# Modulation Sources (What and How We Communicate)

# 12

"Modulation Sources"—what does that mean? An engineer might simply call this chapter *Baseband.* Engineers use that term to distinguish information before it's used to modulate a carrier. So, this chapter covers the various kinds of information we impress on RF (audio, video, digital, remote control) before that information has been moved to some intermediate frequency (IF) or the desired RF.

Baseband carries the connotation "at or near dc" because the final RF is usually much higher than the baseband frequency, yet that is misleading. For example, a baseband ATV signal usually extends up to 5 MHz, which is hardly dc. Nonetheless, compared to the operating frequencies (52 to 806 MHz for broadcast, 420 MHz and higher for amateur) it *is* practically dc.

Here we will discuss characteristics of the information (such as bandwidth), how we prepare it for transmission, optimize the transfer and process it after reception.

Nearly all of the processing discussed in this chapter can be implemented with the emerging digital-signal-processing (DSP) technology. For example, the CLOVER-II system described later in this chapter uses DSP to vary the modulation scheme as required by propagation conditions. Look to the **Digital Signal Processing** chapter for further information about DSP techniques.

## BANDWIDTH

Whenever information is added to a carrier (we say the carrier is *modulated)*, sidebands are produced. Sidebands are the frequency bands on both sides of a carrier resulting from the baseband signal varying some characteristic of the carrier. The modulation process creates two sidebands: the upper sideband (USB) and the lower sideband (LSB). The width of each sideband is generally equal to the highest frequency component in the baseband signal. In some modulation systems, the width of the sidebands may greatly exceed the highest baseband frequency component.

The USB and LSB are mirror images of each other and carry indentical information. Some modulation systems transmit only one sideband and partially or completely suppress the other in order to conserve bandwidth.

According to FCC Rules, *occupied bandwidth* is:

The frequency bandwidth such that, below its lower and above its upper frequency limits, the mean powers radiated are each equal to 0.5 percent (–23 dB) of the total mean power radiated by a given emission.

In some cases a different relative power level may be specified; for example, –26 dB (0.25%) is used to define bandwidth in §97.3(a)(8) of the FCC rules.

Occupied bandwidth is not always easy for amateurs to determine. It can be measured on a spectrum analyzer, which is not available to most amateurs. Occupied bandwidth can also be calculated, but the calculations require an understanding of the mathematics of information theory and are not covered in this book.

The FCC has defined *necessary bandwidth* as:

For a given class of emission, the minimum value of the occupied bandwidth sufficient to ensure the transmission of information at the rate and with the quality required for the system employed, under specified conditions.

# Voice Modes

## AMPLITUDE MODULATION (AM)

AM voice was the second mode used in Amateur Radio (after Morse code). AM techniques are the basis for several other modes, such as single-sideband voice and AFSK. This material was supplied by Jeff Bauer, WA1MBK.

AM is a mixing process. When RF and AF signals are combined in a standard AM modulator, four output signals are generated: the original RF signal (carrier), the original AF signal, and two sidebands, whose frequencies are the sum and difference of the original RF and AF signals, and whose amplitudes are proportional to that of the original AF signal. The sum component is called the upper sideband (USB). It is direct:

A frequency increase of the modulating AF causes a frequency increase in the RF output. The difference component is called the lower sideband (LSB), which is inverted: A frequency increase in the modulating AF produces a decrease in the output frequency. The amplitude and frequency of the carrier are unchanged by the modulation process, and the original AF signal is rejected by the RF output network. The RF envelope (sum of sidebands and carrier), as viewed on an oscilloscope, has the shape of the modulating waveform.

**Fig 12.1B** shows the envelope of an RF signal that is 20% modulated by an AF sine wave. The envelope varies in amplitude because it is the vector sum of the carrier and the sidebands. A spectrum analyzer or selective receiver will show the carrier to be constant. The spectral photograph also shows that the bandwidth of an AM signal is twice the highest frequency component of the modulating wave.

An AM signal cannot be frequency multiplied without special processing because the phase/frequency relationship of the modulating-waveform components would be severely distorted. For this reason, once an AM signal has been generated, its frequency can be changed only by heterodyning.

All of the information in an AM signal is contained in the sidebands, but two-thirds of the RF power is in the carrier. If the carrier is suppressed in the transmitter and reinserted (in the proper phase) in the receiver, significant advantages accrue. When the reinserted carrier is strong compared to the incoming double-sideband signal (DSB), exalted carrier reception is achieved, and distortion from selective fading is reduced greatly. A refinement called synchronous detection uses a PLL to reject interference. Suppressing the carrier also eliminates the heterodyne interference common with adjacent AM signals. More important, eliminating the carrier increases overall transmitter efficiency. Transmitter power requirements are reduced by 66%, and the remaining 34% has a light duty cycle.

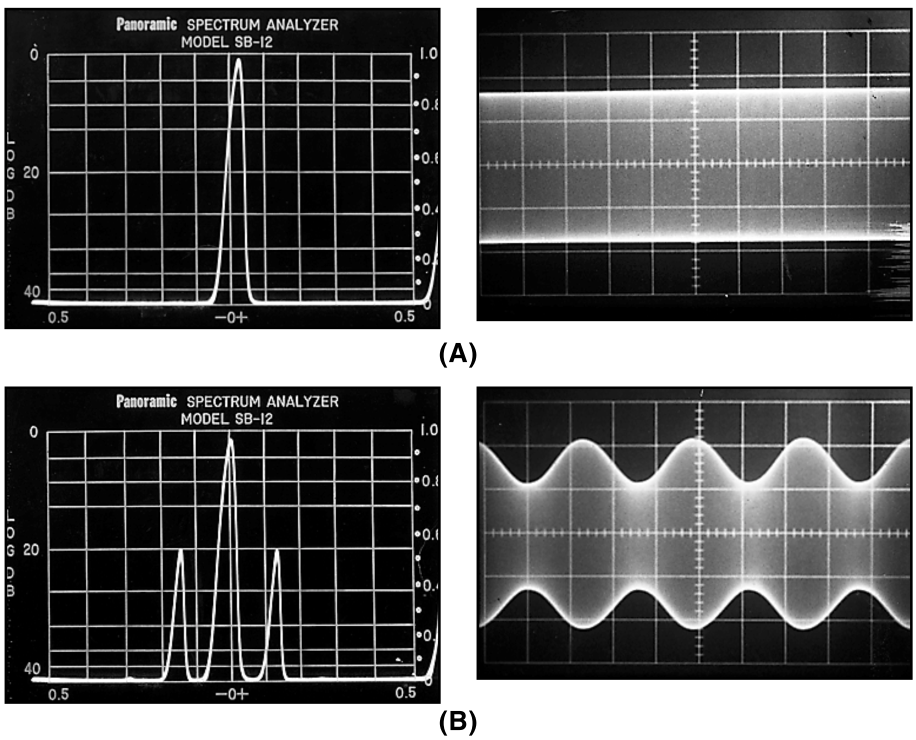



**Fig 12.1—Electronic displays of AM signals in the frequency and time domains. A shows an unmodulated carrier or single-tone SSB signal. B shows a full-carrier AM signal modulated 20% with a sine wave.**

### *Mathematics of AM*

AM can happen at low levels (as in a driver or predriver stages of a transmitter) or high levels, in the final output stage. The numbers are consistent for both methods.

For example, to 100% modulate a 10 W RF carrier, 5 W of clean AF is required from the modulator. Good engineering calls for a 25% overdesign; thus 6.25 W of AF allows plenty of system headroom. The circuitry can then "loaf along" at the 100% level.

### *Overmodulation*

Overmodulation occurs when more audio is impressed on a carrier than is needed for 100% modulation. It is also known as *flattopping*. Overmodulation causes distortion of the information conveyed and produces *splatter* (spurious emissions) on adjacent frequencies. Splatter interferes with others sharing our already-crowded bands: Prevent overmodulation at all times.

Years ago amateurs used ingenious ways to detect and prevent or control overmodulation. Nowadays with solid-state, large-scale integration (LSI) chips and microprocessor control, bullet-proof overmodulation prevention can be designed into transmitters. The most familiar method is called ALC, for automatic level control.

Although modern use of full-carrier AM on the amateur bands is very limited, there is a core group of AM aficionados experimenting with pulse duration modulation (PDM). This system was pioneered in AM broadcast transmitters.

This form of high-level modulation differs from conventional AM in that the PDM modulator operates in switching mode, with audio information contained during on-pulses. The audio amplitude is therefore determined by the duty cycle of the modulator or switching tube. Those interested in further reading on the topic should look in William Orr's (W6SAI) *Radio Handbook* (published by Howard W. Sams and Co) and the *AM Press/Exchange* (for contact information, see the Address List in the **References** chapter).

### *Balanced Modulators*

The carrier can be suppressed or nearly eliminated by using a balanced modulator or an extremely sharp filter. Contemporary amateur transmitters often use both methods.

The basic principle of any balanced modulator is to introduce the carrier in such a way that only the sidebands will appear in the output. The balanced-modulator circuit chosen by a builder depends on constructional considerations, cost and the active devices to be employed.

In any balanced-modulator circuit, there is (theoretically) no output when no audio signal is applied. When audio is applied, the balance is upset, and one branch conducts more than the other. Since any modulation process is the same as "mixing," sum and difference frequencies (sidebands) are generated. The modulator is not balanced for the sidebands, and they appear in the output.

## SINGLE-SIDEBAND (SSB)

A further improvement in communications effectiveness can be obtained by transmitting only one of the sidebands. When the proper receiver bandwidth is used, a single-sideband (SSB) signal will show an effective gain of up to 9 dB over an AM signal of the same peak power. Because the redundant information is eliminated, the required bandwidth of an SSB signal is half that of a comparable AM or DSB emission. Unlike DSB, the phase of the local carrier generated in the receiver is unimportant. **Table 12.1** shows the qualities of a good SSB signal.

