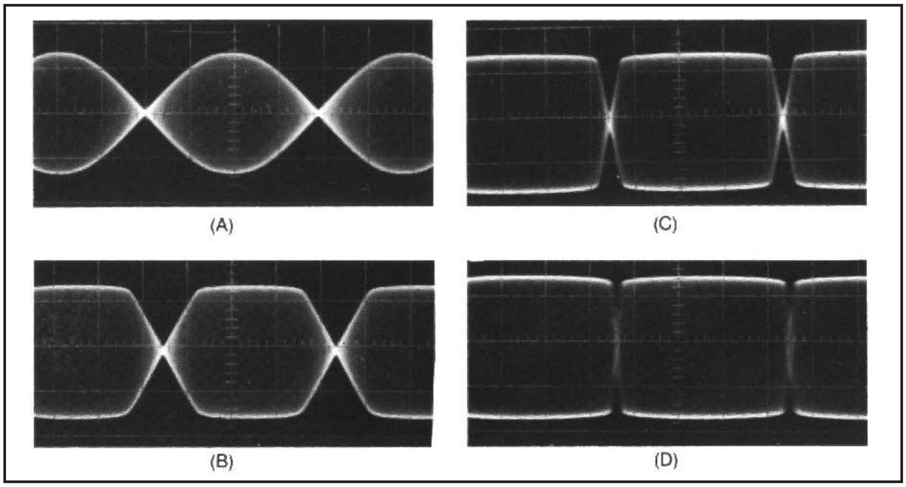


Fig 13.25 — Two-tone envelope patterns with various degrees of RF clipping. All envelope patterns are formed using tones of 600 and 1000 Hz. At A, clipping threshold; B, 5 dB of clipping; C, 10 dB of clipping; D, 15 dB of clipping.

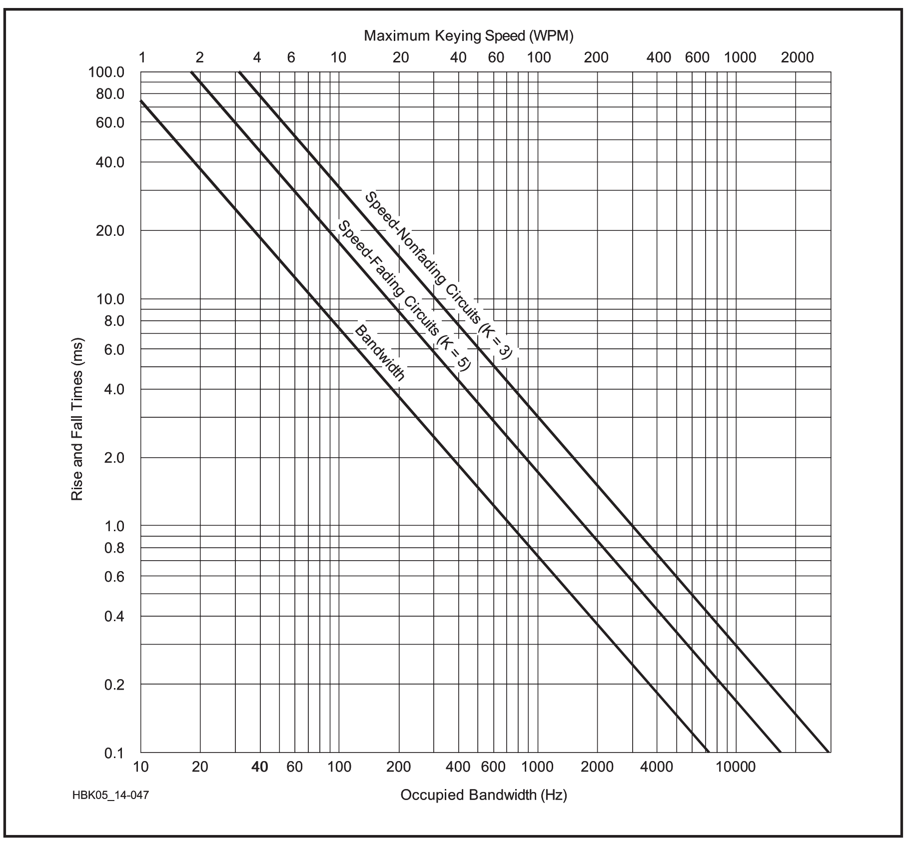


Fig 13.26 — Keying speed versus rise and fall times versus bandwidth for fading and nonfading communications circuits. For example, for transmitter output waveform rise and fall times of approximately 6 ms, draw a horizontal line from 6.0 ms on the rise and fall times scale to the bandwidth line. Then draw a vertical line to the occupied bandwidth scale at the bottom of the graph. In this case the bandwidth is about 130 Hz. Also extend the 6.0 ms horizontal line to the K = 3 line for a nonfading circuit. Finally draw a vertical line from the K = 3 line to the WPM axis. The 6 ms rise and fall time should be suitable for keying speeds up to about 50 WPM in this example.

nating in the first IF amplifier after the SSB filter and highly magnified on their way to the output. This noise can be comparable to the phase noise level if the phase noise is low, as it is in a high-quality radio. Recall also that phase noise is at its worst on modulation peaks, but additive noise may be (and often is) present even when there is no modulation. This is a frequent problem in co-located transmitting and receiving environments. Many transmitter designs do not have the benefit of the narrow filter at 70 MHz, so the amplified noise can extend over a much wider frequency range.

13.1.7 CW Operation

Radiotelegraph or CW operation can be easily obtained from the transmitter architecture design shown in Fig 13.18. For CW operation, a carrier is generated at the center of the SSB filter passband. There are two ways to make this carrier available. One way is to unbalance the balanced modulator so that the LO can pass through. Each kind of balanced modulator circuit has its own method of doing this. The approach chosen in Fig 13.18 is to go around the modulator and the SSB filter.

A shaping network controls the envelope of the IF signal to accomplish two things: control the shape of the Morse code character in a way that limits wideband spectrum emissions that can cause interference, and make the Morse code signal easy and pleasant to copy.

RF ENVELOPE SHAPING

On-off keying (CW) is a special kind of low-level amplitude modulation (a low signal-level stage is turned on and off). It is special because the sideband power is subtracted from the carrier power, and not provided by a separate "modulator" circuit, as in high-level AM. It creates a spectrum around the carrier frequency whose amplitude and bandwidth are influenced by the rates of signal amplitude rise and fall and by the curvature of the keyed waveform. Refer to the discussion of keying speed, rise and fall times, and bandwidth in the **Modulation** chapter and the earlier discussion in this chapter for some information about this issue. For additional information see the article by Sabin on IF signal processing in the References section of this chapter.

Now look at **Fig 13.26**. The vertical axis is labeled Rise and Fall Times (ms). For a rise/fall time of 6 ms (between the 10% and 90% values) go horizontally to the line marked Bandwidth. A –20 dB bandwidth of roughly 120 Hz is indicated on the lower horizontal axis. Continuing to the K = 5 and K = 3 lines, the upper horizontal axis suggests code speeds of 30 WPM and 50 WPM respectively.

These code speeds can be accommodated by the rise and fall times displayed on the vertical axis. For code speeds greater than these the Morse code characters become "soft" sounding and difficult to copy, especially under less-than-ideal propagation conditions.

For a narrow spectrum and freedom from adjacent channel interference, a further requirement is that the spectrum must fall off very rapidly beyond the –20 dB bandwidth indicated in Fig 13.26. A sensitive narrowband CW receiver that is tuned to an adjacent channel that is only 1 or 2 kHz away can detect keying sidebands that are 80 to 100 dB below the key-down level of a strong CW signal.

An additional consideration is that during key-up a residual signal, called "backwave,"

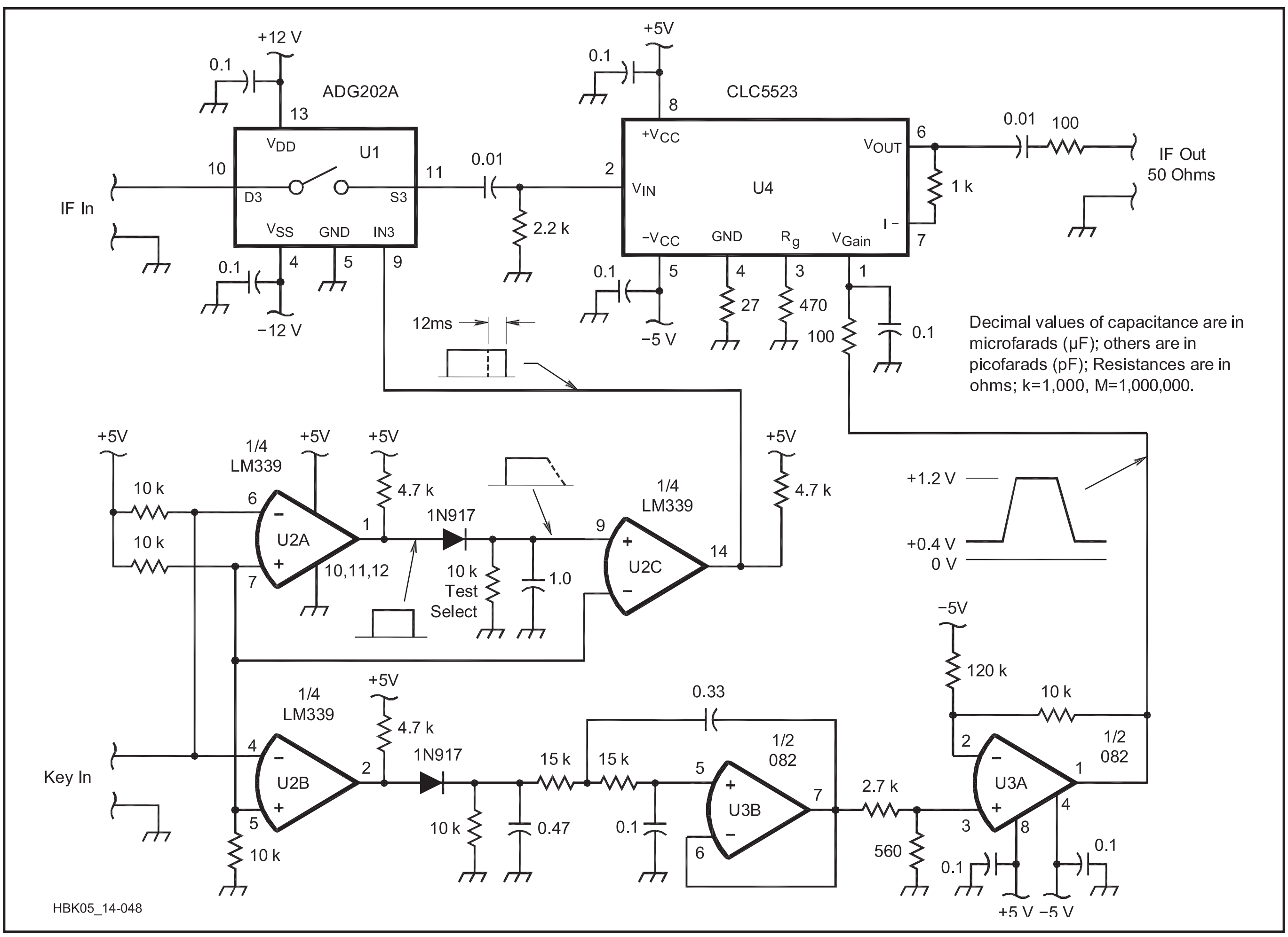


Fig 13.27 — This schematic diagram shows a CW waveshaping and keying circuit suitable for use with an SSB/CW transmitter such as is shown in Fig 13.18.

should not be noticeable in a nearby receiver. A backwave level at least 90 dB below the key-down carrier is a desirable goal.

Microprocessor-controlled transceivers manufactured today control CW keying rise- and fall-time through software. The operator generally accesses the keying shape parameter through a menu selection and adjustment process. Three to four ms is a typical value for most transceivers that balances crisp keying characteristics against excessive off-channel artifacts. See the *QST* Product Reviews for waveforms and discussions of rise- and fall-time settings.

Homebrew equipment usually relies on analog circuitry to control keying waveforms. **Fig 13.27** is the schematic of one waveshaping circuit that has been used successfully. A Sallen-Key third-order op amp low-pass filter (0.1 dB Chebyshev response) shapes the keying waveform, produces the rate of rise and fall and also softens the leading and trailing corners just the right amount. The key closure activates the CMOS switch, U1, which turns on the 455-kHz IF signal. At the key-up time, the input to the wave-shaping filter is turned off, but the IF signal switch remains closed for an additional 12 ms.

The keying waveform is applied to the gain control pin of a CLC5523 amplifier IC. This device, like nearly all gain-control amplifiers, has a *logarithmic* control of gain; therefore some experimental "tweaking" of the capacitor values was used to get the result shown in **Fig 13.28A**. The top trace shows the on/off operation of the IF switch, U1. The signal is turned on shortly before the rise of the keying pulse begins and remains on for about 12 ms after the keying pulse is turned off, so that the waveform falls smoothly to a very low value. The result is an excellent spectrum and an almost complete absence of backwave. Compare this to the factory transmitter waveshapes shown in Figs 13.1 and 13.3. The bottom trace shows the resulting keyed RF output waveshape. It has an excellent spectrum, as verified by critical listening tests. The thumps and clicks that are found in some CW transmitters are virtually absent. The rise and fall intervals have a waveshape that is approximately a cosine. Spread-spectrum frequency-hop waveforms have used this approach to minimize wideband interference.

Fig 13.28B is an accurate *SPICE* simulation of the wave shaping circuit output before the signal is processed by the CLC5523 amplifier. To assist in adjusting the circuit, create a steady stream of 40 ms dots that can be seen on an RF oscilloscope that is looking at the final PA output envelope. It is important to make sure that the excellent waveshape is not degraded on its way to the transmitter output. Single-sideband linear power amplifiers are well suited for a CW transmitter, but they must stay within their linear range, and the backwave problem must be resolved.

When evaluating the spectrum of an incoming CW signal during on-the-air operations, a poor receiver design can contribute problems caused by its vulnerability to a strong but clean adjacent channel signal. Clicks, thumps, front end overload, reciprocal mixing, etc can be created in the receiver. It is important to put the blame where it really belongs.

13.1.8 Wideband Noise

With receiver sensitivity, selectivity, and linearity having reached extraordinary levels of performance, a reduction in transmitted spurious emissions is clearly in the best interests of all amateurs. It does us no good to spend time and effort creating an exceptional receiver if the channel is filled with transmitted noise and distortion products! This is of particular concern when multiple transmitters are located at the same station or facility such as at emergency communications stations, during Field Day and special events, and at multi-transmitter contest stations. (See the article by Grebenkemper in the Reference section.)

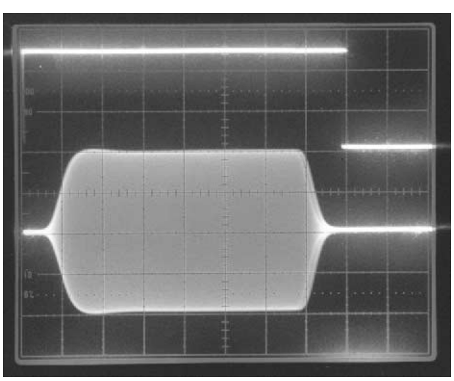
In the block diagram of Fig 13.18, the last mixer and the amplifiers after it are wideband circuits that are limited only by the harmonic filters and by any selectivity that may be in the antenna system. Wide-band phase noise transferred onto the transmitted modulation by the last LO (almost always a synthesizer of some kind) can extend over a wide frequency range, therefore LO cleanliness is always a matter of great concern.

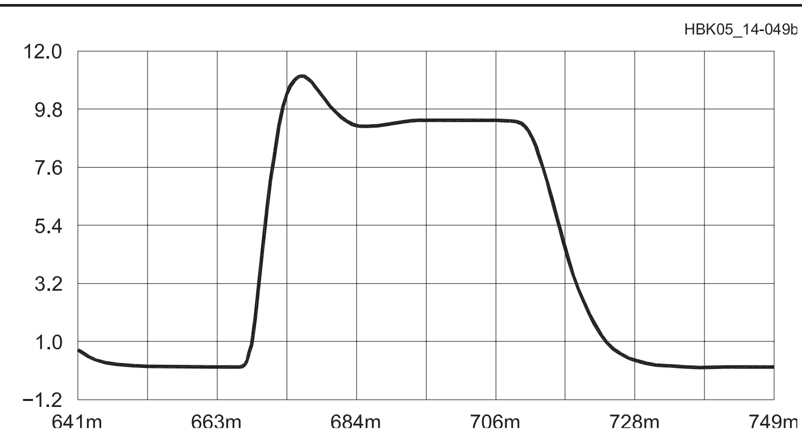
The amplifiers after this mixer are also sources of wide-band "white" or additive noise. This noise can be transmitted even during times when there is no modulation, and it can be a source of local interference. To reduce this noise, use a high-level mixer with as much signal output as possible, and make the noise figure of the first amplifier stage after the mixer as low as possible.

Commercial and military transmitters that are used in close proximity to receivers, such as on ships and aircraft, are always designed to control wideband emissions of both additive noise and phase noise, referred to as "composite" noise.

TRANSMIT MIXER SPURIOUS SIGNALS

The last IF and the last mixer LO in Fig 13.18 are selected so that, as much as possible, harmonic IMD products are far enough away from the operating frequency that they fall outside the passband of the low-


(A)



(B)

Fig 13.28 — At (A) is the oscilloscope display of the CW waveshaping and keying circuit output. The top trace is the IF keying signal applied to S1 of Fig 13.27. The bottom trace is the transmitter output RF spectrum. At (B) is a *SPICE* simulation of the waveshaping network. When this signal is applied to the logarithmic control characteristic of the CLC5523 amplifier, the RF envelope is modified slightly to the form shown in A.

pass filters and are highly attenuated. This is difficult to accomplish over the transmitter's entire frequency range. It helps to use a high-level mixer and a low enough signal level to minimize those products that are unavoidable. Low-order crossovers that cannot be sufficiently reduced are unacceptable, however; the designer must go back to the drawing board.

13.1.9 Automatic Level Control (ALC)

The purpose of ALC is to prevent the various stages in the transmitter from being overdriven. Over-drive can generate too much out-of-band distortion or cause excessive power dissipation, either in the amplifiers or in the power supply. ALC does this by sampling the peak amplitude of the modulation (the envelope variations) of the output signal and then developing a dc gain-control voltage that is applied to an early amplifier stage, as suggested in Fig 13.18.

ALC is usually derived from the last stage in a transmitter. This ensures that this last stage will be protected from overload. However, other stages prior to the last stage may not be as well protected; they may generate excessive distortion. It is possible to derive a composite ALC from more than one stage in a way that would prevent this problem. But designers usually prefer to design earlier stages conservatively enough so that, given a temperature range and component tolerances, the last stage can be the one source of ALC. The gain control is applied to an early stage so that all stages are aided by the gain reduction.

Note that ALC should be minimally active with most digital mode transmissions. The modulation of these signals requires linear amplification to preserve the waveform shape and minimize distortion products. ALC action creates distortion as it alters the power level of the signal. Adjust the radio drive levels so that the ALC is at its minimum level of activity – usually shown as the lower bar of a multi-segment LCD meter or a needle position just above zero. (The same caution applies to any form of audio or speech processing if the digital signal is generated by audio tones applied to the transmitter's microphone input.)

SPEECH PROCESSING WITH ALC

A fast response to the leading edge of the modulation is needed to prevent a transient overload. After a strong peak, the control voltage is "remembered" for some time as the voltage in a capacitor. This voltage then decays partially through a resistor between peaks. An effective practice provides two capacitors and two time constants. One capacitor decays quickly with a time constant of, say 100 ms, the other with a time constant of several seconds. With this arrangement a small amount of speech processing, about 1 or 2 dB, can be obtained. (Explanation: The dB of improvement mentioned has to do with the improvement in speech intelligibility in a random noise background. This improvement is equivalent to what could be achieved if the transmit power were increased that same number of dB.)

The gain rises a little between peaks so that weaker speech components are enhanced. But immediately after a peak it takes a while for the enhancement to take place, so weak components right after a strong peak are not enhanced very much. **Fig 13.29A** shows a complete ALC circuit that performs speech processing.

ALC IN SOLID-STATE POWER AMPLIFIERS

Fig 13.29B shows how a dual directional coupler can be used to provide ALC for a solid-state power amplifier (PA). The basic idea is to protect the PA transistors from excessive SWR and dissipation by monitoring both the forward power and the reflected power.

TRANSMIT GAIN CONTROL (TGC)

This is a widely used feature in commercial and military equipment. A calibrated "tune-up" test carrier of a certain known level is applied to the transmitter. The output carrier level is sampled, using a diode detector. The resulting dc voltage is used to set the gain of a low-level stage. This control voltage is digitized and stored in memory so that it is semi-permanent. A new voltage may be generated and stored after each frequency change, or the stored value may be used. A test signal is also used to do automatic antenna tuning. A dummy load is used to set the level and a low-level signal (a few mW) is used for the antenna tune-up.

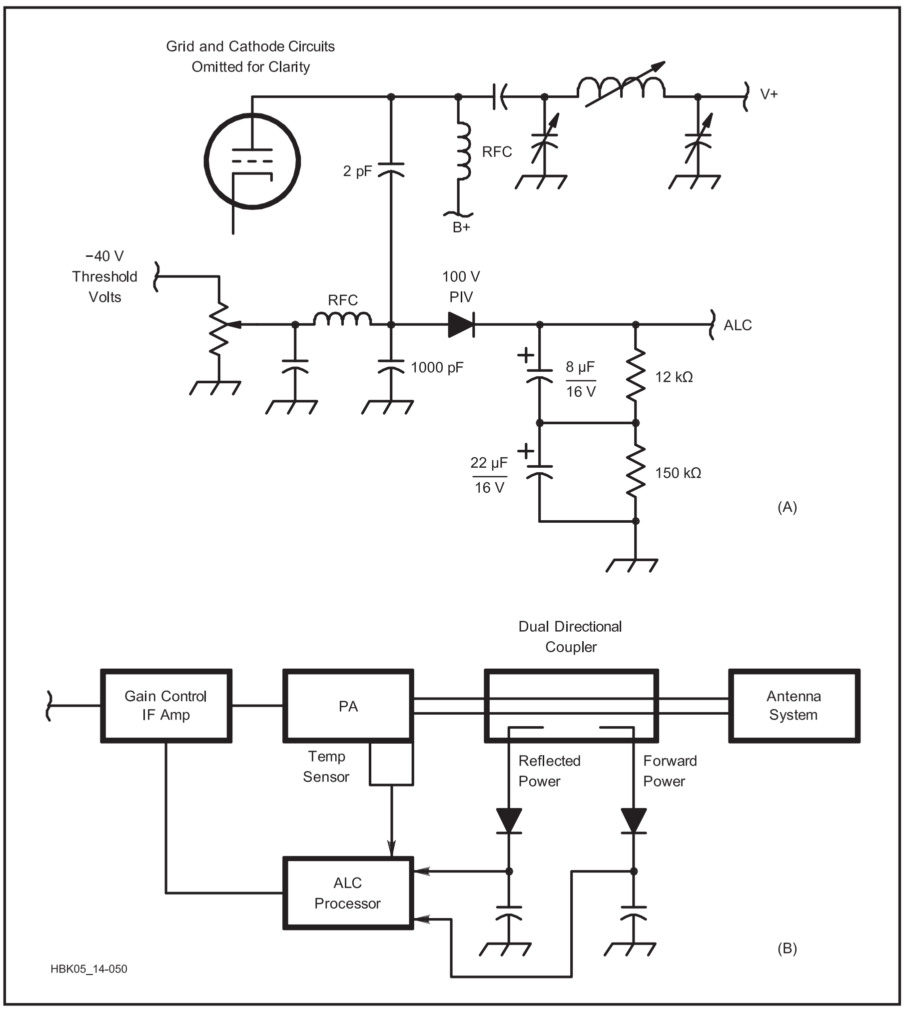


Fig 13.29 — At (A), an ALC circuit with speech processing capability. At (B), protection method for a solid-state transmitter.

13.1.10 *Project:* The MicroT2 — A Compact Single-Band SSB Transmitter

As an example of an SSB transmitter including many aspects of design covered heretofore, we present the MicroT2, shown in **Fig 13.30**, a simple SSB transmitter that generates a high-quality USB or LSB signal on any single band from 1.8 MHz to 50 MHz. Rick Campbell, KK7B, developed the MicroT2 as a companion to the MicroR2 receiver project described in the **Receivers** chapter.

While it is a bit more involved to generate an SSB signal than a CW signal, we greatly simplify the task if all the necessary circuitry is on a single PC board exciter module. Once we have a high-quality low-level SSB signal, a 5 or 500 W SSB transmitter is as easy to build as a 5 or 500 W CW transmitter. Simple transmitters are delightful, but relaxed standards are not. The MicroT2 is designed to be clean, stable and reliable, exceed FCC Part 97 requirements, and sound good, too.

A thorough description of the circuitry in this transmitter can be found in *Experimental Methods in RF Design (EMRFD)* of which the project's designer was a co-author. The complete article for this project, including schematics, parts lists, and adjustment instructions, can be found on the CD-ROM that accompanies this book.

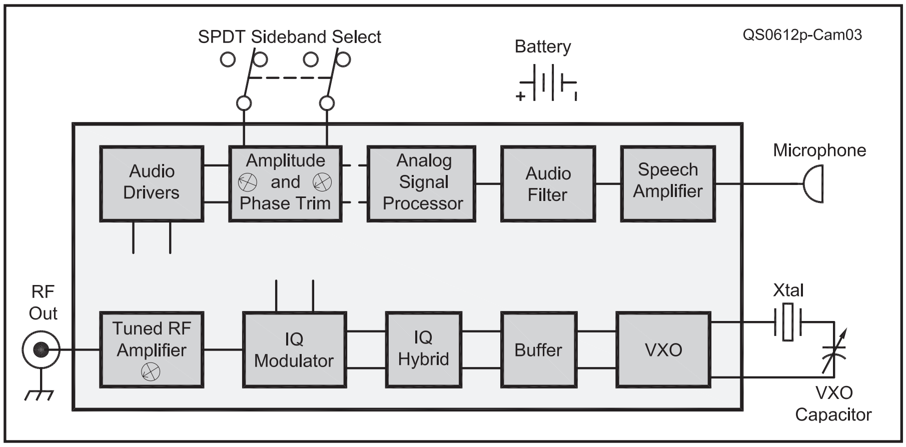


Fig 13.31 — Block diagram of the circuitry on the PC board.

EXCITER BLOCK DIAGRAM

RF Circuitry

Fig 13.31 is the block diagram of the circuitry on the PC board. The exciter uses the phasing method of SSB generation, which makes it easy to operate on different frequencies. In a phasing SSB exciter, two identical signals with a 90° phase difference are generated and then combined so that one sideband adds and the other subtracts. The signal quality from this exciter is not merely adequate for the HF amateur bands — it is exceptional.

The frequency generator consists of a VXO, buffer amplifier and quadrature hybrid. It has a lot of parts: three transistors, a voltage regulator IC, a Zener diode, four toroids, and many resistors and capacitors. There are no adjustments. You just build it and it works. The frequency stability is better than that of most commercial radios, even when portable.

A pair of Mini-Circuits TUF-3 mixers serves as the I and Q balanced modulators. These provide good carrier suppression without adjustment, and reasonable output at a modest distortion level. IM products in the opposite sideband are more than 30 dB down at the exciter output. The aggressive low-pass filtering right at the mixer IF ports prevents wideband noise and harmonic distortion in the audio stages from contributing to the I and Q modulation.

The exciter's output RF amplifier uses a common-gate JFET. The RF amplifier has relatively low gain, good harmonic suppression and a very clean 0 dBm SSB signal at the output. This is an appropriate level to drive a linear amplifier, balanced mixer or transverter. It is also a low enough level that it is easy to adjust the exciter with simple equipment. The exciter output signal meets FCC regulations for direct connection to an antenna for flea power experiments.

The transmitter is completed by the simple two transistor amplifier with 0.5 W PEP output. A seventh order Chebyshev low-pass filter on the output is noncritical and assures a clean signal that easily meets FCC regulations.

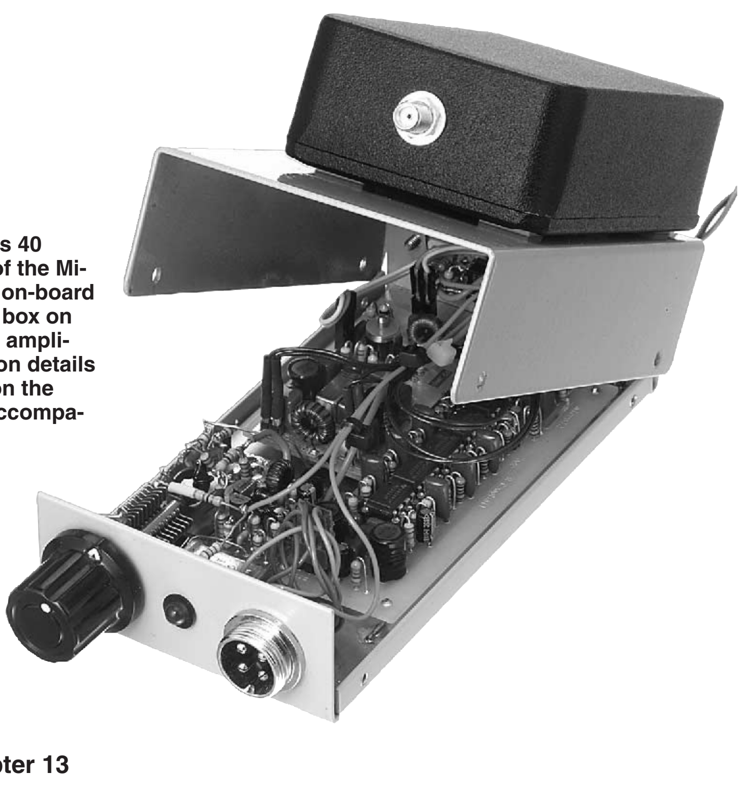


Fig 13.30 — This 40 meter version of the MicroT2 uses the on-board VXO. The black box on top is the 0.5 W amplifier. Construction details may be found on the CD-ROM that accompanies this book.

Audio Section

The speech amplifier drives a passive low-pass filter. The combination of this low-pass filter and the mixer IF port filters limits speech frequencies to just over 3 kHz for natural sounding speech and good spectral purity. There is no ripple in the audio passband.

The position of the sideband select switch in the signal path allows switching without readjusting the amplitude and phase trimmers. For most applications, one sideband will be used exclusively, and the sideband switch may be replaced by a pair of jumpers on the PC board. If that results in the wrong sideband, reverse the connections between the audio driver transistors and mixers.

13.1.11 *Project:* The MkII — An Updated Universal QRP Transmitter

A frequently duplicated project in the now out-of-print book *Solid State Design for the Radio Amateur* (see References) was a universal QRP transmitter. This was a simple two-stage, crystal-controlled, single-band circuit with an output of about 1.5 W. The no-frills design used manual transmit-receive (TR) switching. It operated on a single frequency with no provision for frequency shift. The simplicity prompted many builders to pick this QRP rig as a first solid state project.

Wes Hayward, W7ZOI, updated the design to the three-stage MKII (**Fig 13.32** and **Fig 13.33**), develops an output of 4 W on any single band within the HF spectrum, if provided with 12 V dc. Q1 is a crystal controlled oscillator that functions with either fundamental or overtone mode crystals. It operates at relatively low power to minimize stress to some of the miniature crystals now available. The three stage design — two driver stages and a power amplifier — provides an easy way to obtain very clean keying.

The complete article for this project, including schematics, parts lists, and adjustment instructions, can be found on the CD-ROM that accompanies this book.

Fig 13.32 — The MKII QRP transmitter includes VXO frequency control, TR switching and a sidetone generator. Construction details may be found on the CD-ROM that accompanies this book.

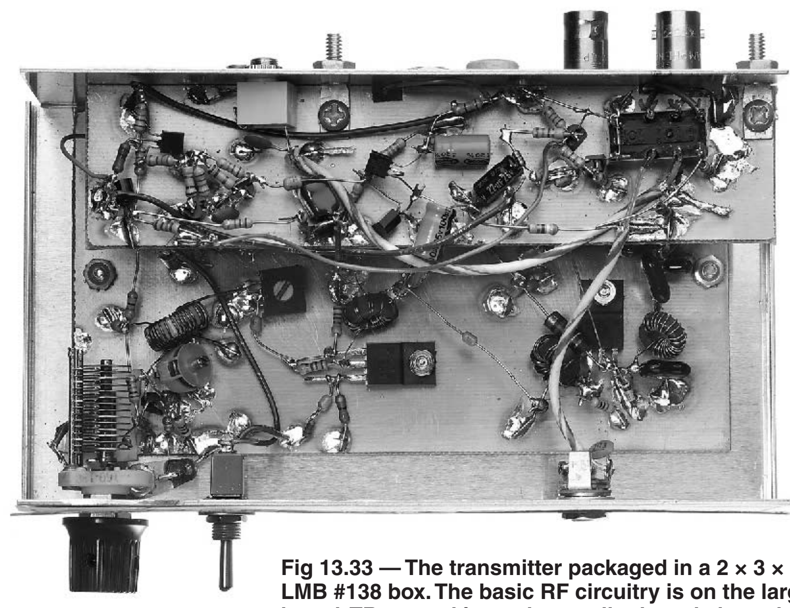


Fig 13.33 — The transmitter packaged in a 2 × 3 × 6 inch LMB #138 box. The basic RF circuitry is on the larger board. TR control is on the smaller board along the top.

13.2 VHF Signal Sources

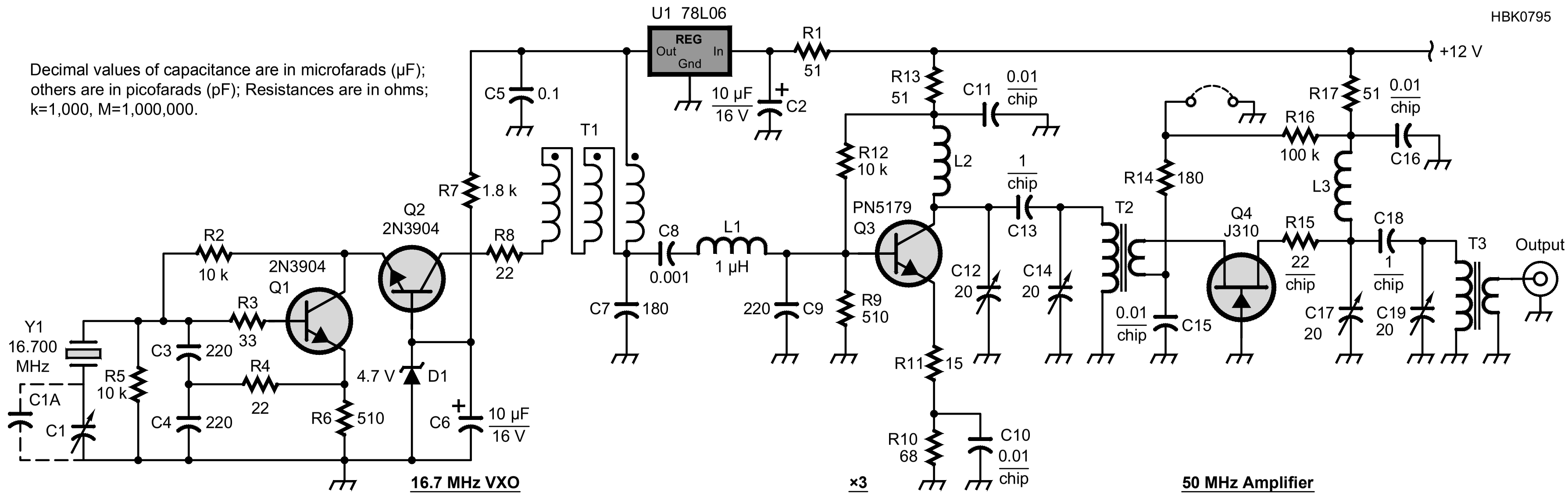
The following section was written by Rick Campbell, KK7B, both as a technical description of circuits to produce exciter-level signals at VHF and a design tutorial for students learning about RF electronics. In both senses, the circuits are useful to amateurs who are interested in building equipment at 50 MHz and above. Amateurs may find it sufficient to use the component values published here to build either a 50 or 144 MHz exciter. The interested reader-student can adapt the circuits for any frequency between 21 and 150 MHz. An additional article with more information on the design of these sources, "VHF Open Sources" by KK7B is available on this book's CD-ROM.