## SSB Generation: The Filter Method

If the DSB signal from the balanced modulator is applied to a narrow band-pass filter, one of the

**Table 12.1**
**Guidelines for Amateur SSB Signal Quality**

| *Parameter* | *Suggested Standard* |
|---|---|
| Carrier suppression | At least 40-dB below PEP |
| Opposite-sideband suppression | At least 40-dB below PEP |
| Hum and noise | At least 40-dB below PEP |
| Third-order intermodulation distortion | At least 30-dB below PEP |
| Higher-order intermodulation distortion | At least 35-dB below PEP |
| Long-term frequency stability | At most 100-Hz drift per hour |
| Short-term frequency stability | At most 10-Hz P-P deviation in a 2-kHz bandwidth |

sidebands can be greatly attenuated. Because a filter cannot have infinitely steep skirts, the response of the filter must begin to roll off within about 300 Hz of the phantom carrier to obtain adequate suppression of the unwanted sideband. This effect limits the ability to transmit bass frequencies, but those frequencies have little value in voice communications. The filter rolloff can be used to obtain an additional 20 dB of carrier suppression. The bandwidth of an SSB filter is selected for the specific application. For voice communications, typical values are 1.8 to 3.0 kHz.

**Fig 12.2** illustrates two variations of the filter method of SSB generation. In A, the heterodyne oscillator is represented as a simple VFO, but may be a premixing system or synthesizer. The scheme at B is perhaps less expensive than that of A, but the heterodyne-oscillator frequency must be shifted when changing sidebands in order to maintain dial calibration.

The ultimate sense (direct or inverted) of the final output signal is influenced as much by the relationship of the heterodyne oscillator frequency to the fixed SSB frequency as by the filter or carrier frequency selection. The heterodyne-oscillator frequency must be chosen to allow the best image rejection. This consideration requires that the heterodyne-oscillator frequency be above the fixed SSB frequency on some bands and below it on others. To reduce circuit complexity, early amateur filter-method SSB transmitters used a 9-MHz IF and did not include a sideband switch. The result was that the output was LSB on 160, 75 and 40 m, and USB on the higher bands. This convention persists today, despite the flexibility of most modern amateur SSB equipment. Appropriate filtering methods and filters for SSB generation are discussed in the **Filters** chapter.

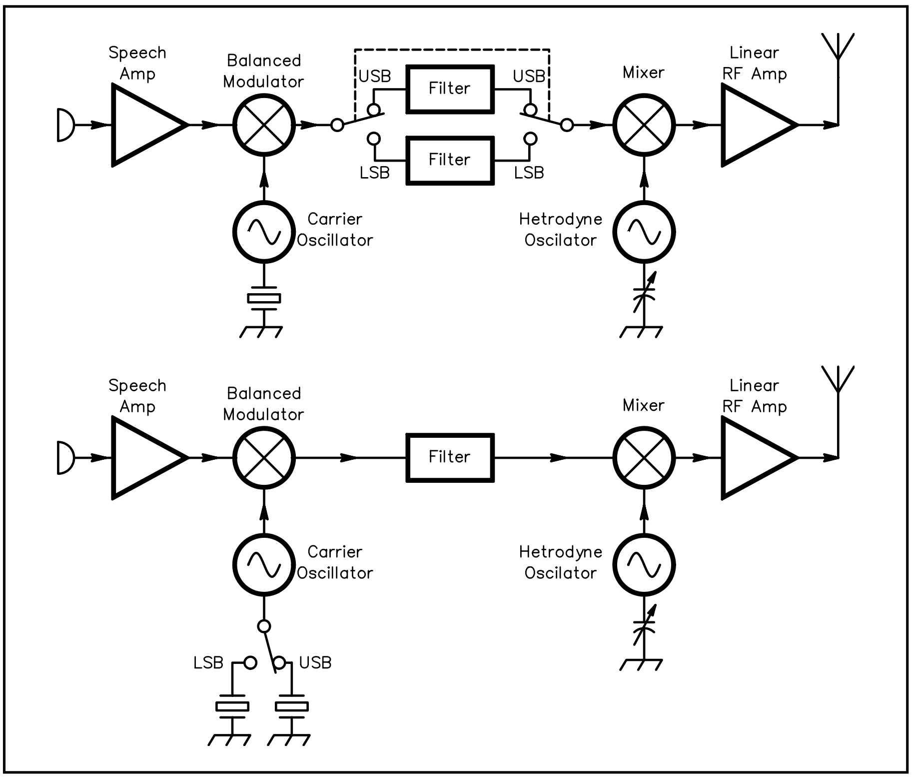



**Fig 12.2—Block diagrams of filter-method SSB generators. They differ in the manner that the upper and lower sideband are selected.**

## SSB Generation: The Phasing Method

**Fig 12.3** shows another method to obtain an SSB signal. The audio and carrier signals are each split into equal components with a 90º phase difference (called *quadrature*) and applied to balanced modulators. When the DSB outputs of the modulators are combined, one sideband is reinforced and the other is canceled. The figure shows sideband selection by means of transposing the audio leads, but the same result can be achieved by switching the carrier leads.

The phase shift and ampli-

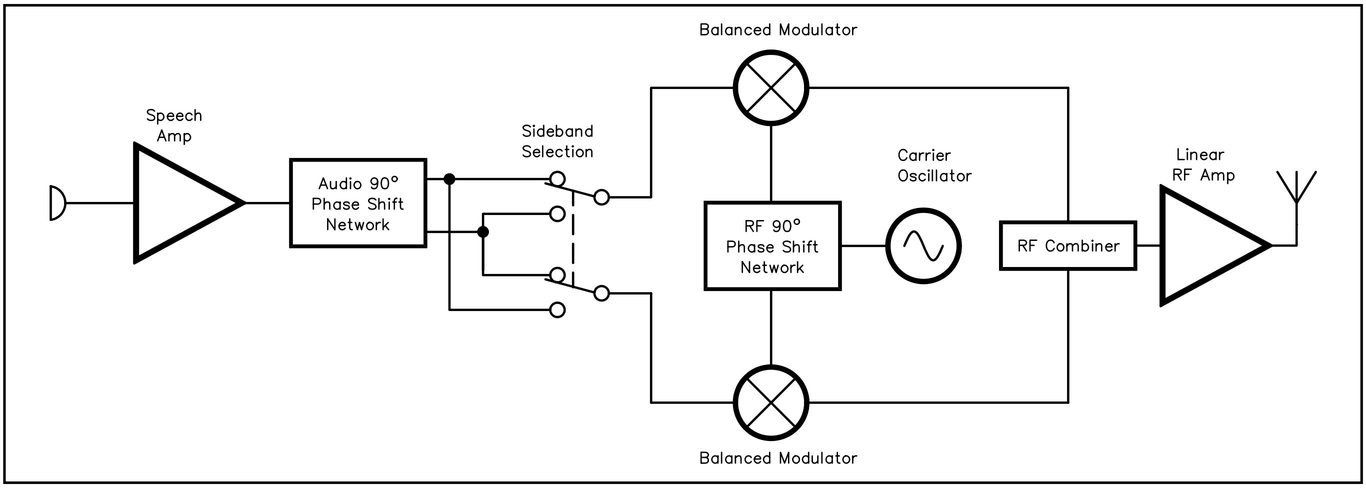



**Fig 12.3—Block diagram of a phasing SSB generator.**

tude balance of the two channels must be very accurate if the unwanted sideband is to be adequately attenuated. **Table 12.2** shows the required phase accuracy of one channel (AF or RF) for various levels of opposite sideband suppression. The numbers given assume perfect amplitude balance and phase accuracy in the other channel.

The table shows that a phase accuracy of ±1° is required to satisfy the criteria tabulated at the beginning of this chapter. It is difficult to achieve this level of accuracy over the entire speech band. Note, however, that speech has a complex spectrum with a large gap in the octave from 700 to 1400 Hz. The phase-accuracy tolerance can be loosened to ±2° if the peak deviations can be made to occur within that spectral gap.

The major advantage of the phasing system is that the SSB signal can be generated at the operating frequency without the need for heterodyning. Phasing can be used to good advantage even in fixed-frequency systems. A loose-tolerance (±4°) phasing exciter followed by a simple two-pole crystal filter can generate a high-quality signal at very low cost.

**Table 12.2**
**Unwanted Sideband Suppression as a Function of Phase Error**

| *Phase Error (deg.)* | *Suppression (dB)* |
|---|---|
| 0.125 | 59.25 |
| 0.25 | 53.24 |
| 0.5 | 47.16 |
| 1.0 | 41.11 |
| 2.0 | 35.01 |
| 3.0 | 31.42 |
| 4.0 | 28.85 |
| 5.0 | 26.85 |
| 10.0 | 20.50 |
| 15.0 | 16.69 |
| 20.0 | 13.93 |
| 30.0 | 9.98 |
| 45.0 | 6.0 |

### *Audio Phasing Networks*

It would be difficult to design a two-port network having a quadrature (90°) phase relationship between input and output with constant-amplitude response over a decade of bandwidth. A practical approach, pioneered by Robert Dome, W2WAM, is to use two networks having a differential phase shift of 90°. This differential can be closely maintained in a simple circuit if precision components are used.

Numerous circuits have been developed to synthesize the required 90° phase shift electronically. Active-filter techniques are used in many of these systems; use precision components for good results.

### *RF Phasing Networks*

If the SSB signal is to be generated at a fixed frequency, the RF phasing problem is trivial; any method that produces the proper phase shift can be used. If the signal is produced at the operating frequency, problems similar to those in the audio networks must be overcome.

A differential RF phase shifter is shown in **Fig 12.4**. The amplitudes of the quadrature signals won't be equal over an entire phone band, but this is of little consequence as long as the signals are strong enough to saturate the modulators.

Where percentage bandwidths are small, such as in the 144.1- to 145-MHz range, the RF phase shift can be obtained conveniently with transmission-line methods. If one balanced-modulator feed line is made an electrical quarter wavelength longer than the other, the two signals will be 90º out of phase. (It is important that the cables be properly terminated.)

One method for obtaining a 90º phase shift over a wide bandwidth is to generate the quadrature signals at a fixed frequency and heterodyne them individually to any desired operating frequency. Quadrature hybrids having multioctave bandwidths are manufactured commercially, by Mini-Circuits Labs and others.