Along with these general-purpose signal sources, a *QST* article by Mark Spencer, WA8SME, describing a standalone 2 meter transmitter to support direction-finding and fox-hunting activities is provided on this book's accompanying CD-ROM.

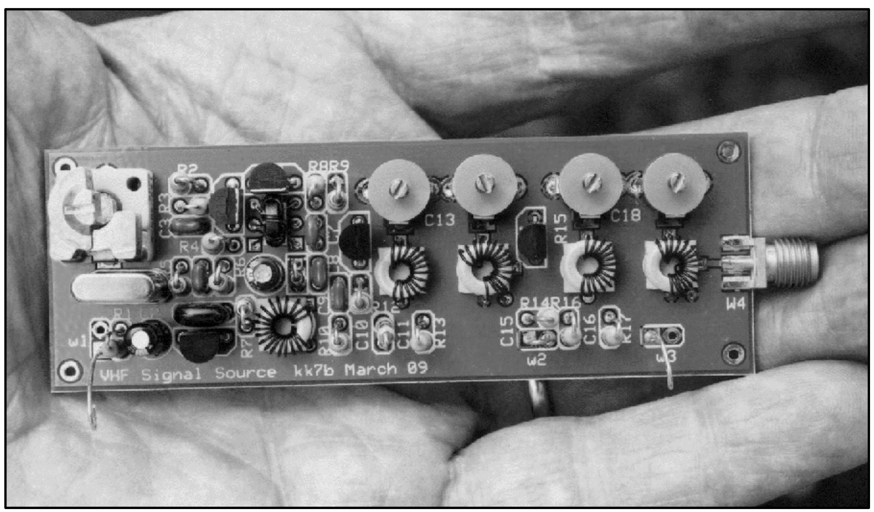
13.2.1 Overview

From 2005 through 2008, Portland State University RF Design students analyzed, simulated, built, tested, and modified beginner's transmitter and simple receiver circuits for 7 MHz. While these basic oscillator-amplifier transmitter and direct conversion receiver circuits have a number of excellent properties as both basic radios and teaching tools, they have several significant weaknesses. Many students successfully designed and built impressive small CW rigs and tested them in the lab, but never connected them to full size antennas or made contacts on the air: 40 meter antennas are too large for typical student quarters and the Technician license is not geared toward entry-level operating on 40 meter CW. Another weakness for classroom use is that many of the critically important device parasitic, lead length, and circuit transmission line challenges for 21st century engineering students may be ignored at 7 MHz.

In 2009 a new VHF signal source was developed for fundamental study that would open the 50 MHz band to student experimenters. Wire antennas for 6 meters are simple, effective, and portable, and the 6 meter band is wide open to Technician licensees. During the project development, it was determined experimentally that the same circuit board could be used on 144 MHz. Engineering students learn to design all the components for any multiple and output frequency between 21 and 150 MHz.



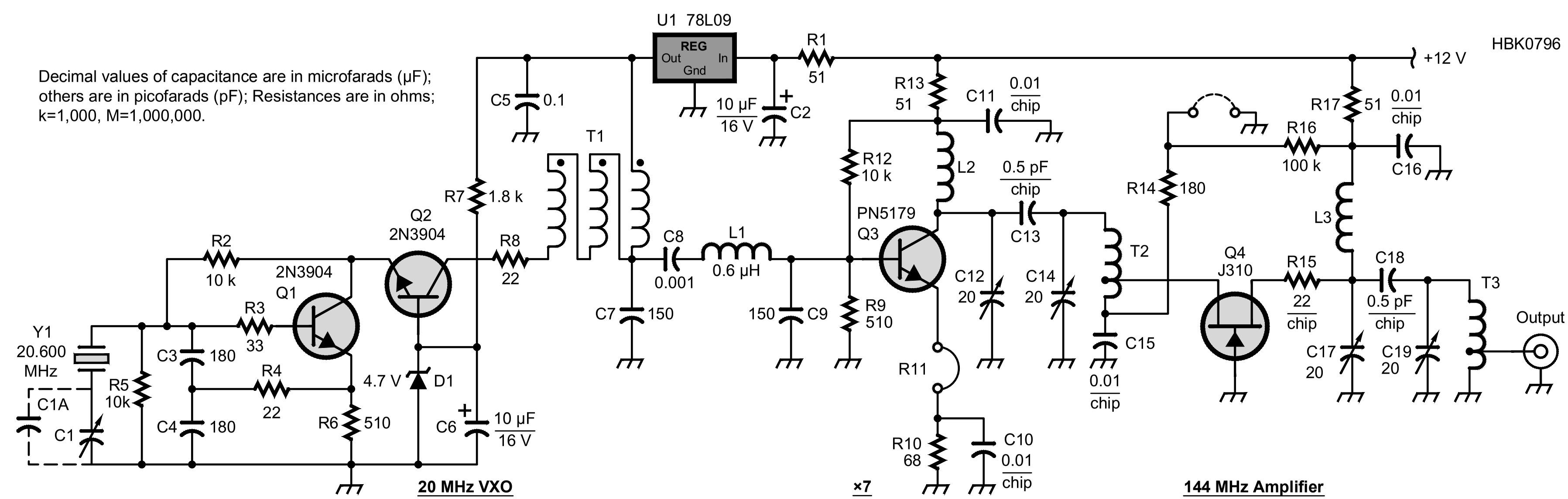
(A)



(B)

Fig 13.34 — (A) Schematic of the 6 meter +10 dBm signal source. Resistors are ¼ W except as noted. The output lowers gracefully to +7 dBm at 9 V with no change in frequency. Below 8 V, the 78L06 regulator drops out. (B) Photograph of the prototype 6 meter signal source from Kanga US (www.kangaus.net). A parts placement diagram for the kit PC board is available on this book's CD-ROM.

C1 — 15 pF PC mount air variable
C2,C6 — 10 µF or 6.8 µF 16 V electrolytic
C3,C4, C9 — 220 pF NP0 miniature disk ceramic
C5 — 100 nF poly
C7 — 180 pF miniature disk ceramic
C8 — 1000 pF miniature disk ceramic
C10, C11, C15, C16 — 10 nF 1206 chip ceramic
C12, C14, C17, C19 — 20 pF green film trimmer
C13, C18 — 1 pF 0806 chip ceramic
D1 — 4.7 V Zener
L1 — 20t #28 AWG on T25-6 toroid core (approx 1 µH)
L2, L3 — 14t #28 AWG on T25-6 toroid core
Q1, Q2 — 2N3904
Q3 — MPS 5179
Q4 — J310
R1, R8, R13, R15, R17 — 51 Ω
R2, R5, R12 — 10 kΩ
R3 — 33 Ω
R4 — 22 Ω
R6, R9 — 510 Ω
R7 — 1.8 kΩ
R10 — 68 Ω
R11 — 15 Ω
R14 — 180 Ω
R15 — 22 Ω 1206 chip
R16 — 100 kΩ
T1 — 6t trifilar on FT23-43 toroid core
T2, T3 — 14t primary, 2t secondary, #28 AWG on T25-6 toroid core
U1 — 78L06
Y1 — 16.700 MHz crystal



(A)

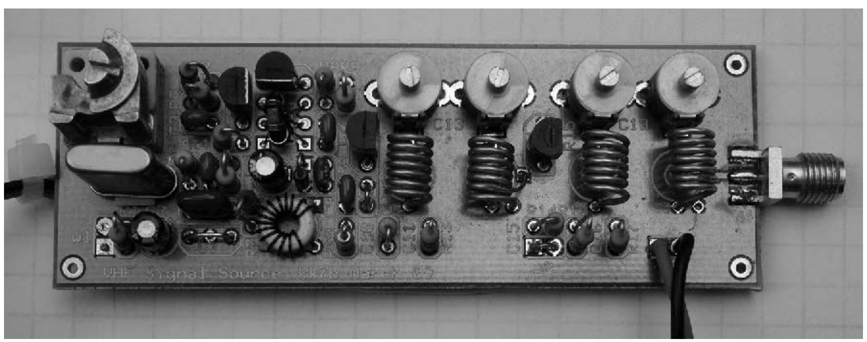

(B)

Fig 13.35 — (A) Schematic of the 2 meter +10 dBm signal source. Resistors are ¼ W except as noted. The output lowers gracefully to +8 dBm at 11 V with no change in frequency. Below 11 V, the 78L09 regulator drops out. (B) Photograph of the prototype 2 meter signal source from Kanga US (www.kangaus.net). A parts placement diagram for the kit PC board is available on this book's CD-ROM.

C1 — 15 pF PC mount air variable
C2,C6 — 10 µF, 16 V electrolytic
C3,C4 — 180 pF NP0 miniature disk ceramic
C5 — 100 nF poly
C7, C9 — 180 pF miniature disk ceramic
C8 — 1000 pF miniature disk ceramic
C10, C11, C15, C16 — 10 nF 1206 chip ceramic
C12, C14, C17, C19 — 20 pF green film trimmer
C13, C18 — 0.5 pF 0806 chip ceramic
D1 — 4.7 V Zener
L1 — 16t on T25-6 toroid core (approx 600 nH)
L2, L3 — 8t #24 AWG bare wire 1 cm long using ⅛ inch drill bit as form
Q1, Q2 — 2N3904
Q3 — MPS 5179
Q4 — J310
R1, R8, R13, R15, R17 — 51 Ω
R2, R5, R12 — 10 kΩ
R3 — 33 Ω
R4 — 22 Ω
R6, R9 — 510 Ω
R7 — 1.8 kΩ
R10 — 68 Ω
R11 — jumper
R14 — 180 Ω
R15 — 22 Ω 1206 chip
R16 — 100 kΩ
T1 — 6t trifilar on FT23-43 toroid core
T2, T3 — 8t #24 AWG bare wire 1 cm long using 1/8 inch drill bit as form; tap 1 turn from cold end
U1 — 78L09
Y1 — 20.600 MHz crystal

The nominal +10 dBm output power is ideal for driving receive mixers, modulators, power amplifiers, or antennas for low power experiments. The first prototype for the 2009 class is shown in **Fig 13.34A**, and the commercial 6 meter circuit board kit from Kanga (**www.kangaus.net**) is shown in Fig 13.34B. The 2 meter version is shown in **Fig 13.35** — note the use of small solenoid inductors wound with bare wire over a 0.1 inch form. (You may note minor component value differences between the schematics and the kit values from Kanga. This does not greatly affect performance and experimentation with the various values is encouraged. Both versions will work fine.)

The signal source may be used as a transmitter at the 10 mW level, connected directly to a dipole antenna. In 2010, RF design student teams with at least one Technician licensee built sources in pairs, along with direct conversion receivers and antennas, and carried them into the field for one-way CW communications. Best reported DX so far is 1.2 miles between dipoles, with no additional gain on receive and 10 mW output.

Any frequency between 24 and 175 MHz may be obtained by adjusting component values as noted in the schematics. For lower order N multipliers, use the 50 MHz emitter components in the frequency multiplier stage, and for N > 3 use the 2 meter components.

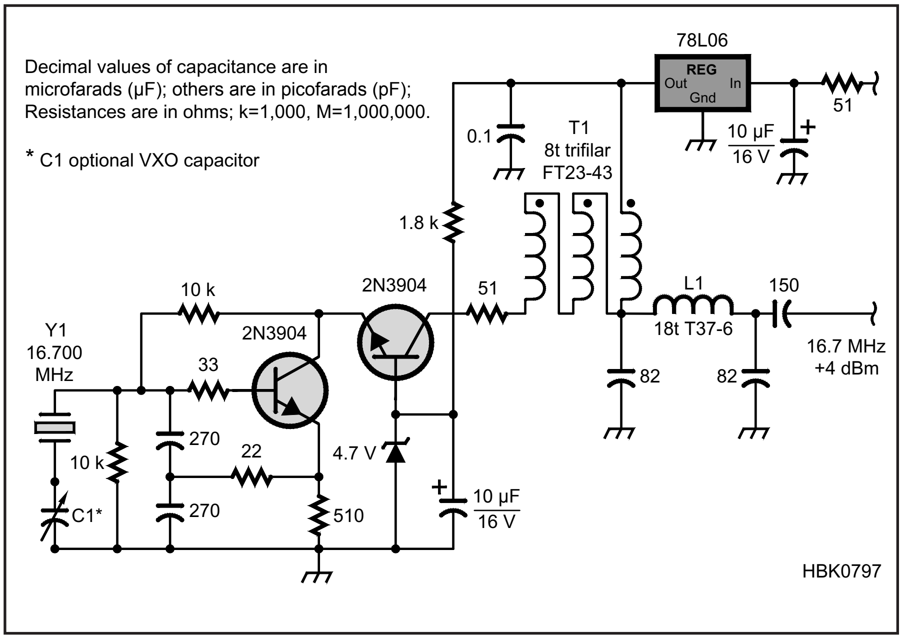


Fig 13.36 — Schematic for the oscillator-buffer circuit.

13.2.2 Oscillator-Buffer

The oscillator-buffer circuit shown in **Fig 13.36** is conventional, but with a number of subtle and elegant details. Q1 and associated resistors and capacitors form a Colpitts oscillator at the parallel resonant fundamental frequency of the crystal. The trimmer capacitor in series with the crystal allows the frequency to be adjusted over a few kHz. The low impedance load on the collector and the series resistors on the base and emitter suppress high frequency parasitic oscillations and reduce the impact of transistor variation on the circuit.

Buffer Q2 supplies a nearly constant 4 V at 2 mA dc to the collector of Q1, while loading the collector with a nearly constant RF impedance on the order of 10 Ω. The 4.7 V Zener on the base of Q2 sets the emitter voltage, and the capacitance of the Zener diode bypasses the base to ground at RF. Zener noise is bypassed to ground by the 10 µF electrolytic capacitor. The resistor from the base to the power supply should be adjusted for one or two mA current through the Zener. 1.5 kΩ is acceptable with a 6 volt regulated supply, and 3.3 kΩ works with the 9 V regulator used in the 144 MHz signal source.

For ultimate stability with a high quality temperature stabilized crystal, the Zener current could be experimentally optimized to the inflection point between Zener and avalanche breakdown, but this is not necessary for room temperature operation of a normal crystal. The temperature of the crystal itself has a much larger impact on frequency drift and overall stability than variations in transistor parameters over temperature.

The output level of the oscillator-buffer circuit is determined by the output network and supply voltage. The peak-to-peak output signal level at the collector is somewhat less than the supply voltage, and the trifilar transformer and output pi-network step it down by a factor of 4. With a 9 volt supply, the output power is about 4 or 5 milliwatts into a 50 Ω load. The pi-network reduces harmonic content, so the output voltage appears sinusoidal on the oscilloscope.

Long- and short-term frequency stability are determined primarily by the crystal temperature and the quality of the series trimmer capacitor. For high stability applications the crystal should be wrapped in packing foam, and the trimmer capacitor should be an air variable. Placing the whole circuit in a temperature controlled environment isolated from any variable sources of heat can reduce oscillator drift to a less than a few Hz per hour.

13.2.3 Frequency Multiplier

The second block in the frequency source is a frequency multiplier. Mathematically its function is simple: for an input sine wave at frequency f the output is a sine wave at integer multiple N times f. In practice the frequency multiplier serves several other important functions, including isolation of the oscillator from the local electronic and electromagnetic environment at the output frequency.

Oscillator-multiplier frequency sources typically need less shielding than oscillators operating directly on the desired output frequency. This is of historical significance in transmitters built on wooden breadboards, and currently important in direct conversion receivers and exciters. Frequency multipliers with N less than 10 are conveniently built with single transistors and diodes. For multipliers with N greater than 10, phase locked loops (PLLs) are commonly used. PLLs are covered in the **Oscillators and Synthesizers** chapter.

Transistor frequency multipliers offer a fascinating introduction to waveform engineering. A simplified view treats the transistor as a non-linear device that generates a distorted waveform rich in harmonics, followed by a filter that selects the desired output. This simple explanation provides the right numbers for the block diagram, but doesn't explain the high efficiency or spectral purity obtained from a carefully designed frequency multiplier.

A deeper understanding considers the output tuned circuit and transistor as a coupled unit, mathematically equivalent to the piston and flywheel of an engine. The flywheel spins at a relatively constant speed, and the piston provides short power strokes over a small angle of the total rotation. The heavier the flywheel, the less the rotation speed varies as short power strokes are applied. In a single cylinder four-stroke internal combustion engine, the flywheel rotates twice for each power stroke — a frequency multiplication of 2. But the analogy goes further. When the piston is not providing power, the cylinder valves are open, and the flywheel easily drives all the mechanical parts around until it's time for the next compression and ignition stroke.

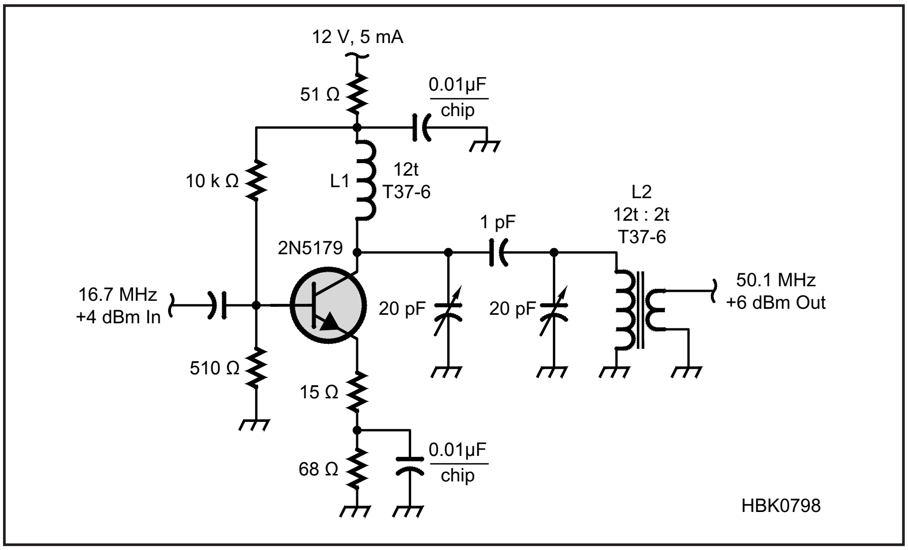


Fig 13.37 — Schematic for the 50 MHz × 3 multiplier circuit.

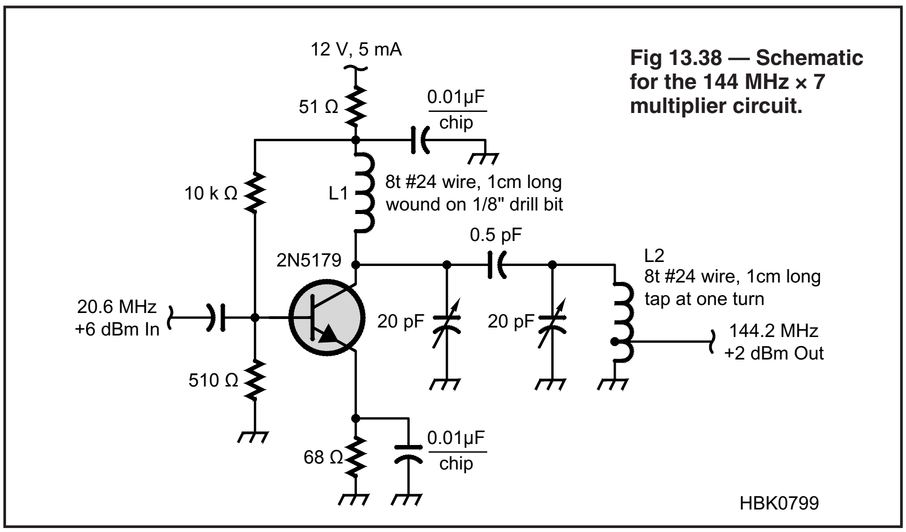


Fig 13.38 — Schematic for the 144 MHz × 7 multiplier circuit.

With a heavy flywheel and low friction, little energy is lost between power strokes.

A high-Q output circuit performs the flywheel function in a frequency multiplier. Calculating and controlling the optimum shape and timing of an engine power stroke are analogous to waveform engineering in a high efficiency Mode J or Class E power amplifier. The difference between a high efficiency power amplifier and a frequency multiplier is simply the number of power strokes per output cycle.

The frequency multiplier in **Fig 13.37** is driven by a sine wave, and the collector load is provided by the high Q tuned circuit on the output. Energy is coupled from the output tuned circuit to a second resonator through a very small capacitor. The collector current and voltage waveforms (power stroke) are determined by the collector load, drive level, and gain of the stage. Constant drive level is provided by the voltage-regulated oscillator-buffer circuit. Gain is adjusted for different harmonic numbers by changing the ac emitter resistance — note the difference between the 50 MHz times 3 and 144 MHz times 7 circuit in **Fig 13.38**. Adjusting a frequency multiplier with a tuned output requires some way of knowing that the resonant load is tuned to the correct harmonic. A spectrum analyzer is ideal, but for decades amateurs have adjusted frequency multipliers by listening on a receiver tuned to desired output frequency with a short wire antenna probe near the stage being tuned.

13.2.4 Output Bandpass Amplifier

The tuned common-gate output amplifier added to the multiplier schematic in **Fig 13.39**

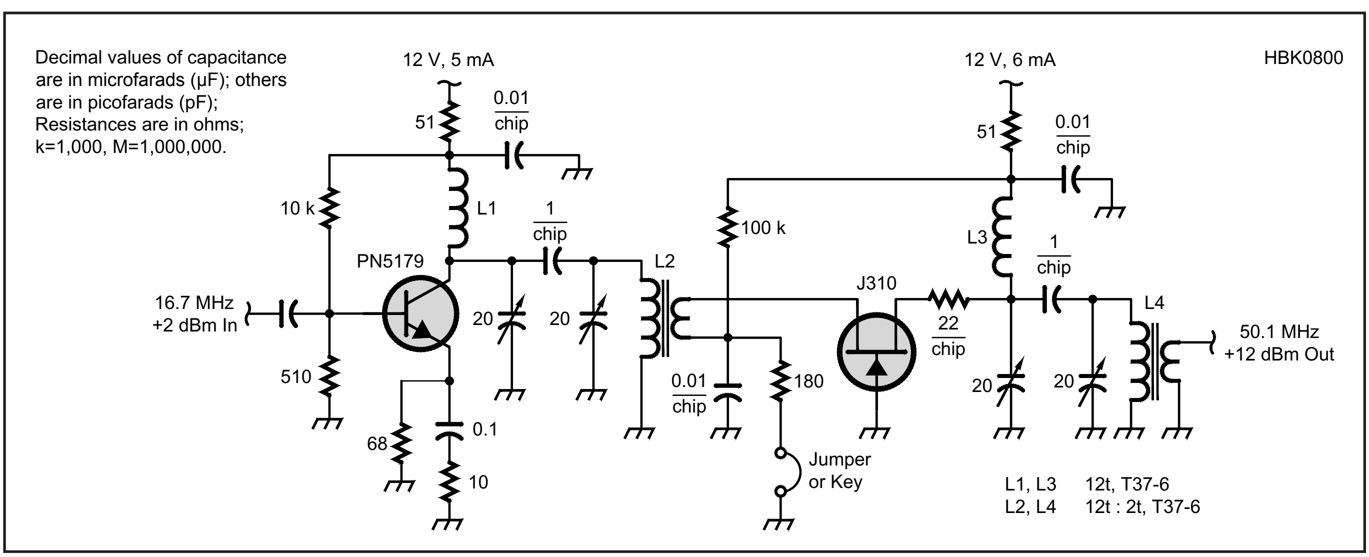


Fig 13.39 — Schematic for the tuned common-gate output amplifier. The multiplier schematic is included to illustrate the double-tuned circuits at the amplifier input and output.

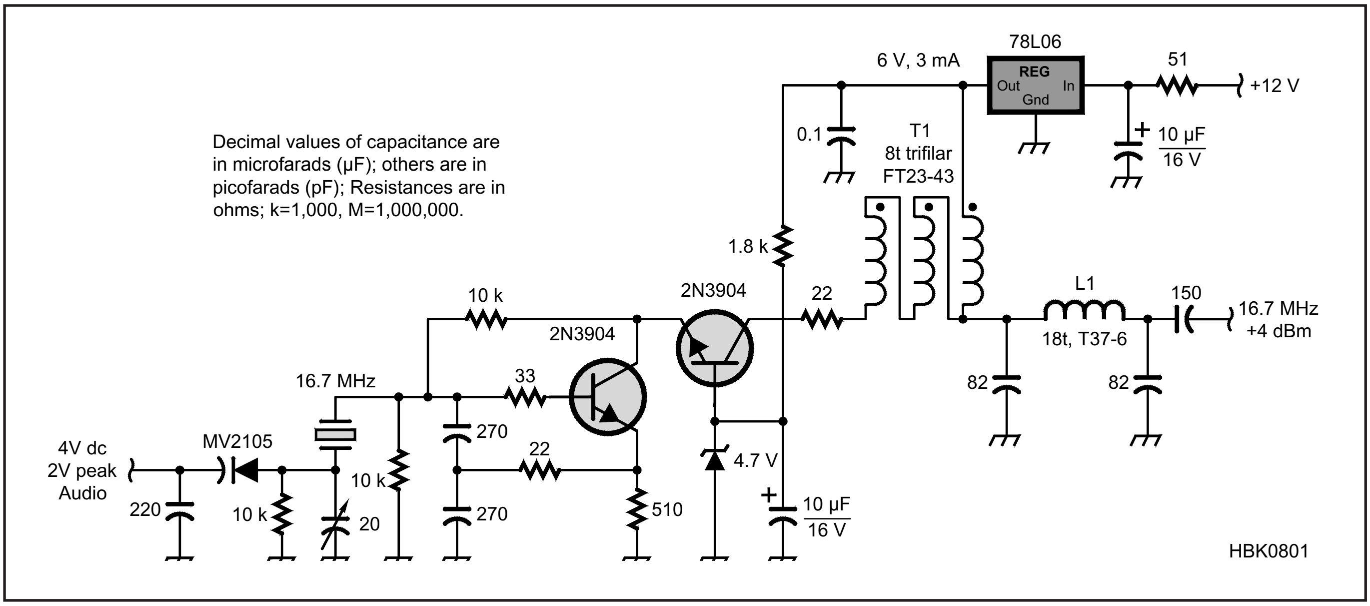


Fig 13.40 — Schematic showing how to add FM to the oscillator-multiplier boards.

provides nominal +10 dBm output power into a 50 Ω load, reduces the undesired harmonics of the fundamental crystal oscillator to below –70 dBc, and isolates the oscillator and multiplier circuit from external load variations. The jumper provides a convenient place to key the output amplifier, with key up isolation better than 40 dB.

Typical output from the 50 MHz VHF source is +12 dBm, and typical output for the 144 MHz signal source is about +8 dBm. The use of a continuous ground plane and chip capacitors for power supply bypassing reduces the need for shielding between stages on the circuit board. It is good practice to enclose any oscillator in a metal thermal and electrical shield box. The common gate configuration provides excellent isolation between the two double-tuned circuits, which means the tuning adjustments have very little interaction. Very little gain is needed, as the frequency multiplier output is a few milliwatts at the desired output frequency.

13.2.5 Tuning Up the Circuit

It is recommended that the circuit be initially built up and tested stage by stage, starting with the oscillator-buffer. The output is observed on a diode probe or oscilloscope after the pi-network, and the turns of toroid inductor L1 squeezed or spread for maximum output into 50 Ω. When experimenting with inductor values on a tight printed circuit board layout, adjustment is easier if the inductor is initially tack soldered to the back of the circuit board. Experiments are much easier with "ugly construction" (see the **Construction Techniques** chapter), and initial prototypes of new designs are often built that way. The frequency multiplier stage components are then added, with the output connected to the link on the second tuned circuit. The correct harmonic may be identified by calculating frequency from the trace on the oscilloscope or by listening with a receiver tuned to the desired output frequency and a short wire antenna near the output. Finally, the output stage amplifier components are added and the complete circuit adjusted for maximum power output into 50 Ω at the desired output frequency.

Once a VHF signal source has been built and tested, it is easy to duplicate the circuit by copying the component placement, turns spacing on the inductors, and presetting the variable capacitors to match the working circuit. After one of the VHF signal sources is working, the rest quickly fall into place. VHF component spacing is necessarily tight and leads are short, so this is not a good first electronic construction project.

13.2.6 Modulating the VHF Sources

A number of basic modulation types may be implemented with the VHF signal sources. As previously noted, the jumper on the output stage switches the tuned common gate output stage between modest gain and over 40 dB attenuation. A telegraph key or simple push-button switch may be connected in place of the jumper for on-off keying of the VHF source output. Engineers refer to on-off keying as OOK, hams call it CW, and the general public refers to it as Morse code. In any case, it is still the most basic mode for communicating information electronically.

FM is easily implemented as shown in **Fig 13.40**. A varactor diode, 220 pF capacitor, and resistor are tacked to the back side of the circuit board. Students have experimented with Zener diodes operated below breakdown as varactor replacements with reasonable results. Several volts of audio floating at a dc offset of about 4 V is applied to the varactor. **Fig 13.41** is the microphone amplifier circuit used with 144 MHz signal sources.

The VHF signal source can produce envelope modulation — AM — using either high level modulation or by driving a diode ring modulator with a little dc offset to introduce carrier. For high level modulation, both the multiplier and output stage may be modulated by introducing an ac signal to the dc power supply. High level envelope modulation is best implemented and adjusted experimentally by watching the output envelope on an oscilloscope while trying circuit modifications and adjusting drive levels.

For single sideband or any of the digital modes, the +10 dBm output level of the VHF signal source is ideal for driving any of a number of different IQ modulator configurations. Several successful 6 meter SSB transmitters have been built by replacing the on-board HF crystal oscillator in the microT2 circuit described elsewhere in this text with an external VHF signal source.

13.2.7 Creating a Direct Conversion Receiver

As mentioned in the introduction, a simple and effective direct conversion receiver is easily built with the VHF signal source as a tunable LO. Tuning range is about 20 kHz on

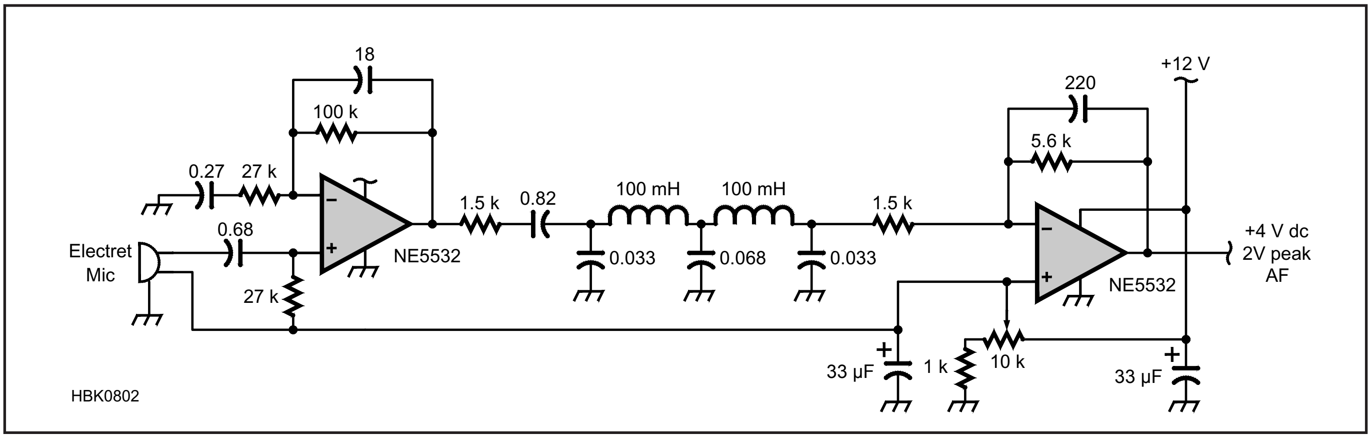


Fig 13.41 — Schematic of the microphone amplifier circuit used with the 2 meter signal sources.

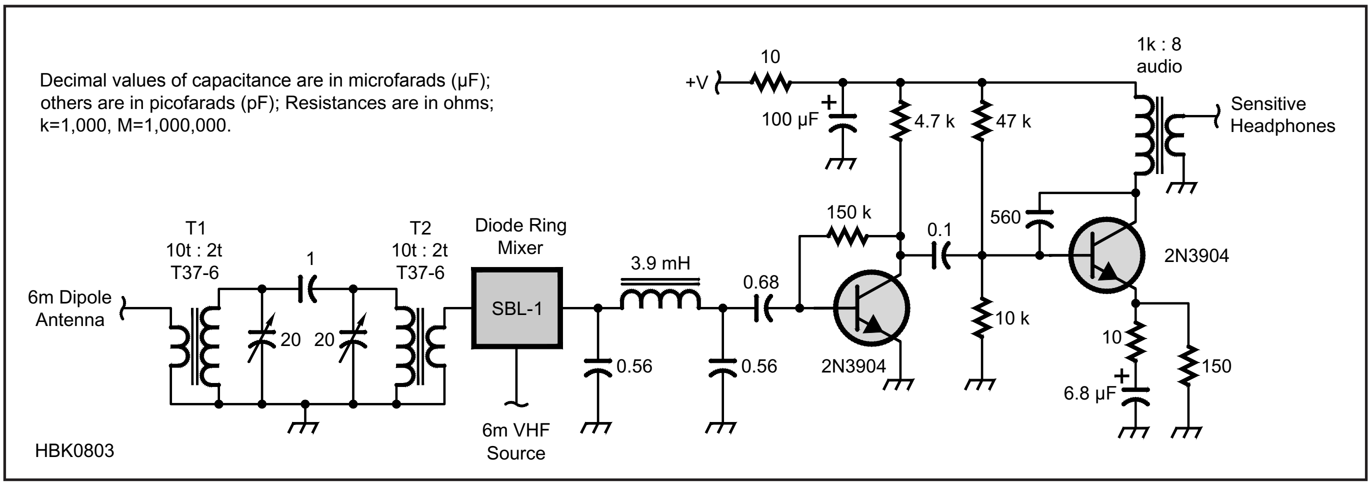


Fig 13.42 — Schematic of the simple 6 meter direct conversion receiver.