Another practical approach uses two VFOs in a master-slave PLL system. Many phase detectors lock the two signals in phase quadrature. A doubly balanced mixer also has this property. One usually thinks of a PLL as having a VCO locked to a reference signal, but a phase differential can be controlled independently of the oscillator. The circuit in **Fig 12.5** illustrates this principle. Two digital phase shifters are sketched in **Fig 12.6**. If ECL ICs are used, this system can work over the entire HF spectrum.

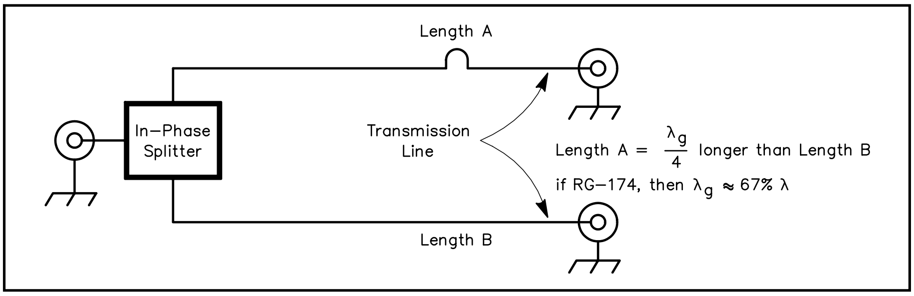



**Fig 12.4—A simple RF phase shifter using transmission lines. It is practical at VHF and UHF. Examples: If L1 and L2 are made from RG-174, L1 is 69.1 inches longer than L2 at 28.5 MHz. 13.7 inches longer at 144.2 MHz and 8.86 inches longer at 222.1 MHz.**

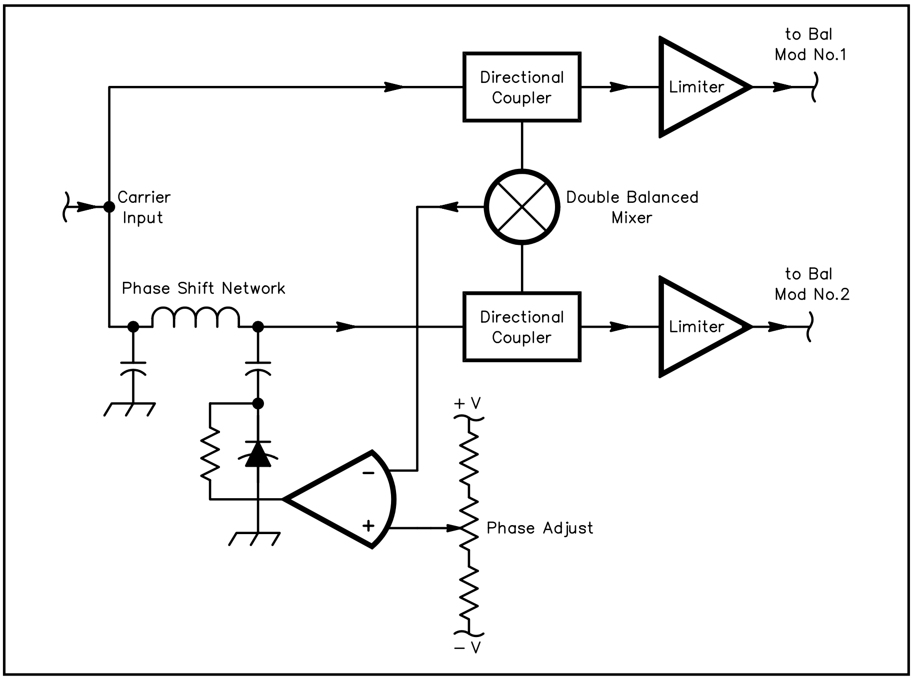



**Fig 12.5—A block diagram of a PLL phase-shifting system that can maintain quadrature (90° phase difference) over a wide frequency range. The doubly balanced mixer is used as a phase detector.**

### *Independent-Sideband (ISB)*

If two SSB exciters, one USB and the other LSB, share a common carrier oscillator, two channels of information can be simultaneously transmitted from one antenna. Methods for ISB generation in filter and phasing transmitters are shown in **Fig 12.7**.

The most obvious amateur application for ISB is the transmission of SSTV with simultaneous audio commentary. On the VHF bands, other combinations are possible, such as voice and code or SSTV and RTTY.

## Amplitude Compandored Single Sideband (ACSSB)

When SSB was tried in the Land Mobile Service, several problems arose. One was that users (who are

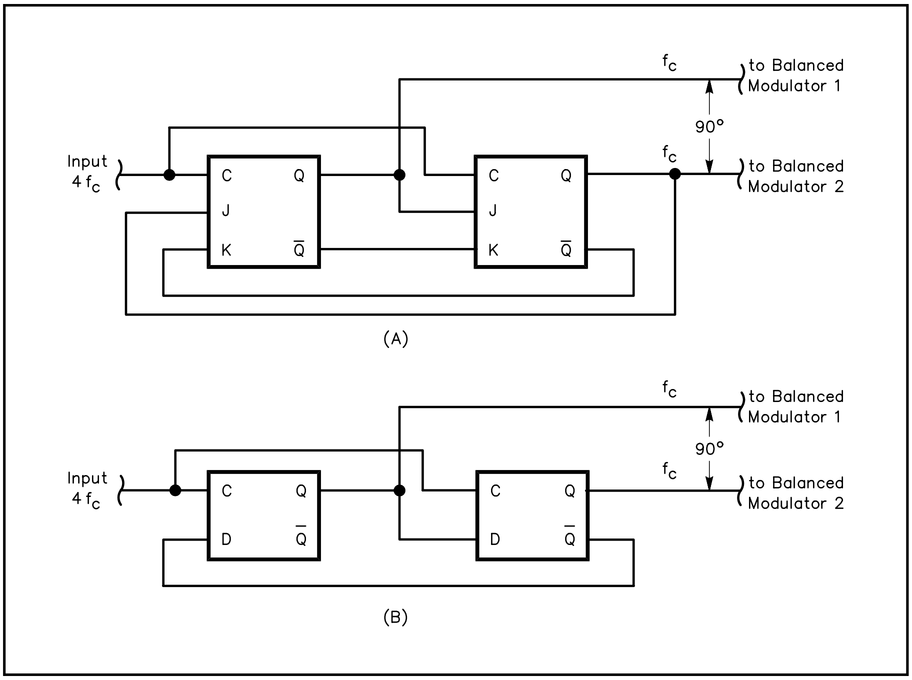



**Fig 12.6—Digital RF phase-shift networks. Circuit A uses JK flip-flops; B uses D flip-flops. The carrier frequency must be quadrupled before processing.**

not trained operators) couldn't master the control known as CLARIFIER to land-mobile users or receiver incremental tuning, RIT, to amateurs. In addition, users were annoyed by SSB's fading and noise performance, compared to that of FM.

So, to get the spectrum savings of SSB over FM, Land-Mobile Service engineers came up with a form of SSB that satisfied the users accustomed to FM. At present, there is almost no amateur use of this modulation.

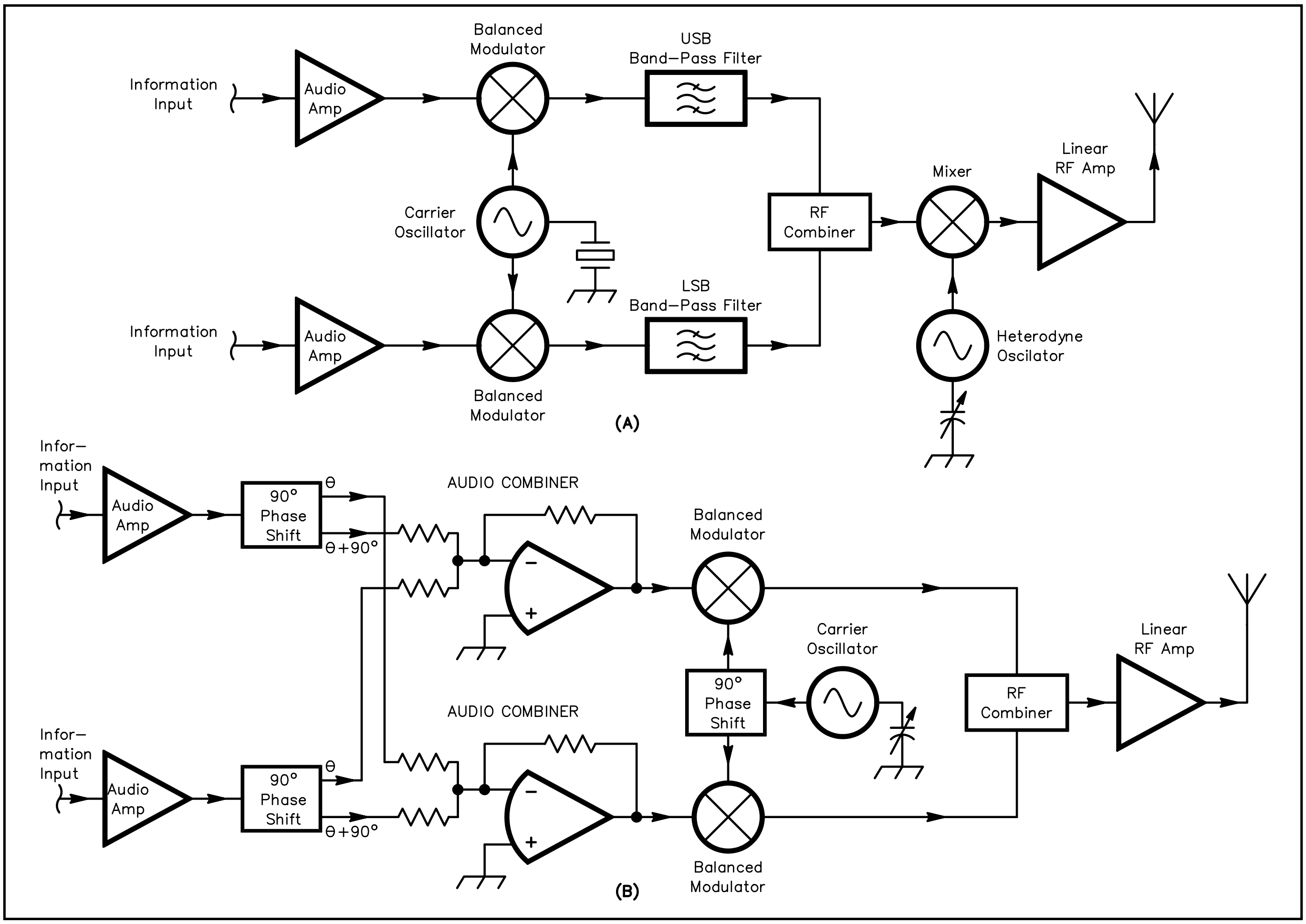



**Fig 12.7—Independent-sideband generators. A shows a filter system, B a phasing system. The "RF combiner" may be either a hybrid combiner or a summing amplifier.**

## FREQUENCY MODULATION (FM)

When the frequency of the carrier is varied in accordance with the variations in a modulating signal, the result is frequency modulation (FM). Varying the phase of the carrier current is called phase modulation (PM). Frequency and phase modulation are not independent, since the frequency cannot be varied without also varying the phase, and vice versa. This section was written by Dean Straw, N6BV.