6 meters and 60 kHz on 2 meters. **Fig 13.42** is the schematic. Note how it combines and builds on previous circuits.

An LNA identical to the tuned buffer amplifier in the VHF signal source could be added after operation is confirmed, including the keying circuit as a receive mute. The diode ring mixer may be a Mini-Circuits SBL-1, TUF-1, ADE-1 or similar.

The audio amplifier with low-pass filter features a number of basic designs covered in other electronics classes, and in any of the basic electronic literature. Students are highly encouraged to build the direct conversion receiver using high performance VHF ugly construction techniques rather than laying out a circuit board. As noted by Wes Hayward in his original "Progressive Communications Receiver" article (Nov 1981 *QST*), when a performance difference between careful ugly construction and printed circuit board implementations of his HF receiver could be measured, the advantage went to ugly construction. For student use, ugly construction offers easy circuit modification and experimentation. Portland State design engineering students are required to modify and attempt to improve their circuits once they are operating — these are not a set of canned lab exercises!

Experiments with simple low power 50 MHz CW transmitters and direct conversion receivers using basic components rather than complex black-box integrated circuits are an excellent introduction to the fundamentals of 21st century radio. Cell phones, computers, and software applications are now taken for granted, but component-level RF electronics and electromagnetics are much closer to the underlying science. Students and practitioners of the radio frequency arts as a technical pastime can start here and follow their own experimental design paths.

13.3 Increasing Transmitter Power

The functions described so far that process input data and information and result in a signal on the desired output radio frequency generally occur at a low level. The one exception is full-carrier AM, in which the modulation is classically applied to the final amplification stage. More modern linear transmitter systems generate AM in the same way as SSB at low levels, typically between 1 mW and 1 W.

13.3.1 Types of Power Amplifiers

The **RF Power Amplifiers** chapter provides a detailed view of power amplifiers; however, we will take a quick peek here to set the stage for the following discussions. Amplifiers use dc power applied to active devices in order to increase the power or level of signals. As will all real devices, they introduce some distortion in the process, and are generally limited by the level of distortion products. Power amplifiers can be constructed using either solid-state devices or vacuum tubes as the active device. At higher powers, typically above a few hundred watts, vacuum tubes are more frequently found, although there is a clear trend toward solid state at all amateur power levels.

Independent of the device, amplifiers are divided into classes based on the fraction of the input cycle over which they conduct. A sinusoidal output signal is provided either by the *flywheel* action of a resonant circuit or by other devices contributing in turn. The usual amplifier classes are summarized in **Table 13.1**. Moving from Class A toward Class C, the amplifiers become progressively less linear but more efficient. The amplifiers with a YES in the LINEAR column thus are not all equally linear however A, AB or Class B amplifiers can be suitable for operation in a linear transmitter chain. Class C amplifiers can be used only for amplification of signals that do not have modulation information contained in the amplitude, other than on-off keyed signals. Thus class C amplifiers are useful for amplification of sinusoids, CW, FM, or as the nonlinear stage at which high-level AM modulation is employed.

In handheld digital cellular transceivers and base stations, the power amplifier must operate in a linear fashion to meet spectral purity requirements for the complex digital modulation schemes used. Since linear amplifiers are generally not very efficient, this is a major contributor to the energy consumption. In an effort to extend battery life in portable units, and reduce wasted energy in fixed equipment, considerable research is underway to use nonlinear amplifiers to provide linear amplification. Such techniques include pre-distortion of the low-level signal, polar modulation, envelope elimination and restoration, Cartesian-loop feedback and others.

Table 13.1
Summary of Characteristics of Power Amplifier Classes
Values are Typical

Class	*Conduction*	*Linear*	*Efficiency*
A	360°	Yes	30%
AB	270°	Yes	55%
B	180°	Yes	65%
C	90°	No	74%

13.3.2 Linear Amplifiers

While transmitters at power levels of 1 mW to 1 W have been successfully used for communication across many portions of the spectrum, most communications systems operate with more success at higher powers. The low level stage is usually referred to as an exciter, while higher power is provided by one or more linear amplifier stages as shown in **Fig 13.43**.

The power levels shown at the various points in Fig 13.43 are fairly typical for a high powered amateur station. The 1500 W PEP output represents the legal limit for US amateurs in most bands (200 W PEP on 30 meters and 50 W ERP_D on the 60 meter channels are notable exceptions). The first amplifier block may contain more than one stage, while the final output amplifier is often composed of multiple parallel active devices.

Typical power supply requirements for the amplifier stages are noted for a number of reasons. First, while power is rarely an issue at the exciter level, often it is a significant issue at the power levels shown for the amplifiers. The power supplies represent a large portion of the cost and weight of the system as the power increases. Some manufacturers are beginning to use switching-type power supplies for high-power amplifiers, resulting in a major reduction in size and weight.

Note also that a gross amplifier efficiency of about 50% is assumed for the amplifiers, taking into account ancillary subsystems as well as the inefficiency of the active devices in linear mode. The 50% that doesn't result in actual RF output is radiated as heat from the amplifier and must be removed from the amplifier as it is generated to avoid component damage. This represents another cost and weight factor that increases rapidly with power level.

The voltages shown for the supplies are those typical of modern solid state amplifiers. While virtually all commercial equipment now includes solid state amplifiers at the 100 W level, vacuum tube active devices are frequently found at higher levels, although the trend is clearly moving toward solid state. Vacuum tube amplifiers typically operate at voltages in the 2 to 4 kV range, requiring stringent measures be taken to avoid arcing across components. In addition, vacuum tube amplifiers typically dissipate up to 100 W of filament power that must be added to the power supply and heat dissipation planning.

13.3.3 Nonlinear Amplifiers

Nonlinear transmitters are somewhat different in architecture than the linear systems discussed previously. The configuration of a high-level AM modulated transmitter is shown in **Fig 13.44**. Note that none of the upper RF stages (the "RF chain") need to be particularly linear. The final stage must be nonlinear to have the modulation applied. Thus the RF stages can be the more power-efficient Class C amplifiers if desired.

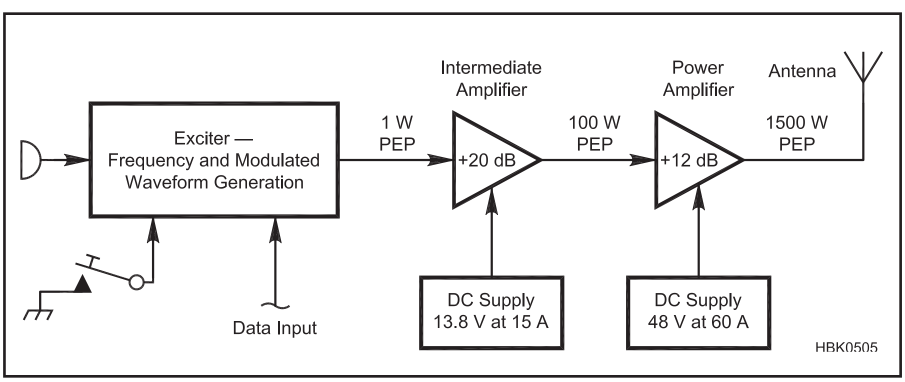


Fig 13.43 — Block diagram of a solid-state linear transmitter chain with multiple amplifier stages.

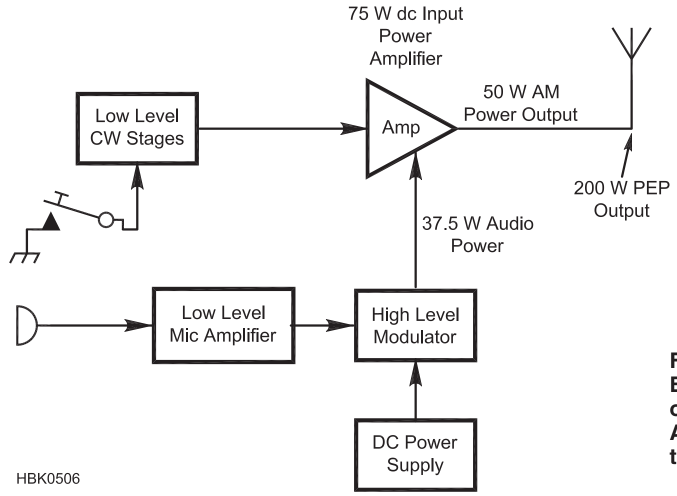


Fig 13.44 — Block diagram of a high level AM modulated transmitter.

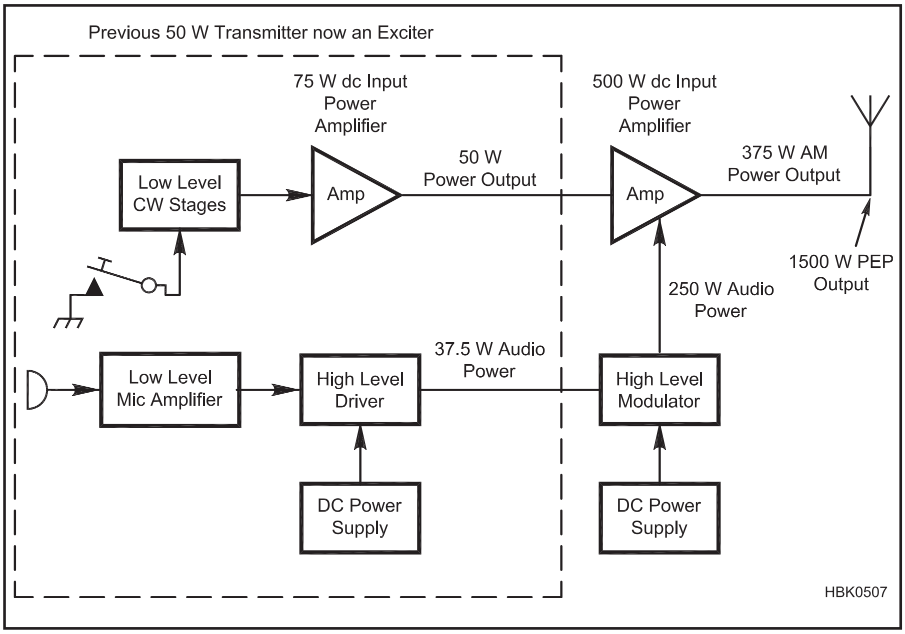


Fig 13.45 — Block diagram of a high level AM modulated transmitter with added output stage.

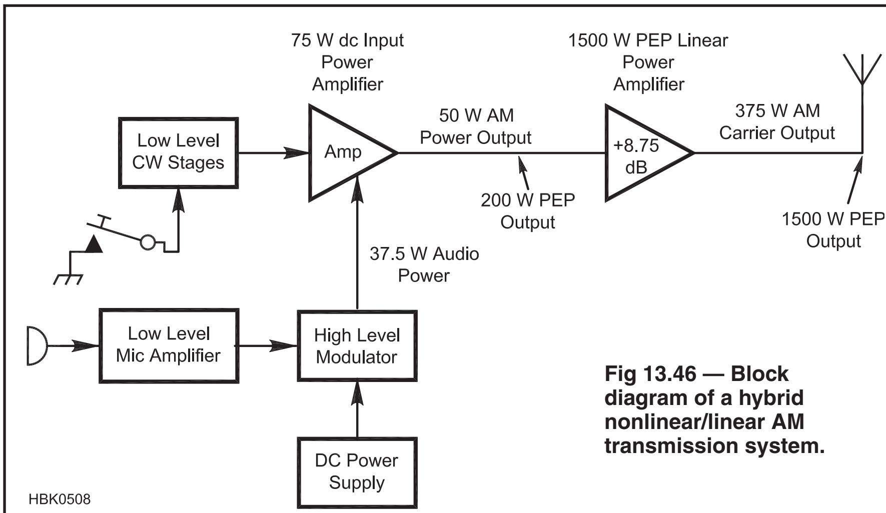


Fig 13.46 — Block diagram of a hybrid nonlinear/linear AM transmission system.

There are some observations to be made here. Note that the RF chain is putting out the full carrier power whenever in transmit mode, requiring a 100% duty cycle for power and amplifier components, unlike the SSB systems discussed previously. This imposes a considerable weight and cost burden on the power supply system. Note also that the PEP output of a 100% modulated AM system is equal to four times the carrier power.

The typical arrangement to increase the power of such a system is to add not only an RF amplifier stage capable of handling the desired power, but also to add additional audio power amplification to fully modulate the final RF stage. For 100% high-level plate modulation, an audio power equal to half the dc input power (plate voltage times plate current of a vacuum tube amplifier) needs to be provided. This arrangement is shown in **Fig 13.45**. In the example shown, the lower level audio stages are provided by those of the previous 50 W transmitter, now serving as an exciter for the power amplifier and as a driver for the modulating stage. This was frequently provided for in some transmitters of the AM era, notably the popular E. F. Johnson Ranger series, which provided special taps on its modulation transformer for use as a driver for higher-power systems.

It is worth mentioning that in those days the FCC US amateur power limit was expressed in terms of dc *input* to the final stage and was limited to 1000 W, rather than the 1500 W PEP *output* now specified. A fully modulated 1000 W dc input AM transmitter would likely have a carrier output of 750 W or 3000 W PEP — 3 dB above our current limit. If you end up with that classic Collins KW-1 transmitter, throttle it back to make it last and stay out of trouble!

13.3.4 Hybrid Amplifiers

Another alternative that is convenient with current equipment is to use an AM transmitter with a linear amplifier. This can be successful if the relationship that PEP = 4 × Carrier Power is maintained. **Fig 13.46** shows a 1500 W PEP output linear amplifier following a typical 50 W AM transmitter. In this example, the amplifier would be adjusted to provide a 375 W carrier output with no modulation applied to the exciter. During voice peaks the output seen on a special PEP meter, or using an oscilloscope, should be 1500 W PEP.

Note that during AM operation, the amplifier is producing a higher average power than it would without the carrier being present, as in SSB mode. The duty cycle specification of the amplifier should be checked to be sure it can handle the heavier load. If the amplifier has an RTTY rating, it should be safe to run an AM carrier at 66% of the RTTY output, following the required on and off time intervals.

13.4 Transceiver Construction and Control

With each generation of transceivers, in what has become a highly competitive marketplace, additional features were added as technology marched on. The deficiencies of early transceivers, in comparison to separate receivers and transmitters, quickly disappeared to the point that 100 W (or higher power) class transceivers exceeded the features of the best separate receivers and transmitters of the past.

The current generation of transceivers was designed using modern solid-state components that permit abundant functionality in a small enclosure. The designers and manufacturers have taken advantage of the possibilities of integrated electronics and microprocessors, incorporating many more functions into a transceiver than could have been envisioned in the early days. This is largely a function of improved technology becoming available at reduced cost.

Separate receivers and transmitters could be built with similar features and performance, but the required duplication of subsystems would make each unit cost about the same as a transceiver. Several manufacturers do make stand-alone receivers using components from transceivers, but they are generally aimed toward different markets, such as shortwave listeners or military/commercial users.

13.4.1 Upconverting Architecture

Fig 13.47 shows a traditional superheterodyne architecture for transceivers in which the sideband filter, some amplifiers, and other filters are shared between transmit and receive modes through the use of extensive switching. A limitation of that architecture is that it is not trivial to provide operation on frequencies near the first IF. The typical transceiver designer selected a first IF frequency away from the desired operating frequencies and proceeded on that basis.

New amateur bands at 30, 17, and 12 meters were approved at the 1979 ITU World Administrative Conference. The difficulties of managing image rejection on the new bands and the desire for continuous receiver coverage of LF, MF and HF bands (general-coverage receive) required a significant change in the architecture of receivers and transceivers. Thus, the upconverting architecture discussed in the **Receivers** chapter became popular in the 1980s and, with a few notable exceptions, became almost universal in commercial products over the following decade.

The solution was to move to the upconverting architecture shown in **Fig 13.48**. By selecting a first IF well above the highest receive frequency, the first local oscillator can cover the entire receive range without any gaps. With the 70 MHz IF shown, the full range from 0 to 30 MHz can be covered by an LO covering 70 to 100 MHz, less than a 1.5:1 range, making it easy to implement with modern PLL or DDS technology. Note that the high IF makes image rejection very easy and, rather than the usual tuned bandpass front end, we can use more universal low-pass filtering. The low-pass filter is generally shared with the transmit side and designed with octave cutoff frequencies to reduce transmitter harmonic content. A typical set of HF transceiver low-pass filter cut-off frequencies would be 1, 2, 4, 8, 16 and 32 MHz.

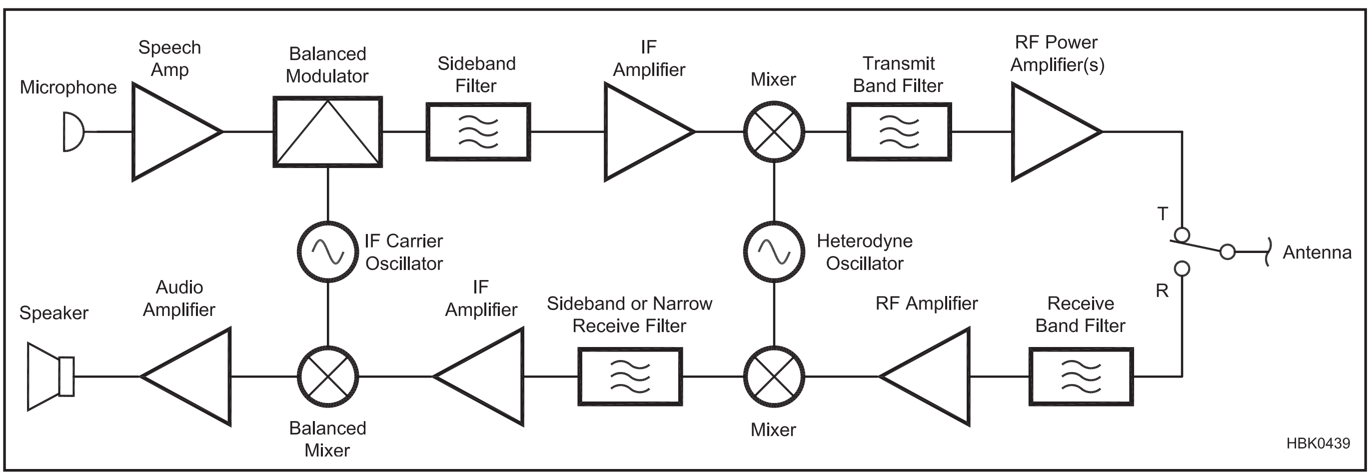


Fig 13.47 — Block diagram of a simple SSB transceiver sharing oscillator frequencies.

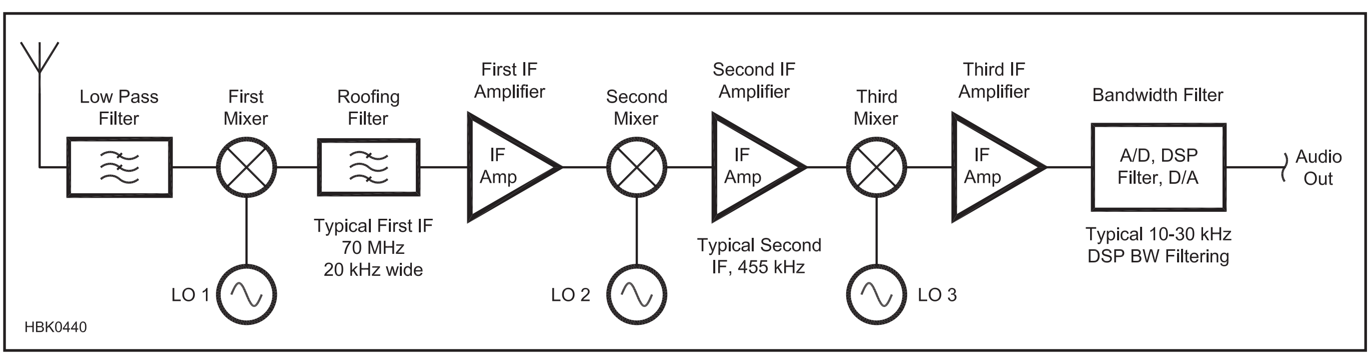


Fig 13.48 — Simplified block diagram of upconverting general coverage transceiver, receiver section shown.

This architecture offers significant benefits. By merely changing the control system programming, any frequency range or ranges can be provided with no change to the architecture or hardware implementation. Unlike the more traditional transceiver architecture (Fig 13.47), continuous receive frequency coverage over the range is actually easier to provide than to not provide, offering a marketing advantage for those who also like to do shortwave or broadcast listening.

13.4.2 Break-In CW Operation

Most current 100 W class HF transceivers use high-speed relays (with the relay actually following the CW keying) or solid-state PIN diodes to implement full break-in CW. Some RF power amplifiers use high-speed vacuum relays for the TR switching function. See the section on TR Switching later in this chapter for more information about circuits to perform this function. Two projects for adding QSK switching to linear amplifiers are included in the **Station Accessories** chapter.

The term "semi-break-in" is used to designate a CW switching system in which closing the key initiates transmission, but switching back to receive happens between words, not between individual dits. Some operators find this less distracting than full break-in, and it is easier to implement with less-expensive relays for the TR switching.

13.4.3 Push-To-Talk for Voice

Another advance in amateur station switching followed longstanding practices of aircraft and mobile voice operators who had other things to contend with besides radio switches. Microphones in those services included built-in switches to activate TR switching. Called push-to-talk (PTT), this function is perhaps the most self-explanatory description in our acronym studded environment.

Relays controlled the various switching functions when the operator pressed the PTT switch. Some top-of-the-line transmitters of the period included at least some of the relays internally and had a socket designed for PTT microphones. **Fig 13.49** is a view of the ubiquitous Astatic D-104 microphone with PTT stand, produced from the 1930s to 2004, and still popular at flea markets and auction sites. PTT operation allowed the operator to be out of reach of the radio equipment while operating, permitting "easy chair" operation for the first time.

Modern transceivers include some form of PTT (or "one switch operation"). Relays, diodes, transistors and other components seamlessly handle myriad transmit-receive changeover functions inside the transceiver. Most transceivers have additional provisions for manually activating PTT via a front-panel switch. And many have one or more jacks for external PTT control via foot switches, computer interfaces or other devices.

Fig 13.49 — A classic Astatic D-104 mic with PTT stand.

13.4.4 Voice-Operated Transmit-Receive Switching (VOX)

How about break-in for voice operators? SSB operation enabled the development of voice operated transmit/receive switching, or VOX. During VOX operation, speaking into the microphone causes the station to switch from receive to transmit; a pause in speaking results in switching back to receive mode. Although VOX technology can work with AM or FM, rapidly turning the carrier signal on and off to follow speech does not provide the smooth operation possible with SSB. (During SSB transmission, no carrier or signal is sent while the operator is silent.)

VOX OPERATION

VOX is built into current HF SSB transceivers. In most, but not all, cases they also provide for PTT operation, with switches or menu settings to switch among the various control methods. Some operators prefer VOX, some prefer PTT and some switch back and forth depending on the operating environment.

VOX controls are often considered to be in the "set and forget" category and thus may be controlled by a software menu or by controls on the rear panel, under the top lid or behind an access panel. The following sections discuss the operation and adjustment of radio controls associated with VOX operation. Check your transceiver's operating manual for the specifics for your radio.

Before adjusting your radio's VOX controls, it's important to understand how your particular mic operates. If it has no PTT switch, you can go on to the next section! Some mics with PTT switches turn off the audio signal if the PTT switch is released, while some just open the control contacts. If your mic does the former, you will need to lock the PTT switch closed, have a different mic for VOX, or possibly modify the internal mic connections to make it operate with the VOX. If no audio is provided to the VOX control circuit, it will never activate. If the mic came with your radio, or from its manufacturer, you can probably find out in the radio or mic manual.

VOX Gain

Fig 13.50 shows some typical transceiver VOX controls. The VOX GAIN setting determines how loud speech must be to initiate switchover, called "tripping the VOX." With a dummy load on the radio, experiment with the setting and see what happens. You should be able to advance it so far that it switches with your breathing. That is obviously too sensitive or you will have to hold your breath while receiving! If not sensitive enough, it may cause the transmitter to switch off during softly spoken syllables. Notice that the setting depends on how close you are to the microphone, as well as how loud you talk. A headset-type microphone (a "boom set") has an advantage here in that you can set the microphone distance the same every time you use it.

The optimum setting is one that switches to transmit whenever you start talking, but isn't so sensitive that it switches when the mic picks up other sounds, such as a cooling fan turning on or normal household noises.

Fig 13.50 — The function of VOX controls is described in the text. They require adjustment for different types of operating, so front-panel knobs make the most convenient control arrangement. In some radios, VOX settings are adjusted through the menu system.

VOX Delay

As soon as you stop talking, the radio can switch back to receive. Generally, if that happens too quickly, it will switch back and forth between syllables, causing a lot of extra and distracting relay clatter. The VOX DELAY control determines how long the radio stays in the transmit position once you stop talking. If set too short, it can be annoying. If set too long, you may find that you miss a response to a question because the other station started talking while you were still waiting to switch over.

You may find that different delay settings work well for different types of operation. For example, in a contest the responses come quickly and a short delay is good. For casual conversation, longer delays may be appropriate. Again, experiment with these settings with your radio connected to a dummy load.

Anti-VOX

This is a control with a name that may mystify you at first glance! While you are receiving, your loudspeaker is also talking to your mic — and tripping your VOX — even if you aren't! Early VOX users often needed to use headphones to avoid this problem. Someone finally figured out that if a sample of the speaker's audio signal were fed back to the mic input, out-of-phase and at the appropriate amplitude, the signal from the speaker could be cancelled out and would not cause the VOX circuit to activate the transmitter. The ANTI-VOX (called ANTI-TRIP in the photo) controls the amplitude of the sampled speaker audio, while the phase is set by the transceiver design.

As you tune in signals on your receiver with the audio output going to the speaker, you may find that the VOX triggers from time to time. This will depend on how far you turn up the volume, which way the speaker is pointed and how far it is from the mic. You should be able to set the ANTI-VOX so that the speaker doesn't trip the VOX during normal operation.

Generally, setting ANTI-VOX to higher values allows the speaker audio to be louder without activating the VOX circuit. Keep in mind that once you find a good setting, it may need to be changed if you relocate your mic or speaker. With most radios, you should find a spot to set the speaker, mic and ANTI-VOX so that the speaker can be used without difficulty.

13.4.5 TR Switching

As the complexity of a transceiver increases, the business of switching between receive and transmit becomes quite complex. In commercially built equipment, this function is usually controlled by a microprocessor that manages any necessary sequencing and interlock functions that would require an excessive amount of circuitry to implement with discrete components. For an example of just how complex TR switching could be in an advanced transceiver, look at the schematic for any mid-level or top-of-the-line solid-state transceiver sold in the 1980s or 1990s!

Nevertheless, the basic functions of TR switching are well within scope for the amateur building a transceiver. Understanding TR switching will also assist in troubleshooting a more complex commercial radio. Even full break-in keying is possible: Two schematics of circuits for fast TR switching from an article by Jack Kuecken, KE2QJ,

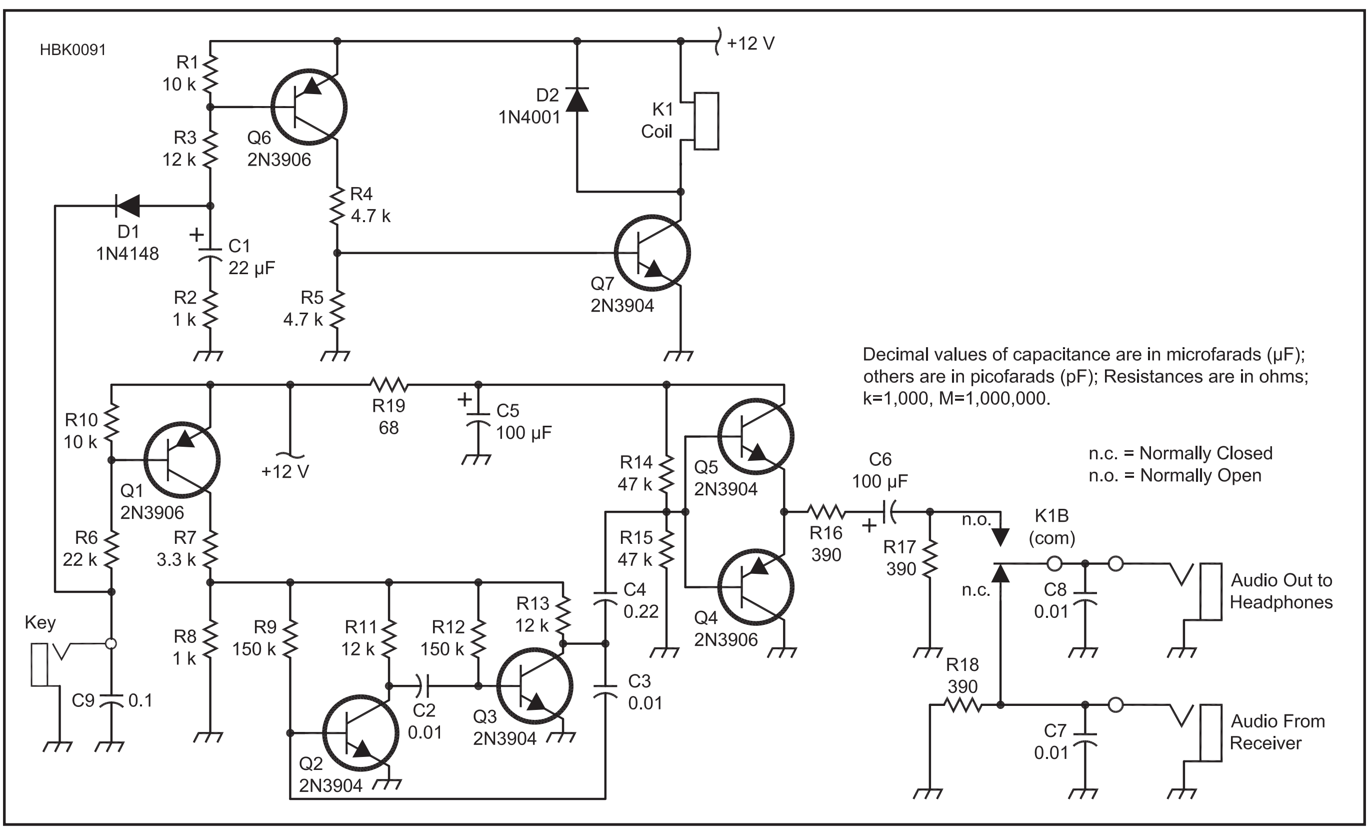


Fig 13.51 — Detailed schematic diagram and parts list for transmit-receive control section and sidetone generator of the universal QRP transmitter. Resistors are ¼ W, 5% carbon film. A kit of component parts is available from KangaUS (www.kangaus.com).

C1 — 22 µF, 25 V electrolytic
C2, C3, C7, C8 — 0.01 µF, 50 V ceramic
C4 — 0.22 µF, 50 V ceramic
C5, C6 — 100 µF, 25 V electrolytic
C9 — 0.1 µF, 50 V ceramic
K1 — DPDT 12 V coil relay. An NAIS DS2Y-S-DC12, 700 Ω, 4 ms relay was used in this example.
Q1, Q4, Q6 — 2N3906, PNP silicon small signal transistor
Q2, Q3, Q5, Q7 — 2N3904, NPN silicon small signal transistor

are included below, and the full article is included on this book's CD-ROM.

Numerous schemes are popular for switching an antenna between transmitter and receiver functions. But these schemes tend to get in the way when one is developing both simple receivers and transmitters, perhaps as separate projects. A simple relay-based TR scheme is then preferred and is presented here. In this system, used in the MkII Updated Universal QRP Transmitter by Wes Hayward, W7ZOI (see the full article on this book's CD-ROM), the TR relay not only switches the antenna from the receiver to the transmitter, but disconnects the headphones from the receiver and attaches them to a sidetone oscillator that is keyed with the transmitter.

The circuitry that does most of the switching is shown in **Fig 13.51**. A key closure discharges capacitor C1. R2, the 1 kΩ resistor in series with C1, prevents a spark at the key. Of greater import, it also does not allow us to "ask" that the capacitor be discharged instantaneously, a common request in similar published circuits. Key closure causes Q6 to saturate, causing Q7 to also saturate, turning the relay on. The relay picked for this example has a 700 Ω, 12 V coil with a measured 4 ms pull-in time.

Relay contacts B switch the audio line. R17 and R18 suppress clicks related to switching. A depressed key turns on PNP switch Q1, which then turns on the sidetone multivibrator, Q2 and Q3. The resulting audio is routed to switching amplifier Q4 and Q5. Although the common bases are biased to half of the supply voltage, emitter bias does not allow any static dc current to flow. The only current that flows is that related to the sidetone signal during key down intervals. Changing the value of R16 allows the audio volume to be adjusted, to compensate for the particular low-impedance headphones used.

Depending on the architecture of the transceiver, there will likely have to be some additional control circuitry in order to avoid annoying switching artifacts. These generally fall into the category of transients in the receiver audio and turning on the transmitter too slowly to capture initial code elements, also known as "dot shortening." A review of the transceiver design from which this circuit is taken and of other homebuilt transceiver designs will illustrate the problem and the methods used to address it.

If full break-in TR switching is required, high-speed switching components such as a reed relay or PIN diodes are required. KE2QJ provided a pair of such circuits (**Fig 13.52** and **Fig 13.53**) that can be adapted for internal use in a home-built transceiver, although their original purpose was to integrate a stand-alone receiver with a transceiver and linear amplifier. The full article is available on this book's CD-ROM.

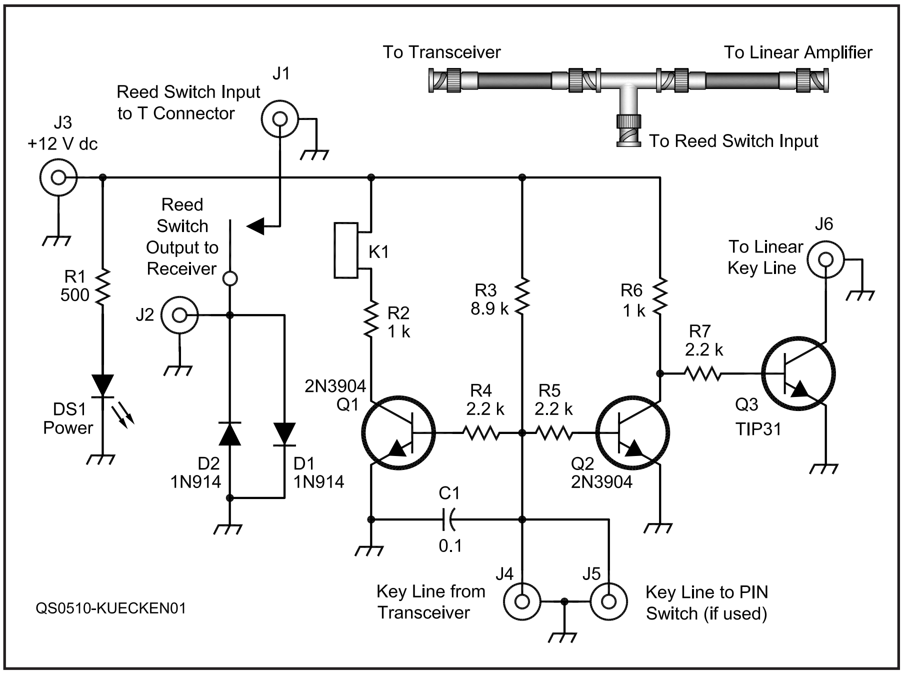


Fig 13.52 — Schematic and parts list for the reed relay TR switch. Resistors are ¼ W.