The primary advantage of FM is its ability to produce a high signal-to-noise ratio when receiving a signal of only moderate strength. This has made FM popular for mobile communications services and high-quality broadcasting. However, because of the wide bandwidth required and the distortion suffered in skywave propagation, the use of FM has generally been limited to frequencies higher than 29 MHz.

When compared to AM or SSB, FM has some impressive advantages for VHF operation. In an FM transmitter, modulation takes place in a low-level stage. Amplifiers following the modulator can be operated Class C for best efficiency, since operation need not be linear. The frequency tolerances needed for channelized FM operation are much less severe than for SSB, helping to keep cost down.

The effectiveness of FM and PM for communication purposes depends almost entirely on the methods used for receiving. If the FM receiver responds to frequency and phase changes but is insensitive to amplitude changes, it can discriminate against many forms of noise.

**Fig 12.8** is a representation of frequency modulation. When an audio modulating signal is applied, the carrier frequency is increased during one half cycle of the modulating signal and decreased during the half cycle of opposite polarity. In this figure RF cycles occupy less time (higher frequency) when the modulating signal is positive, and more time (lower frequency) when the modulating signal is negative. The change in the carrier frequency is called *frequency deviation* and is proportional to the instantaneous amplitude of the modulating signal. The deviation is small when the instantaneous amplitude of the modulating signal is small and is greatest when the modulating signal reaches its peak, either positive or negative.

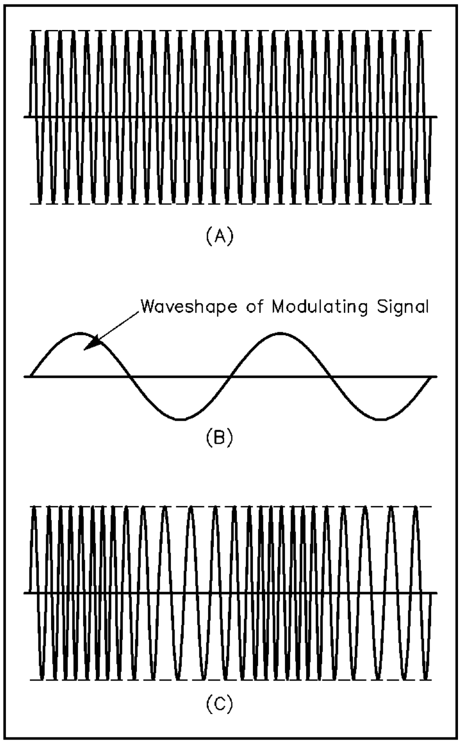



**Fig 12.8—Graphical representation of frequency modulation. In the unmodulated carrier (A) each RF cycle occupies the same amount of time. When the modulating signal (B) is applied, the radio frequency is increased and decreased according to the amplitude and polarity of the modulating signal (C).**

### *Phase Modulation (PM)*

If the phase of the current in a circuit shifts there is an instantaneous frequency change during the time the phase is shifting. The amount of frequency change is directly proportional to how rapidly the phase is shifting and to the total amount of the phase shift. The rapidity of the phase shift is directly proportional to the frequency of the modulating signal. In a properly operating PM system, the amount of phase shift is proportional to the instantaneous amplitude of the modulating signal. Phase modulators have a built-in *preemphasis*, where deviation increases with modulating frequency.

## FM AND PM SIDEBANDS

The sidebands generated by FM and PM occur at integral multiples of the modulating frequency on either side of the carrier. This is in contrast to AM, where a modulating frequency will produce a single set of sidebands on either side of the carrier frequency. An FM or PM signal therefore inherently occupies a wider channel than AM. The number of additional sidebands that occur in FM and PM depend on the relationship between the modulating frequency and the frequency deviation. The ratio between the frequency deviation, in hertz, and the modulating frequency, also in hertz, is called the *modulation index.*

$$\chi = \frac{D}{m} = \phi \qquad (1)$$

where

$\chi$ = modulation index

D = peak deviation ($^1/_2$ difference between maximum and minimum frequency)

m = modulation frequency in hertz

$\phi$ = phase deviation in radians (a radian = 180°/$\pi$ or approximately 57.3°).

For example, the maximum frequency deviation in an FM transmitter is 3000 Hz either side of the carrier frequency. The modulation index when the modulation frequency is 1000 Hz is 3000/1000 = 3.0. At the same deviation with 3000 Hz modulation, the index would be 1; at 100 Hz it would be 30 and so on.

Given a constant input level to the modulator, in PM the modulation index is constant regardless of the modulating frequency. In FM it varies with the modulating frequency, as shown above. In an FM system the ratio of the maximum carrier-frequency deviation to the highest modulating frequency used is called the *deviation ratio*. Thus

$$\text{deviation ratio} = \frac{D}{M} \qquad (2)$$

where

D = peak deviation

M = maximum modulation frequency in hertz.

The deviation ratio used above 29 MHz for narrow-band FM is 5000 Hz (maximum deviation) divided by 3000 Hz (maximum modulating frequency) or 1.67.

**Fig 12.9** shows how the amplitudes of the carrier and the various sidebands vary with the modulation index for single-tone modulation. The first pair of sidebands are displaced from the carrier by an amount equal to the modulating frequency, the second by twice the modulating frequency, and so on. For example, if the modulating frequency is 2000 Hz and the carrier frequency is 29,500 kHz, the first sideband pair is

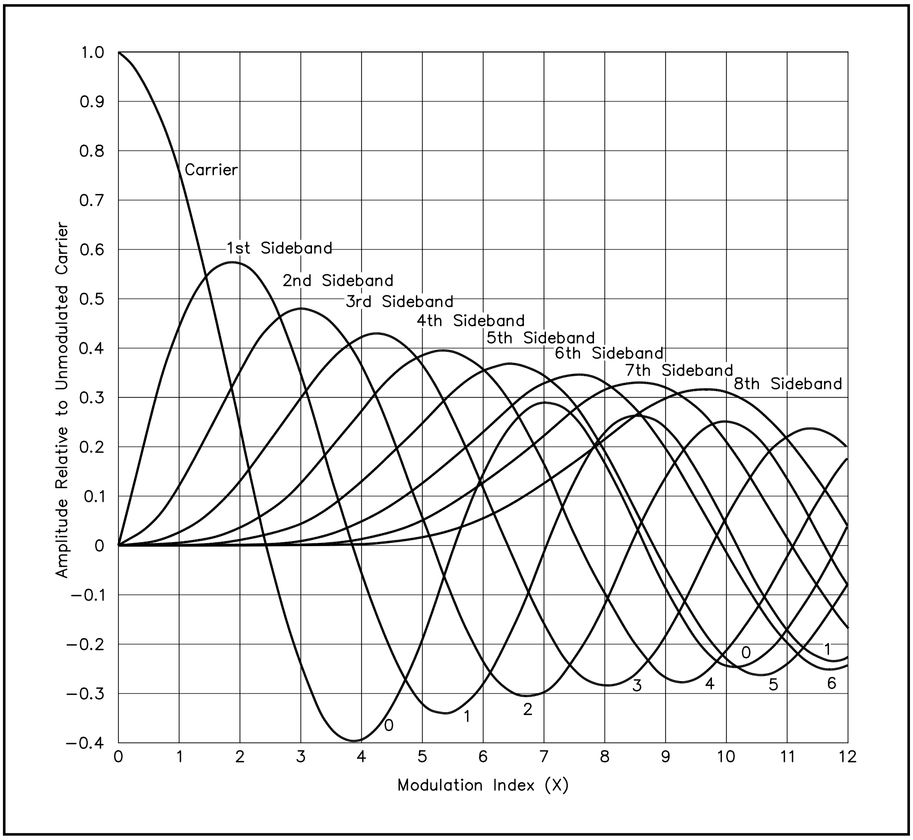



**Fig 12.9—Amplitude variation of the carrier and sideband pairs with modulation index. This is a graphical representation of mathematical functions developed by F. W. Bessel. Note that the carrier completely disappears at modulation indexes of 2.405, 5.52 and 8.654.**

at 29,498 and 29,502 kHz, the second pair is at 29,496 and 29,504 kHz, the third at 29,494 and 29,506 kHz, and so on. The amplitudes of these sidebands depend on the modulation index, not on the frequency deviation.

The carrier strength varies with the modulation index—at a modulation index of 2.405, the carrier disappears entirely. As the index is raised further, the carrier level becomes negative, since its phase is reversed compared to the phase without modulation. In FM and PM the energy going into the sidebands is taken from the carrier—the total power remains the same regardless of the modulation index. Since there is no change in amplitude with modulation, an FM or PM signal can be amplified without distortion by an ordinary Class-C amplifier, either as a straight-through amplifier or frequency-multiplier stage.

If the modulated signal is passed through one or more frequency multipliers, the modulation index is multiplied by the same factor as the carrier frequency. For example, if modulation is applied on 3.5 MHz and the final output is on 28 MHz, the total frequency multiplication is eight times. If the frequency deviation is 500 Hz at 3.5 MHz, it will be 4000 Hz at 28 MHz. Frequency multiplication offers a means for obtaining practically any desired amount of frequency deviation, whether or not the modulator itself is capable of giving that much deviation without distortion.

If the modulation index (with single-tone modulation) does not exceed 0.6 or 0.7, the most important extra sideband, the second, will be at least 20 dB below the unmodulated carrier level. This represents an effective channel width about equal to an AM signal. The energy in speech is distributed among many audio frequencies. On average, the modulation index for any one frequency component is smaller than that for a single audio tone having the same peak amplitude. Thus, the effective modulation index for speech can be somewhat higher while retaining the same average bandwidth. The rule-of-thumb for determination of bandwidth requirements for an FM system using *narrow-band* (5 kHz deviation) modulation is

$$B_n = 2\,(M + D) \tag{3}$$

where

$B_n$ = necessary bandwidth in hertz
M = maximum modulation frequency in hertz
D = peak deviation in hertz.

For narrow-band FM, the bandwidth equals $2 \times (3000 + 5000) = 16000$ Hz. Additional bandwidth may be needed to compensate for cumulative errors in the transmitter and receiver frequencies.

### *FM vs Phase Modulation (PM)*

FM cannot be applied to an amplifier stage, but phase modulation (PM) can. PM is therefore readily adaptable to transmitters employing oscillators of high stability, such as the crystal-controlled oscillators. The amount of phase shift that can be obtained with good linearity yields a maximum practicable modulation index of about 0.5. Because phase shift is proportional to the modulating frequency, this index can be used only at the highest frequency present in the modulating signal, assuming that all frequencies will at one time or another have equal amplitudes.

The frequency response of the speech-amplifier system above 3000 Hz must be sharply attenuated to prevent splatter on adjacent channels. Due to its inherent preemphasis, PM received on an FM receiver sounds “tinny.” The audio must be processed for PM to have the same modulation-index characteristic as an FM signal. The speech-amplifier frequency-response curve is thus shaped so the output voltage is inversely proportional to frequency over most of the voice range. When this is done the maximum modulation index can only be used below a relatively low audio frequency, perhaps 300 to 400 Hz in voice transmission, and must decrease in proportion to an increase in frequency. The net result is that the maximum linear frequency deviation is only one or two hundred hertz. In order to increase the deviation up to narrowband level, we must typically multiply the frequency by eight or more.

## GENERATING FM

### Direct FM

A simple circuit for producing *direct FM* in amateur transmitters is the *reactance modulator*. An active device is connected to the RF tank circuit of an oscillator to act as a variable inductance or capacitance. **Fig 12.10A** is a representative circuit using a MOSFET. This modulator acts as though an inductance were connected across the tank. The frequency increases in proportion to the amplitude of the current in this modulator. If the modulated oscillator is free running, it must usually be operated on a relatively low frequency to maintain good carrier stability. Fig 12.10B shows how a varactor may be used to FM a crystal oscillator directly. The supply voltage for either modulator and oscillator should be regulated to reduce residual FM. The oscillator frequency is multiplied up to the final output frequency.

In many modern frequency-synthesized transceivers, a VCO used in one of the phase-locked loops (PLL) is often frequency modulated directly. A PLL consists of a phase detector, a filter, a dc amplifier and a voltage-controlled oscillator (VCO). See **Fig 12.11**. The VCO runs at a frequency close to that desired when the loop is in lock. The phase detector produces an error voltage if any frequency difference exists between the VCO divided by the variable divider N and the reference signal. The error voltage is applied to the VCO to keep it locked on the carrier frequency when there is no modulation present. The loop bandwidth of the PLL is made narrow enough so that the audio can change the VCO frequency, while the PLL

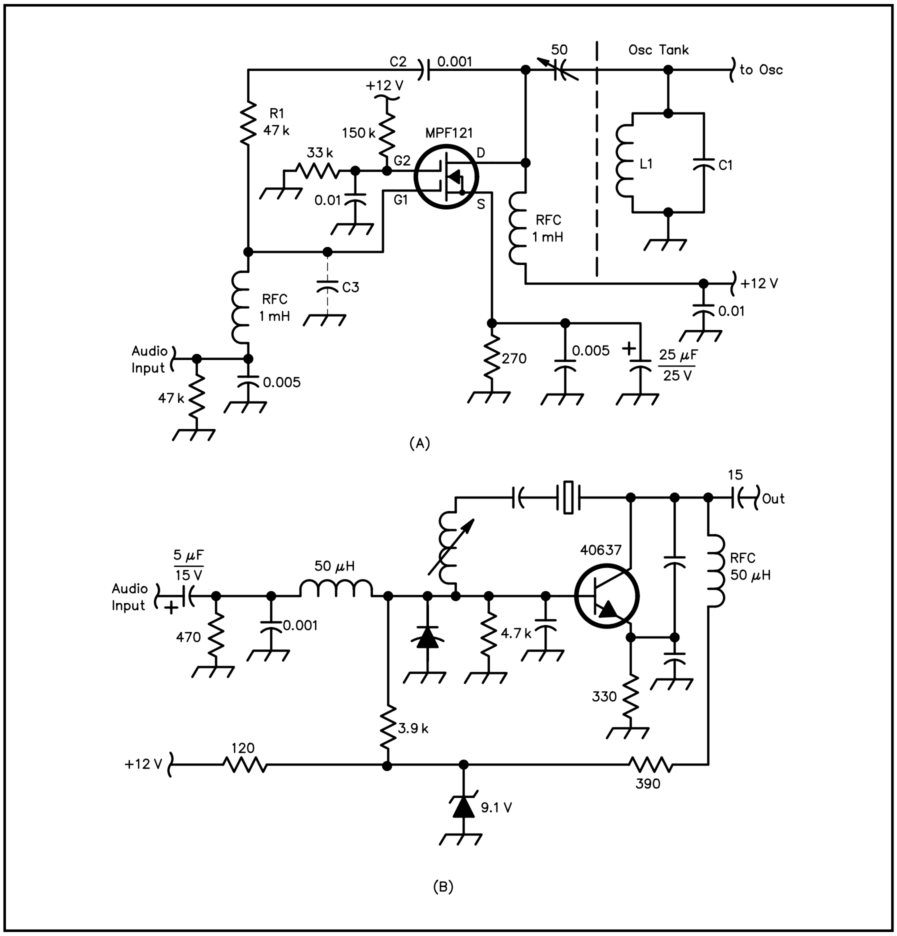



Fig 12.10—At A, reactance modulator using a high-transconductance MOSFET. At B, reactance modulator using a varactor diode.

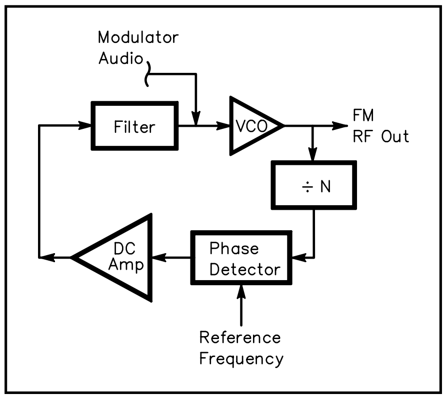



**Fig 12.11—Simple phase-locked loop (PLL) where VCO is FMed directly. The loop filter is designed to be narrow enough so that the loop will lock onto the desired channel frequency, while audio frequencies will modulate the VCO outside the loop bandwidth.**

still keeps the unmodulated carrier frequency on-channel.

### *Indirect FM*

The same type of reactance-modulator circuit used to vary the tuning of an oscillator tank in direct FM can be used to vary the tuning of an amplifier tank. See **Fig 12.12A**. This varies the phase of the tank current to create phase modulation. When audio shaping is used in the speech amplifier, an FM-compatible signal will be generated by the phase modulator. The phase shift that occurs when a circuit is detuned from resonance depends on the amount of detuning and the Q of the circuit. The higher the Q, the smaller the amount of detuning needed to secure a given number of degrees of phase shift. Since reactance modulation of an amplifier stage results in simultaneous amplitude modulation, this must be eliminated using succeeding Class-C limiting stages.

### *Speech Processing for FM*

Several forms of speech processing produce worthwhile improvements in FM system performance. The peak amplitude of the audio signal applied to an FM or PM modulator should be limited so that transmitter cannot be driven into overdeviation. Peak limiting is often maintained using a simple audio clipper between the speech amplifier and modulator. An audio low-pass filter with a cut-off frequency between 2.5 and 3 kHz eliminates harmonics produced by the clipper. Since excessive clipping can cause severe distortion of a voice signal, a more effective audio processor consists of a compressor followed by a clipper and low-pass filter.

An audio shaping network called *preemphasis* is added to an FM transmitter to attenuate the lower audio frequencies, spreading out the energy evenly in the audio band. Preemphasis applied to an FM transmitter gives the deviation characteristic of PM. The reverse process, called *deemphasis*, is used at the receiver to restore the audio to its original relative proportions. See Fig 12.12B and C.

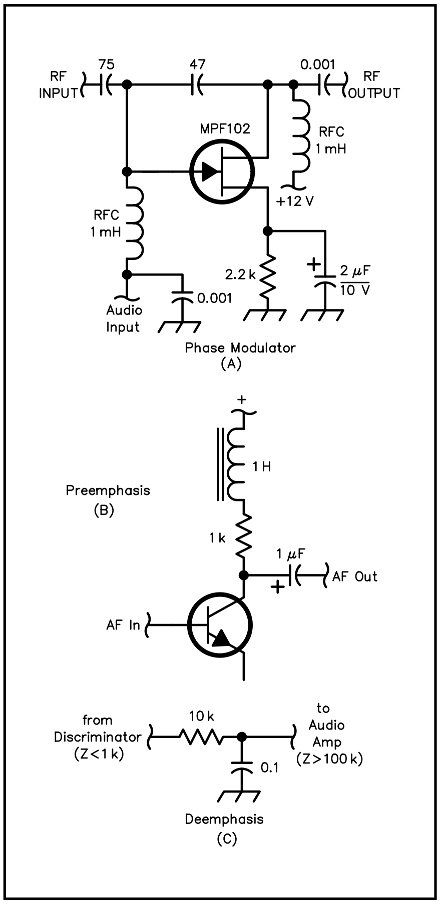



Fig 12.12— At A, a phase-shifter type of phase modulator. At B, preemphasis and at C, deemphasis circuits.