D1, D2 — High speed switching diode, 1N914 or equivalent
DS1 — LED
J1, J2 — Chassis mount BNC connectors
J3-J6—Chassis mount phono connectors
K1—SPST normally open reed relay with 12 V coil
Q1, Q2—Small signal PNP transistor, 2N3904 or equivalent
Q3—NPN power transistor, TIP31 or equivalent

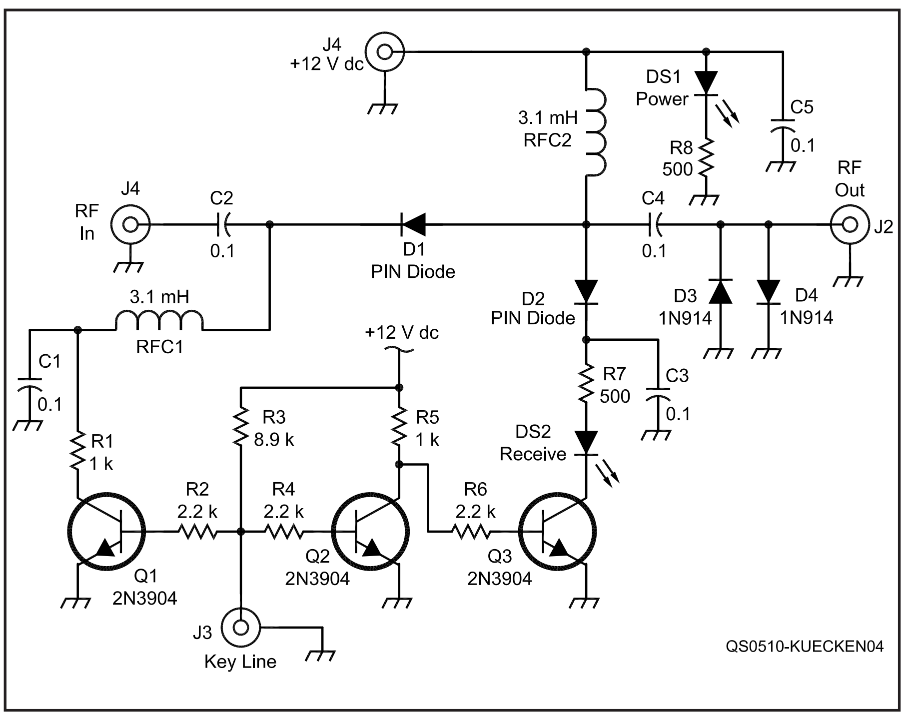


Fig 13.53 — Schematic and parts list for the PIN diode TR switch. Resistors are ¼ W.

C1-C5 — 0.1 µF ceramic capacitor
D1, D2 — PIN diode
D3, D4 — Switching diode, 1N914 or equivalent
DS1, DS2 — LED
J1, J2 — Chassis mount BNC connector
J3, J4 — Chassis mount phono connector
Q1-Q3 — Small signal PNP transistor, 2N3904 or equivalent
RFC1, RFC2 — 3.1 mH RF choke, 225 turns #30 AWG enameled wire wound on a 5⁄16 inch diameter, 5⁄8 inch long plastic tube

If you already have a receiver and transmitter and want to integrate them under the control of a separate TR switch, the K8IQY "Magic Box" (**www.4sqrp.com/MagicBox.php**) is available as a kit. This is a microprocessor-controlled design that can handle up to 10 W of transmitter power, switches at up to 50 WPM, and includes an audio sidetone output, as well. Complete documentation for the kit is available online, including schematics and design information. The kit could be extended to handle more transmit power with heavier components and the appropriate circuit changes.

13.4.6 TR Switching With a Linear Amplifier

Virtually every amateur HF transceiver includes a rear panel jack called something along the lines of KEY OUT, intended to provide a contact closure while in transmit mode. This jack is intended to connect to a corresponding jack on a linear amplifier called something like KEY IN. Check the transceiver and amplifier manuals to find out what they are called on your units. A diagram of the proper cabling to connect the transceiver and amplifier will be provided in the manual.

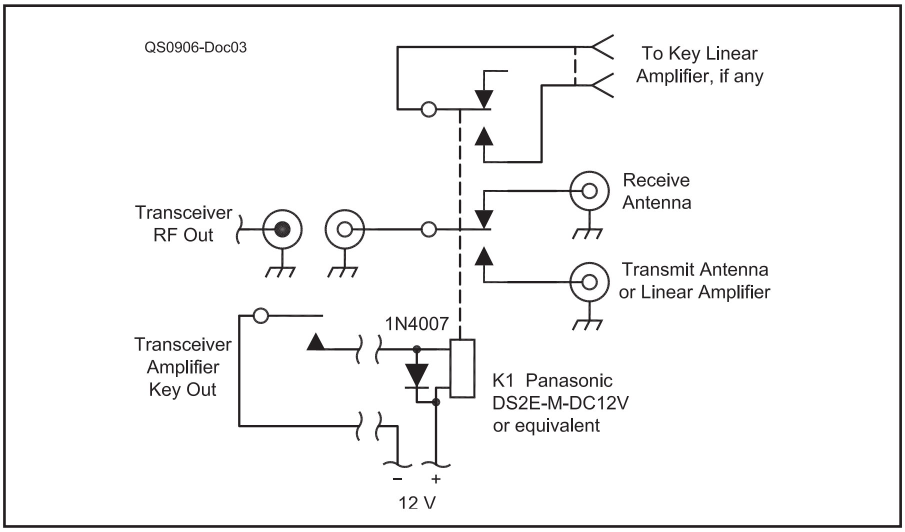


Fig 13.54 — Schematic of an external box that allows a modern transceiver to key a linear amplifier with TR switch voltage or current requirements that exceed the transceiver's ratings. As a bonus, it can also be used to allow reception from a low-noise receiving antenna.

ENSURING AMPLIFIER-TRANSCEIVER COMPATIBILITY

While you're there, check the ratings to find out how much voltage and current the transceiver can safely switch. In earlier days, this switching was usually accomplished by relay contacts. More modern radios tend to use solid-state devices to perform the function. Although recent amplifiers are usually compatible with the switching capabilities of current transceivers, the voltage and/or current required to switch the relays in an older linear amplifier may exceed the ratings. Fortunately, it is pretty easy to find out what your amplifier requires, and almost as easy to fix if it's not compatible with your radio.

If your amplifier manual doesn't say what the switching voltage is, you can find out with a multimeter or DMM. Set the meter to read voltage in a range safe for 250 V dc or higher. Connect the positive meter probe to amplifier key jack's center conductor, and connect the negative meter probe to the chassis ground (or other key jack terminal if it's not grounded). This will tell you what the open circuit voltage is on the amplifier key jack. You may need to try a lower voltage range, or an ac range, or switch the probes (if the key line is a negative dc voltage) to get a reading.

Now set the meter to read current. Start with a range that can read 1 A dc, and with the leads connected as before, you should hear the amplifier relay close and observe the current needed to operate the TR relay or circuit. Adjust the meter range, if needed, to get an accurate reading.

These two levels, voltage and current, are what the transceiver will be asked to switch. If *either* reading is higher than the transceiver specification, do not connect the transceiver and amplifier together. Doing so will likely damage your transceiver. You will need a simple interface circuit to handle the amplifier's switching voltage and current.

The simple, low-cost relay circuit shown in **Fig 13.54** can be used to key an older amplifier with a modern transceiver. It offers an added benefit: Another potential use of the transceiver KEY OUT jack is to switch to a separate low-noise receive antenna on the lower bands. While most high-end transceivers have a separate receive-only antenna connection built in, many transceivers don't. If you don't need one of the extra functions, just leave off those wires.

13.5 Transceiver Projects

There are many transceiver designs available, aimed mainly at the advanced builder. We will provide two. One describes the construction of a 5 W single-band SSB/CW transceiver that was the winner of the ARRL's first *Homebrew Challenge*. The other is the description of the design and construction process that resulted in a 100 W transceiver for all HF bands with performance (confirmed in the ARRL Lab) that is as good as some of the best commercial products. Construction details for these projects and support files are available on the ARRL website. Additional projects may be found on the *Handbook* CD-ROM.

13.5.1 Transceiver Kits

One of the simplest kits to build is also one of the least expensive. The ARRL offers the MFJ Enterprises (**www.mfjenterprises.com**) Cub 40 meter transceiver kit (**Figs 13.55** and **13.56**) bundled with *ARRL's Low Power Communication — The Art and Science of QRP*. (See **www.arrl.org/shop**.) Other kits are available from many manufacturers.

Ten-Tec, maker of some of the best HF transceivers, offers single band CW QRP kits for your choice of 80, 40, 30 or 20 meters. Elecraft offers a range of kits from the KX1 and KX3 travel radios to the K2 and K3 high performance multiband CW and SSB transceiver semi-kit (mechanical assembly only, no soldering required). The K2 and K3 can both start out as 10 W models and be upgraded to 100 W, if desired. DZkit (**www.dzkit.com**) offers the Sienna high performance transceiver kit.

If your interest is strictly QRP transceiver kits, there are many small and specialty manufacturers. The QRP Amateur Radio Club, Inc. (QRP ARCI) maintains a manufacturer list on their website at **www.qrparci.org**, available in the "Links" section of the site.

Fig 13.55 — MFJ Cub QRP transceiver built from a kit.

Fig 13.56 — MFJ Cub QRP transceiver kit showing parts supplied.

13.5.2 Project: The TAK-40 SSB CW Transceiver

Jim Veatch, WA2EUJ, was one of two winners of the first ARRL *Homebrew Challenge* with his TAK-40 transceiver shown in **Fig 13.57**. This challenge was to build a homebrew 5 W minimum output voice and CW transceiver using all new parts for under $50. The resulting radio had to meet FCC spurious signal requirements. The submitted radios were evaluated as to operational features and capabilities by a panel of ARRL technical staff. Jim's transceiver met the requirements for a transceiver that required a connected PC for setup — to load the PIC controller. This information was presented in May 2008 *QST* in an article provided by Jim as a challenge requirement. The information here is an overview of the project. The complete article including construction, adjustment, and final assembly information is available on the CD-ROM accompanying this book.

The following is a list of the criteria for the Homebrew Challenge and a brief description of how the TAK-40 meets the requirements:

• *The station must include a transmitter and receiver that can operate on the CW and voice segments of 40 meters.* The TAK-40 covers 7.0 to 7.3 MHz.

• *It must meet all FCC regulations for spectral purity.* All spurious emissions from the TAK-40 are at least 43 dB below the mean power of fundamental emission.

• *It must have a power output of at least 5 W PEP.* The TAK-40 generates at least 5 W PEP for voice and CW modes. The ALC can be set as high as 7 W if desired.

• *It can be constructed using ordinary hand tools.* Construction of the TAK-40 uses all leaded components and assembly requires only hand tools, soldering iron and an electric drill (helpful but not strictly necessary).

• *It must be capable of operation on both voice and CW.* The TAK-40 operates USB and LSB as well as CW. USB was included to allow the TAK-40 to easily operate in digital modes such as PSK31 using a PC and sound card.

• *Parts must be readily available either from local retailers or by mail order. No "flea market specials" allowed.* The TAK-40 is constructed from materials available from DigiKey, Mouser, Jameco and Amidon.

• *Any test equipment other than a multimeter or radio receiver must either be constructed as part of the project or purchased as part of the budget.* The TAK-40 requires only a multimeter for construction, and extensive built-in setup functions in the software include a frequency counter to align the oscillators and a programmable voltage source for controlling the oscillators.

• *Equipment need only operate on a single band, 40 meters. Multiband operation is acceptable and encouraged.* The TAK-40 operates on the 40 meter band.

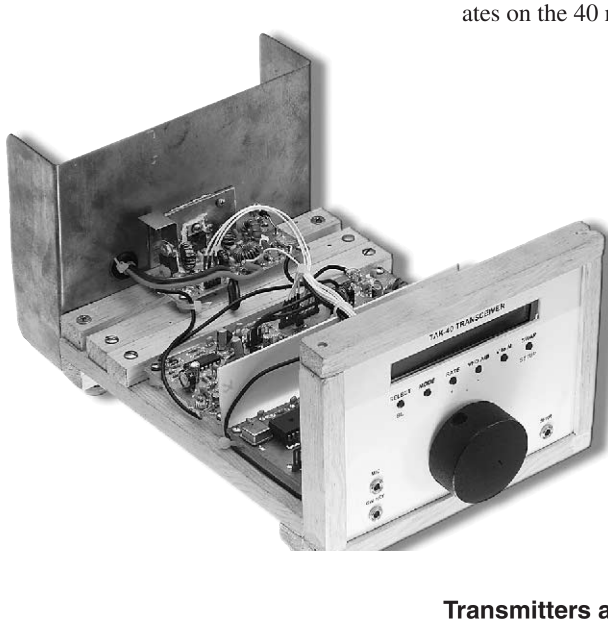


Fig 13.57 — Homebrew Challenge winner TAK-40 from Jim Veatch, WA2EUJ.

• *The total cost of all parts, except for power supply, mic, key, headphones or speaker, and usual supplies such as wire, nuts and bolts, tape, antenna, solder or glue must be less than $50.* The total cost of the parts required to build the TAK-40 was $49.50 at the time of the contest judging. It still remains an excellent price for the capabilities, even with the manufactured circuit boards as discussed below.

The TAK-40 also includes some features that make it very smooth to operate.

• Automatic Gain Control — regulates the audio output for strong and weak signals.

• S-Meter — simplifies signal reports.

• Digital frequency readout — reads the operating frequency to 100 Hz.

• Dual Tuning Rates — FAST for scanning the band and SLOW for fine tuning.

• Speech Processor — get the most from the 5 W output.

• Automatic Level Control — prevents overdriving the transmitter.

• Transmit power meter — displays approximate power output.

• Boot loader — accepts firmware updates via a computer (cable and level converter optional).

CIRCUIT DESCRIPTION

The TAK-40 transceiver is constructed as four modules:

• Digital section and front panel: The digital board contains the microprocessor, front panel controls, liquid crystal display (LCD), the digital-to-analog converter for the VFO, the beat frequency oscillator (BFO) and the oscillator switching matrix.

• Variable frequency oscillator (VFO): The VFO is implemented as a separate module for best stability and overall performance.

• Intermediate frequency (IF) board: RF

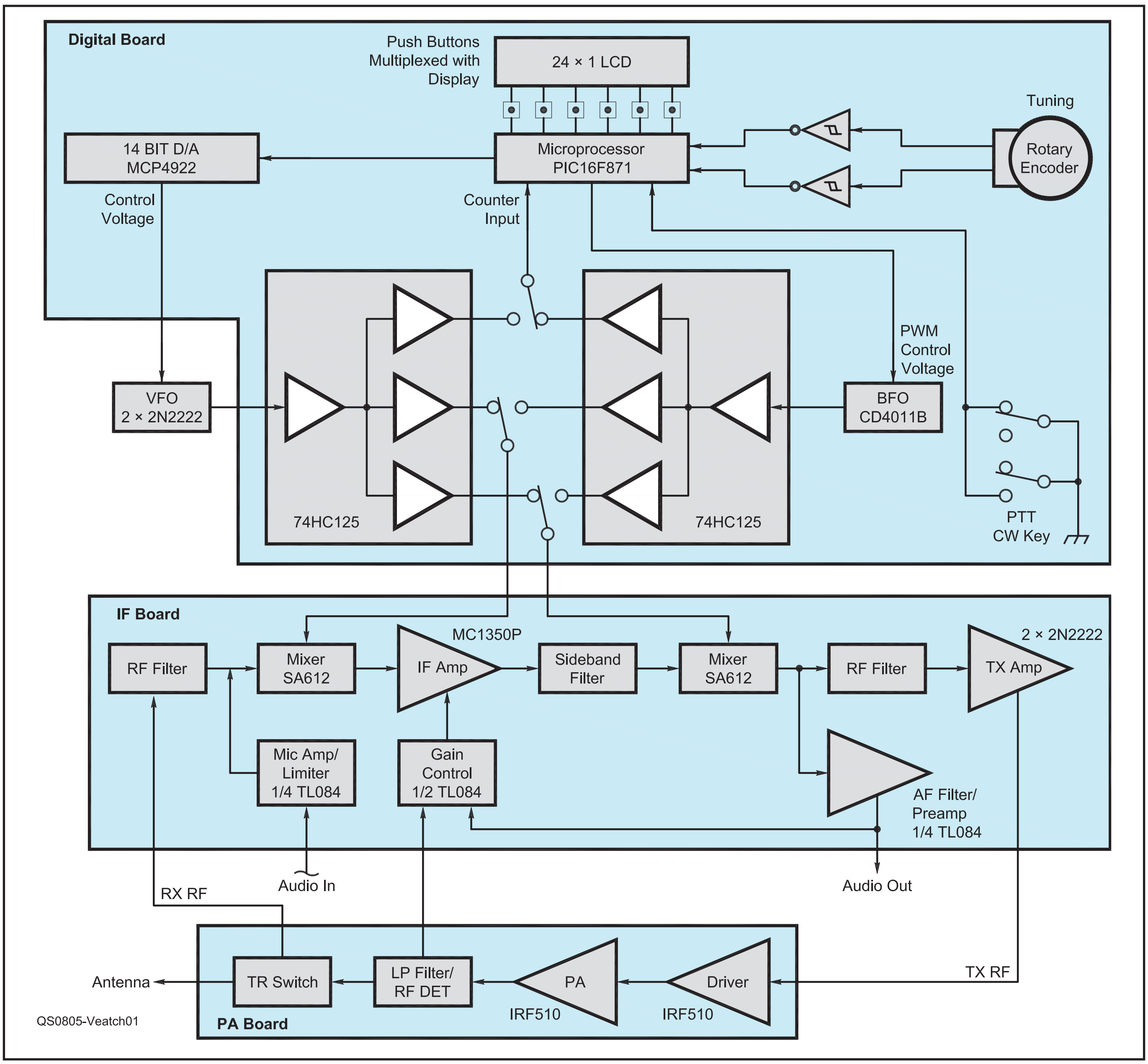


Fig 13.58 — Block diagram of TAK-40, homebrew challenge winner. This radio included many more features and capabilities than we expected to find in a $50 radio! Complete schematics, parts lists and firmware can be found at www.arrl.org/QST-in-depth. Look in the 2008 section for QS0508Veatch.zip.

filtering includes a 6-element crystal ladder filter, an integrated IF amplifier, two SA612 ICs used as a mixer and a BFO, and the audio output amplifier. The low-level RF transmit signal is also generated on this board.