## RECEPTION OF FM SIGNALS

A block diagram of an FM receiver is shown in **Fig 12.13B**. The

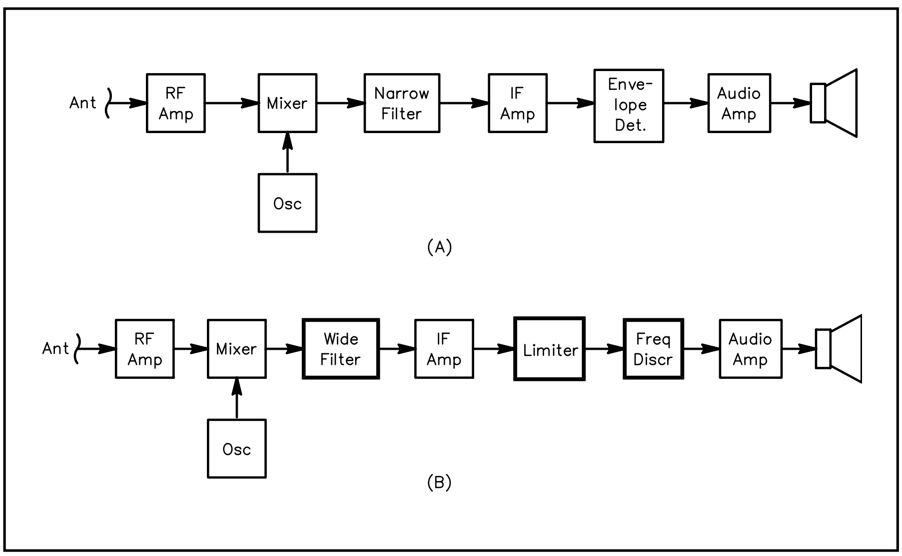



**Fig 12.13—At A, block diagram of an AM receiver. At B, an FM receiver. Dark borders outline the sections that are different in the FM set.**

FM receiver employs a wide-bandwidth filter and an FM detector, and has one or more limiter stages between the IF amplifier and the FM detector. The limiter and discriminator stages in an FM set can eliminate a good deal of impulse noise, except noise that manages to acquire a frequency-modulation characteristic.

FM receivers exhibit a characteristic known as the *capture effect* when QRM is present. The loudest signal received, even if it is only two or three times stronger than other stations on the same frequency, will be the only transmission demodulated.

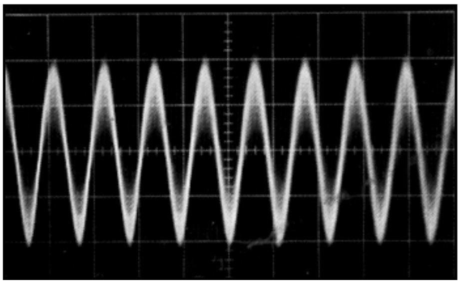
### *Limiters*

The circuit in the FM receiver that has the task of chopping noise and amplitude modulation from an incoming signal is the *limiter*. Most types of FM detectors respond to both frequency and amplitude variations of the signal. Thus, the limiter stages preceding the detector are included so only the desired frequency modulation will be demodulated. This action can be seen in **Fig 12.14**.

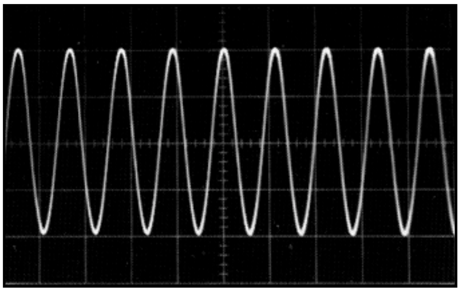
For an amplifier to act as a limiter, the applied voltages are chosen so that the stage overloads predictably, even with a small amount of signal input. Limiting action in an FM receiver should start with an RF input of 0.2 µV or less, so a large amount of gain is required between the antenna terminal and the limiter stages. ICs offer simplification of the IF system, as they pack a lot of gain into a single package.

When sufficient signal arrives at the receiver to start limiting action, the set *quiets*—that is, the background noise disappears. The sensitivity of an FM receiver is rated in terms of the amount of input signal required to produce a given amount of quieting, usually 20 dB. Modern receivers achieve 20 dB quieting with 0.15 to 0.5 µV of input signal.

**Fig 12.15A** shows a two-stage limiter using discrete transistors. The base bias on either transistor may be varied to provide limiting at a desired level. The input-signal voltage required to start limiting action is called the *limiting knee*. This refers to the point at which collector current ceases to rise with increased input signal. Modern ICs have limiting knees of 100 mV for the circuit shown in Fig 12.15B, using the RCA CA3028A or Motorola MC1550G, or 200 mV for the MC1590G of Fig 12.15C. Because high-gain ICs contain many active stages a single IC can provide superior limiting performance compared to most discrete designs.



(A)



(B)

**Fig 12.14—At A, input wave form to a limiter stage shows AM and noise. At B, the same signal, after passing through two limiter stages, is devoid of AM components.**

### *Detectors*

The first FM detector to gain popularity was the *frequency discriminator*. The characteristic of such a detector is shown in **Fig 12.16**. When the FM signal has no modulation, and the carrier is at point 0, the detector has no output. When audio input to the FM transmitter swings the signal higher in frequency, the rectified output increases in the positive direction. When the frequency swings lower, the output amplitude increases in the negative direction. Over a range where the discriminator is linear (shown as