• Power amplifier (PA): This module contains a driver and the final power amplifier (PA) board with filtering to meet the FCC spectral purity requirements.

The overall design is a classic superheterodyne with a 4-MHz IF and a 3- to 3.3-MHz VFO. The same IF chain is used for transmitting and receiving by switching the oscillator signals between the two mixers. **Fig 13.58** shows the block diagram of the TAK-40 transceiver. Each board is described below with a detailed schematic and parts list on the *QST* binaries website - **www.arrl.org/QST-in-depth**. Look in the 2008 section for file **QS0508Veatch.zip**.

CONSTRUCTION

The best way to build this transceiver would be to buy the printed circuit boards (PCB) but it would not have met the competition's $50 budget limit! Files included in the package on the *QST* website can be sent to **www.expresspcb.com** and they will send you two complete sets of boards for just over $100. (If you order more, the pre-set price decreases.) By ordering the PCBs you would complete the radio sooner than if you built it using any other technique and with a higher probability of success. The process by which the TAK-40 was constructed for the competition is described in the full article.

Fortunately the TAK-40 requires relatively little chassis wiring. A small harness for the LCD and push-button switches, cable for the rotary encoder audio in and out and key line wiring are all that is required for the front panel. The IF board connects to the VFO, IF and PA boards for control and metering. Two RF lines run between the IF and PA boards. The prototype is built on a wooden frame and the front panel printed on photo paper in an inkjet printer.

13.5.3 *Project:* A Homebrew High Performance HF Transceiver — the HBR-2000

Have you ever dreamed of building an Amateur Radio transceiver? Have you thought how good it would feel to say, "The rig here is homebrew"? Markus Hansen, VE7CA, shows us that it's still possible to roll your own full featured HF transceiver — and get competitive performance! The transceiver in **Fig 13.59** is a high-performance, 100 W HF transceiver that the author named the HBR-2000.

This project is a condensed version of an article that appeared in March 2006 *QST* and should provide an inspiration to all homebrewers. (The full original article is available in PDF format on the CD-ROM included with this book.) The goal of presenting the project here is not necessarily to suggest that it be replicated but to illustrate a successful approach to such a project.

The secret to being able to successfully build a major project such as this is to divide it into many small modules, as indicated in the block diagram (**Fig 13.60**). Each module represents a part of the whole, with each module built and tested before starting on the next. To choose the actual circuits that were to be built into each module, the author searched past issues of *QST*, *QEX*, *The ARRL Handbook* and publications dedicated to home brewing such as *Experimental Methods in RF Design*, published by ARRL. It's important to learn by reading, building a circuit and then taking measurements. After you build a particular circuit and measure the voltage at different points, you begin to understand how that particular circuit works.

Do not go on to the next module until finishing the previous module, including testing it to make sure it worked as expected. After building a module, follow the same process to decide on the circuit and build the next module. Then connect the two modules together and check to make sure that, when combined together, they performed as expected. It is really that simple, one step at a time. Anyone who has had some building experience can build a receiver and transmitter using this procedure.

For a receiver, begin by building the audio amplifier and product detector module, then the BFO circuit, testing each separately and then the growing receiver, step by step. From there, build the IF/AGC module, then the VFO and the heterodyne LO system, then the receiver mixer, and on and on until reaching the antenna. At that point you have a functioning receiver. It is a thrilling day when you hook up an antenna to the receiver and tune across the Amateur Radio bands listening to signals emanating from the speaker.

After you build a particular module, you don't want RF from outside sources to get into the modules you build and you don't want RF signals produced inside the modules to travel to other parts of the receiver, other than through shielded coaxial lines. The reason that you don't want RF floating around the receiver is that stray RF can produce unwanted birdies in the receiver, adversely affect the AGC system or cause other subtle forms of mischief. To prevent this from happening, enclose each module in a separate RF tight box and use coax for all the RF lines with BNC or phono connectors on each end. All dc and control lines are connected via feed thru insulators.

Modules are enclosed in boxes made from unetched copper clad material. For the covers, cut sheet brass ½ inch wider and longer than the size of the PC box. Then lay the box on top of the brass and center the box so that there is about ¼ inch overlap around the perimeter of the box, and draw a line around the box with a felt pen. Then cut the corners out with tin snips and bend the edges of the brass cover over in a vise. By drilling small holes around the perimeter of the box, inserting wires through the holes, soldering the wires to the inside of the box and to the overlapping edges, you produce an RF-tight enclosure. See **Fig 13.61** for an example of this technique.

TYPES OF CONSTRUCTION

The audio board and IF board are etched PC boards purchased from Far Circuits. The construction method the author learned to appreciate the most is "ugly construction," discussed in the **Circuit Construction** chapter. Each module is designed for an input and output impedance of 50 Ω except for the audio output (8 Ω for speaker connection). Thus 50-Ω coax cable with BNC connectors can be employed to connect the RF paths between the different modules. The concept is shown in **Fig 13.62**.

Fig 13.59 — Head on view of the HBR-2000. This looks a lot like a commercial transceiver.

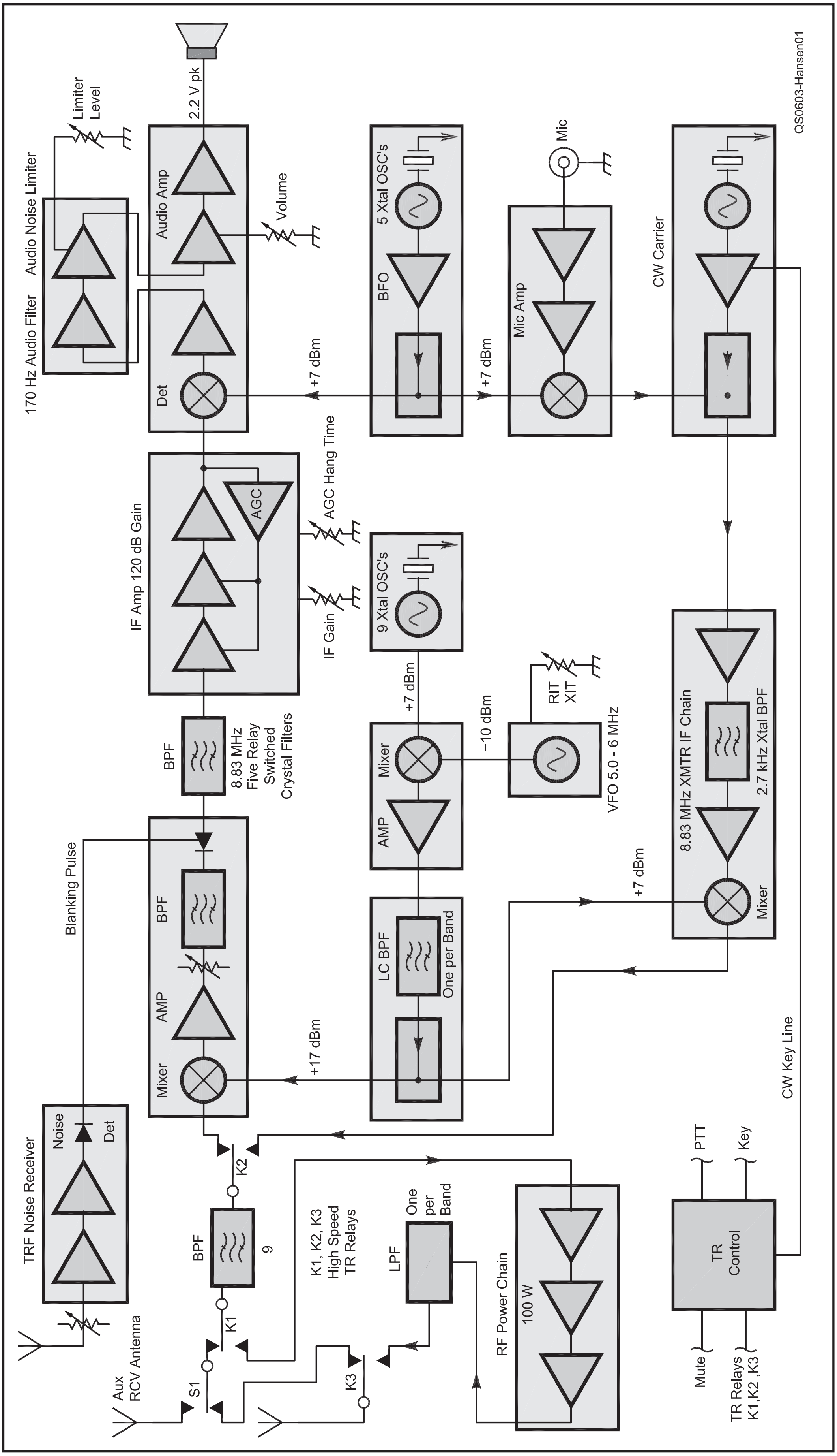


Fig 13.60 — Block diagram of the HBR-2000.

Fig 13.61 — Sample of a box made with PC board with a sheet brass cover overlapping all four sides. Note the BNC connectors and feed thru insulator.

MEASUREMENTS AND TEST EQUIPMENT

Making meaningful and accurate measurements is a major part of producing a successful project. You must make measurements as you progress or you have no way of knowing whether a module is performing according to design specifications.

One of the specifications chosen for the HBR-2000 was that it should have a very strong front end. To accomplish this, the design uses a mixer that requires that the LO port be fed with +17 dBm at 50 Ω. Having the test equipment to measure these parameters confirms the expected results.

To help in the construction of this project, the author purchased surplus test instruments including an oscilloscope and signal generator. He also built test equipment such as crystal-controlled, very low power, oscillators and attenuators to measure receiver sensitivity, and high-power oscillators and a combiner to measure receiver blocking and dynamic range. He also built a spectrum analyzer, which turned out to be one of the most useful instruments because the HBR-2000 has 19 band-pass filters and 22 low-pass filters. (This instrument was described in August and September 1998 *QST* by Wes Hayward, W7ZOI, and Terry White, K7TAU.) Later he built the RF power meter featured in June 2001 *QST*, authored by W7ZOI and Bob Larkin, W7PUA. These instruments allow adjustment and measurement of the performance of each module built. A good selection of test equipment can be affordable. The author's lab is shown in **Fig 13.63**.

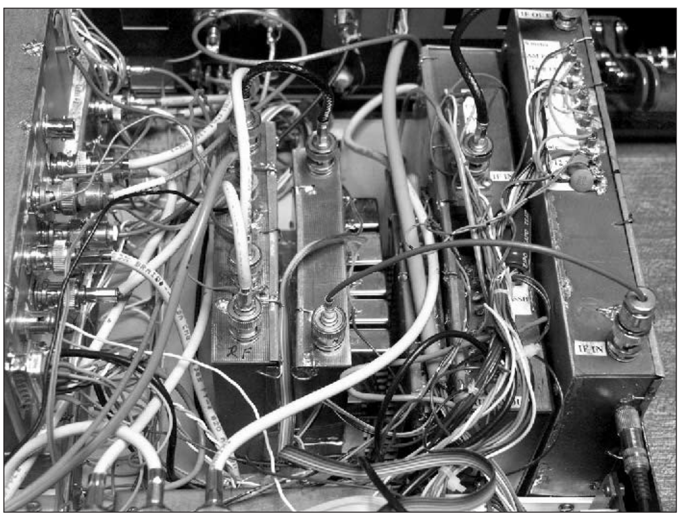


Fig 13.62 — The inside of the HBR-2000. All the boxes are connected together with 50 Ω coax and BNC connectors for the RF runs. All dc power and control leads, in and out of the boxes, are through feedthrough insulators.

Fig 13.63 — The VE7CA work shop shows some of the home built and surplus test equipment used in the construction and evaluation of the HBR-2000.

OTHER CIRCUITS

Here are some of the sources used for deciding on the circuits for the other modules in the HBR-2000 design. The low-distortion audio module is from the "R1 High Performance Direct Conversion Receiver" by Rick Campbell, KK7B, in August 1993 *QST*. The VFO design is from June 1991 *QST* ("Build a Universal VFO" by Doug DeMaw, W1FB).

The first mixer, post-mixer amplifier and crystal heterodyne oscillator design was taken from "A Progressive Communication Receiver" design by Wes Hayward and John Lawson, K5IRK, which appeared in November 1981 *QST* and was featured in *The ARRL Handbook* for many years. This is a classic radio article with many good circuit ideas.

The receiver input RF band-pass filter and diplexer designs along with the noise blanker and 100 W amplifier circuits were taken from the three-part series beginning in the May/June 2000 issue of *QEX* titled The "ATR-2000: A Homemade, High-Performance HF Transceiver" by John Stephenson, KD6OZH. The power amplifier and output circuit filters are in a shielded sub-enclosure as shown in **Fig 13.64** and **Fig 13.65**.

The BFO and power supply circuits were lifted right out of the *ARRL Handbook*. The transmitter portion of the transceiver consists of combinations of various circuits found in *Experimental Method in RF Design*.

RECEIVER SPECIFICATIONS

For the skeptics, the actual measured performance of the HBR-2000, as confirmed in the ARRL Lab, is shown in **Table 13.2**. Receiver sensitivity measurements on all bands are within ±0.5 dB of –130 dBm. All measurements were made with an IF filter bandwidth of 400 Hz. Test oscillators are

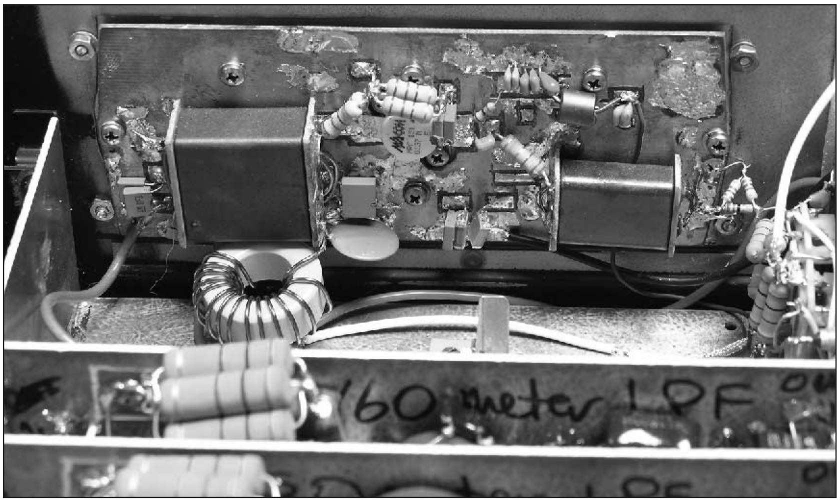


Fig 13.64 — The 100 W amplifier board.

Fig 13.65 — The 100 W amplifier, 10 low-pass filters and the power supplies are located in a separate enclosure.

Table 13.2
HBR-2000 Test Measurements

Spacing:	20 kHz	5 kHz	2 kHz
Two-tone blocking dynamic range:	>126 dB	124 dB	122 dB
Third-order IMD dynamic range:	103.5 dB	102.5 dB	93.0 dB
Third-order intercept:	25.5 dBm	24.0 dBm	14.5 dBm
Image rejection all bands:	>135 dBm		
Receive to transmit time:	8 ms (incl 2 ms click filter)		
CW, full QSK transmit to receive time:	17 ms (30 WPM = 20 ms dot)		

two separately boxed crystal oscillators, low-pass filtered and designed for a 50-Ω output impedance. MDS measurements were made with a HP-8640B signal generator and a true reading RMS voltmeter across the speaker output.

The receiver specifications were made following ARRL procedures as outlined in the ARRL "Lab Test Procedures Manual" at **www.arrl.org/product-review**. Making accurate receiver measurements is not a trivial matter and should be approached with the understanding of the limitations of the test equipment being used and thorough knowledge of the subject.

You may notice that the author made no attempt to miniaturize the HBR-2000. With large knobs and large labeling, he is able to operate the transceiver without the need for reading glasses. When you build your own equipment, you get to decide the front panel layout, what size knobs to use and where they should be located. That is a real bonus!

Why not plan to build your dream station? If you haven't built any Amateur Radio equipment, begin with a small project. As you gain experience you will eventually have the confidence to build more complex equipment. Then someday you too can say with a smile, "my rig here is homebrew."

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