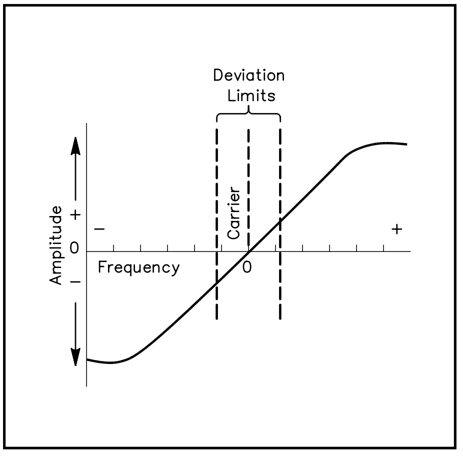



**Fig 12.16—The characteristic of an FM discriminator.**

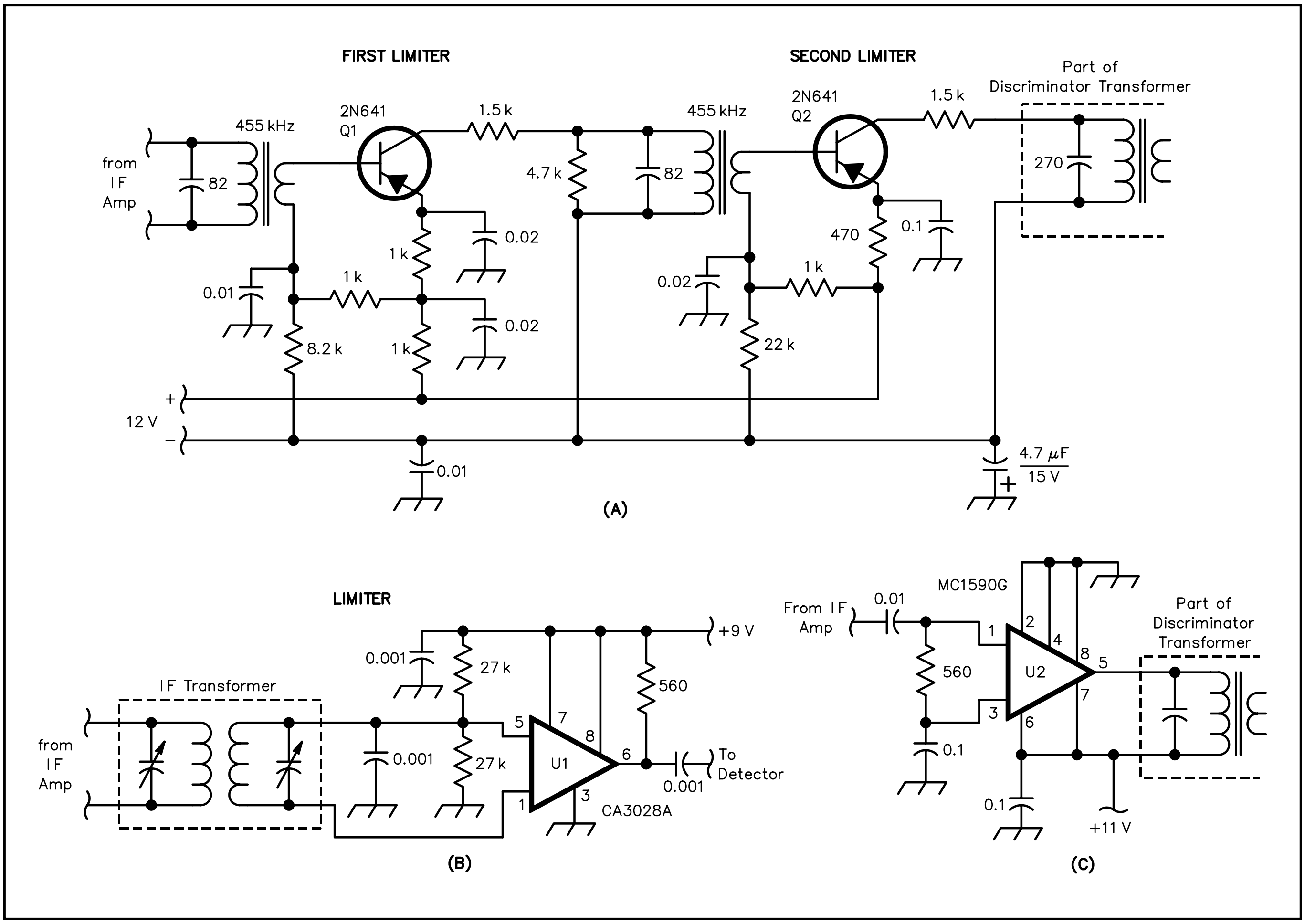



**Fig 12.15—Typical limiter circuits using (A) transistors, (B) a differential IC, (C) a high-gain linear IC.**

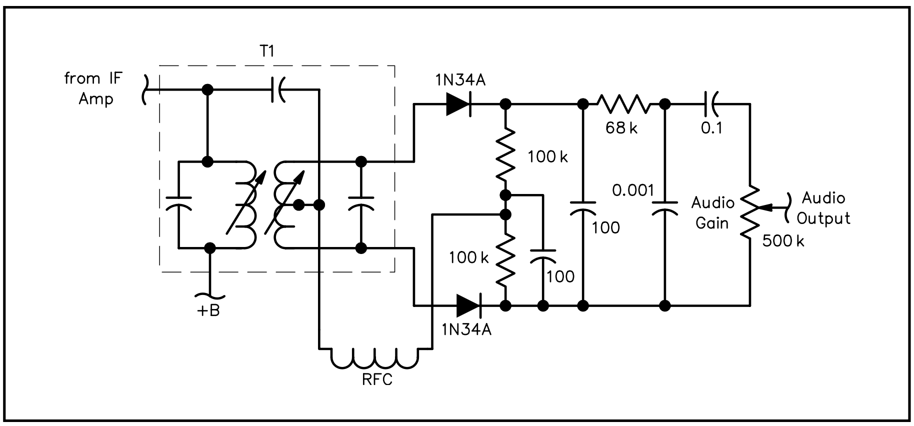



**Fig 12.17—Typical frequency-discriminator circuit used for FM detection. T1 is a Miller 12-C45.**

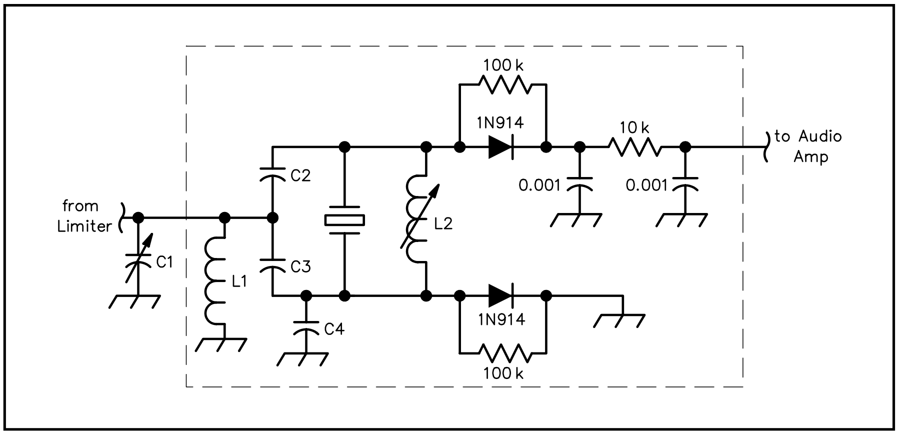



**Fig 12.18—A crystal discriminator, C1 and L1 are resonant at the IF. C2 is equal in value to C3. C4 corrects any circuit imbalance so equal amounts of signal are fed to the detector diodes.**

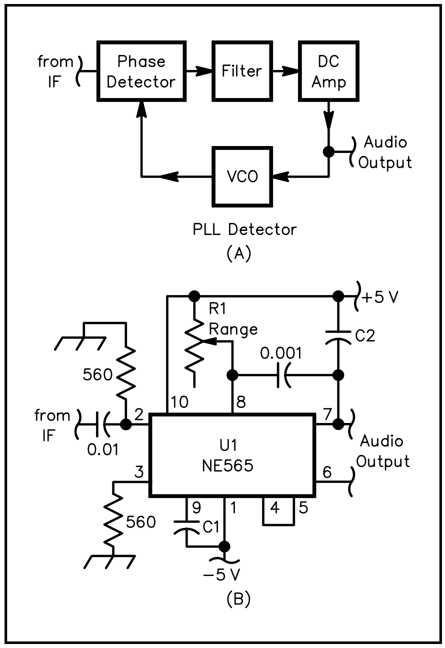



**Fig 12.19—At A, block diagram of a PLL demodulator. At B, complete PLL circuit.**

the straight portion of the line), the conversion of FM to AM will also be linear. A practical discriminator circuit is shown in Fig 12.17.

### *Other Detector Designs*

The difficulties often encountered in building and aligning LC discriminators have inspired research that has resulted in a number of adjustment-free FM detector designs. The crystal discriminator utilizes a quartz resonator, shunted by an inductor, in place of the tuned-circuit secondary used in a discriminator transformer. A typical circuit is shown in **Fig 12.18**.

### *The PLL*

The *phase-locked loop* (PLL) has made a significant impact on transmitter and receiver design, both for frequency generation and for modulation/demodulation. It can act as an FM detector in a process similar to that used for direct-frequency modulation in a transmitter PLL. As the VCO tracks the frequency of an incoming signal, the voltage at the phase detector output becomes demodulated audio. See **Fig 12.19**.

# Text (Digital) Modes

## MORSE TELEGRAPHY (CW)

Telegraphy by on-off keying (OOK, or amplitude-shift keying ASK) of a carrier is the oldest radio modulation system. It is also known as CW (for continuous wave). While CW is used by amateurs and other communicators to mean OOK telegraphy by Morse code, parts of the electronics industry use CW to signify an unmodulated carrier.

This discussion centers on aural reception of CW, but computers are used to send and receive CW as well. A table of characters and their Morse equivalents appears in the **References** chapter.

### TRANSMITTING

#### WPM vs Bauds

The speed of Morse telegraphy is usually expressed in WPM, rather than bauds, which are the common measure in other digital modes. The following formulas relate WPM to bauds:

$$\text{WPM} = 2.4 \times \text{dot/s} \tag{4}$$

$$\text{WPM} = 1.2 \times \text{B} \tag{5}$$

where

WPM = telegraph speed in words per minute

2.4 = a constant calculated by comparing dots per second with plain language Morse code sending the word "PARIS"

1.2 = a constant calculated by comparing the signaling rate in bauds with plain-language Morse code sending the word "PARIS"

B = telegraph speed in bauds.

Thus a keying speed of 25 dot/s or 50 bauds is equal to 60 WPM.

#### Rise Time vs Bandwidth

Keying a carrier on and off produces double (upper and lower) sidebands corresponding to the period of the keying pulse. A string of dits at 50 baud will have sidebands at multiples of 25 Hz above and below the carrier. The rise time of the pulses affects the distribution of power among the sidebands. As rise time increases or pulse rate decreases, the bandwidth of the signal decreases. In addition, the rise time affects how our ears hear the signal.

League publications have long promoted 5-ms rise and fall times for CW keying envelopes (see **Fig 12.20**). This shape is based on an assumed necessary bandwidth of 150 Hz

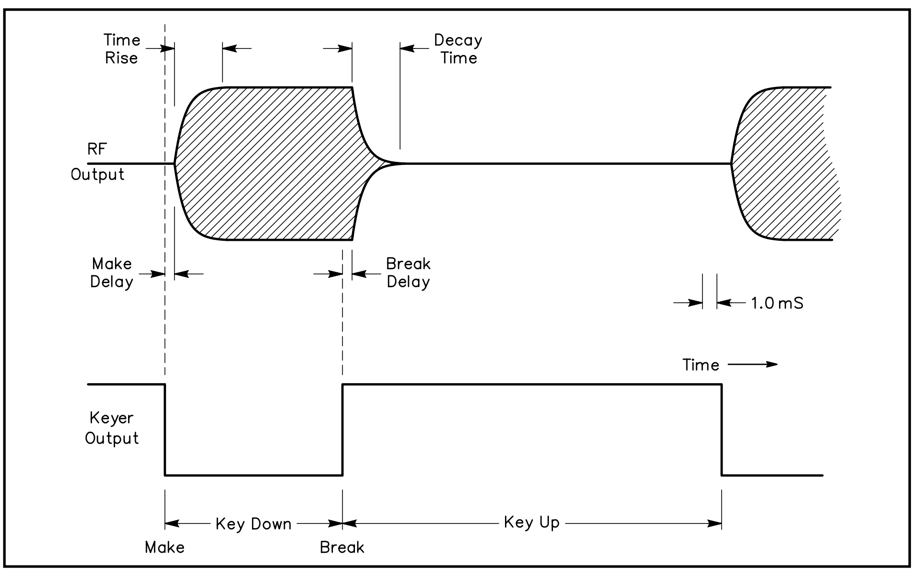



**Fig 12.20—Optimum CW keying waveforms.**

for a 60-WPM (50-baud) pulse train and an equation relating necessary CW bandwidth to keying speed in Appendix 6 of the CCIR Radio Regulations and §2.202 of the FCC rules. The relationship is shown in **Fig 12.21.**

K is part of the bandwidth equation. Low K values produce softer keying, while high values sound harder. The CCIR and FCC recommend K = 3 for nonfading circuits and K = 5 for fading circuits with aural reception. K = 3, is the minimum used for comfortable aural reception, but K = 1 is useful for machine recognition.

Given-a 150-Hz bandwidth, how fast can we communicate over a fading path? With an occupied bandwidth of 150 Hz and K = 5, Fig 12.21 yields 36 WPM. Therefore, 5-ms rise and fall times are suitable for up to 36 WPM on fading circuits and 60 WPM on nonfading circuits.

## RECEIVING

For aural reception, a Morse-code OOK RF signal is not completely demodulated to its original dc pulse, because only thumping would be heard. Instead, the signal is moved (by mixing) down to AF, usually near 700 Hz.

Proper reception of a Morse-code transmission requires that the receiver bandwidth be at least that of the necessary bandwidth plus any frequency error. Thus, if you have 150-Hz receiver bandwidth, it would be necessary for you to carefully tune your receiver to receive a 150-Hz-bandwidth transmission. In practice, it is common to use 500 or 250-Hz IF filters.

Many operators find that it is easier to distinguish between multiple signals as the frequency of the desired signal is lowered to 500 Hz or less. Some modern transceivers provide a CW OFFSET adjustment to accommodate this preference. The same result can be achieved by adjusting the RIT control, although the audio from incoming signals will no longer match the sidetone with this technique.

Those who desire a narrower bandwidth often use audio filters, either an op-amp audio-peak filter or a low-pass switched-capacitor. Such filters may be part of the radio or added as accessories. Look in the **Filters** chapter for projects.

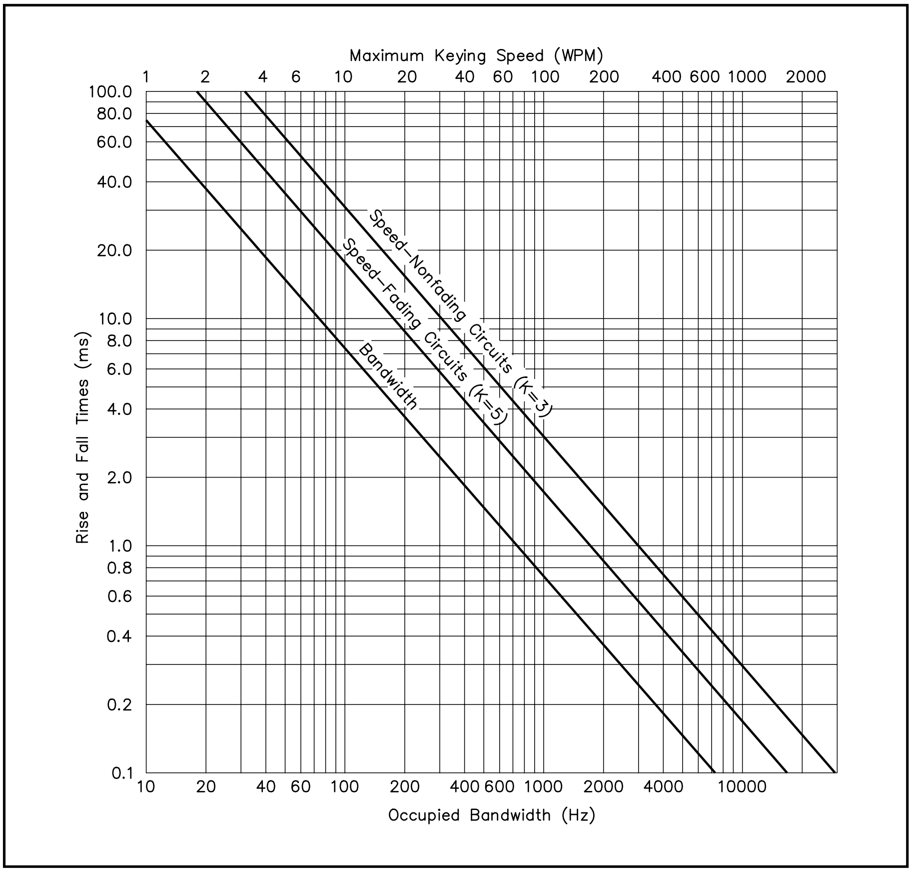



**Fig 12.21—Keying speed vs rise and fall times vs bandwidth for fading and nonfading communications circuits. For example, to optimize transmitter timing for 25 WPM on a nonfading circuit, draw a vertical line from the WPM axis to the K = 3 line. From there draw a horizontal line to the rise/fall time axis (approximately 15 ms). Draw a vertical line from where the horizontal line crosses the bandwidth line and see that the bandwidth will be about 60 Hz.**

# BAUDOT (ITA2) RADIOTELETYPE

## The Baudot Code: ITA2

One of the first data communications codes to receive widespread use had five bits (traditionally called “levels”) to present the alphabet, numerals, symbols and machine functions. In the US, we use International Telegraph Alphabet No. 2 (ITA2), commonly called *Baudot,* as specified in FCC §97.309(a)(1). The code is defined in the ITA2 Codes table in the **References** chapter. In Great Britain, the almost-identical code is called *Murray* code. There are many variations in five-bit coded character sets, principally to accommodate foreign-language alphabets.

Five-bit codes can directly encode only $2^5 = 32$ different symbols. This is insufficient to encode 26 letters, 10 numerals and punctuation. This problem can be solved by using one or more of the codes to select from multiple code-translation tables. ITA2 uses a LTRS code to select a table of upper-case letters and a FIGS code to select a table of numbers, punctuation and special symbols. Certain symbols, such as carriage return, occur in both tables. Unassigned ITA2 FIGS codes may be used for the remote control of receiving printers. This scheme can be expanded, as shown by the ASCII-over-AMTOR discussion latter in this chapter.

FCC rules provide that ITA2 transmissions must be sent using start-stop pulses as illustrated in **Fig 12.22**. The bits in the figure are arranged as they would appear on an oscilloscope.

### *Speeds and Signaling Rates*

The signaling speeds for all forms of RTTY are those used by the old TTYs: 60, 67, 75 or 100 WPM. **Table 12.3** relates speeds, signaling rates and pulse times. In practice, the real speeds do not exactly match their names. The names have been rounded through years of common usage. The Signaling Rates table in the **References** chapter lists names, signaling rates and data patterns for common RTTY speeds.

There’s a problem specifying signaling speed of RTTY because the length of the start and stop pulses vary from that of the data bits. The answer is to base the signaling speed on the shortest pulses used. The *baud* is a unit of signaling speed equal to one pulse (event) per second. The signaling rate, in bauds, is the reciprocal of the shortest pulse length. For example, the “Western Union,” “60 speed” and “45 bauds” speeds all signal at 1/0.022 = 45.45 bauds.

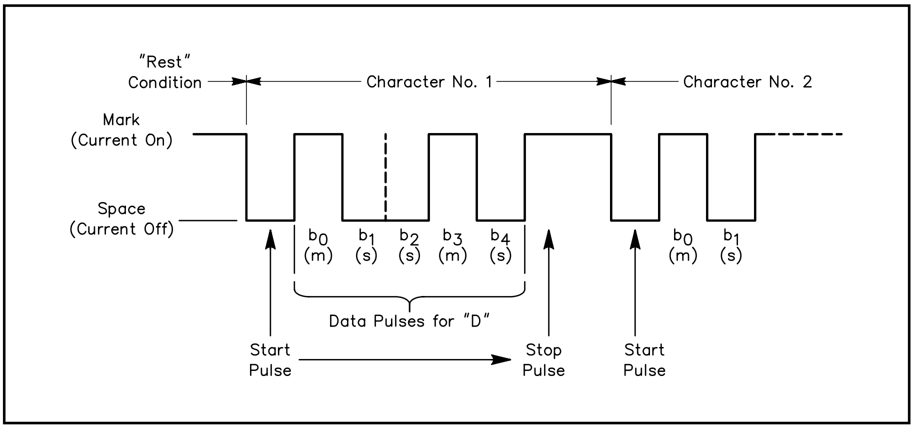



**Fig 12.22—A typical Baudot timing sequence for the letter “D.”**

**Table 12.3**
**Baudot Signaling Rates and Speeds**

| *Signaling Rate (bauds)* | *Data Pulse (ms)* | *Stop Pulse (ms)* | *Speed (WPM)* | *Common Name* |
|---|---|---|---|---|
| 45.45 | 22.0 | 22.0 | 65.00 | Western Union |
| | 22.0 | 31.0 | 61.33 | “60 speed” |
| | 22.0 | 33.0 | 60.61 | 45 bauds |
| 50.00 | 20.0 | 30.0 | 66.67 | European; 50 bauds |
| 56.92 | 17.57 | 25.00 | 76.68 | “75 speed” |
| | 17.57 | 26.36 | 75.89 | 57 bauds |
| 74.20 | 13.47 | 19.18 | 100.00 | “100 bauds” |
| | 13.47 | 20.21 | 98.98 | 74 bauds |
| 100.0 | 10.00 | 15.00 | 133.33 | 100 bauds |

### *Transmitter Keying*

When TTYs and TUs (terminal units) roamed the airwaves, frequency-shift keying (FSK) was the order of the day. DC signals from the TU controlled some form of reactance (usually a capacitor or varactor) in a transmitter oscillator stage that shifted the transmitter frequency. Such direct FSK is still an option with some new radios.

### *AFSK*

Multimode communications processors (MCPs), however, generally connect to the radio AF input and output, often through the speaker and microphone connectors, sometimes through auxiliary connectors. They simply feed AF tones to the microphone input of an SSB transmitter or transceiver. This is called AFSK for "audio frequency-shift keying." When it is properly designed and adjusted, this method of modulation cannot be distinguished from FSK on the air.

When using AFSK, make certain that audio distortion, carrier and unwanted sidebands do not cause interference. Particularly when using the low tones discussed later, the harmonic distortion of the tones should be kept to a few percent. Most modern AFSK generators are of the continuous-phase (CPFSK) type. Older types of noncoherent-FSK (NCFSK) generators had no provisions for phase continuity and produced sharp switching transients. The noise from phase discontinuity caused interference several kilohertz around the RTTY signal.

Also remember that equipment is withstanding a 100% duty cycle for the duration of a transmission. For safe operation, it is often necessary to reduce the transmitter power output (25 to 50% of normal) from that safe for CW operation.

### *What are Low Tones?*

US amateurs customarily use the same modems (2125 Hz mark, 2295 Hz space) for both VHF AFSK and HF via an SSB transmitter. Because of past problems (when 850-Hz shift was used), some amateurs use "low tones" (1275 Hz mark, 1445 Hz space). Both high and low tones can be used interchangeably on the HF bands because only the *amount* of shift is important. The frequency difference is unnoticed on the air because each operator tunes for best results. On VHF AFSK, however, the high and low tone pairs are not compatible.

### *Transmit Frequency*

It is normal to use the lower sideband mode for RTTY on SSB radio equipment. In order to tune to an exact RTTY frequency, remember that most SSB radio equipment displays the frequency of its (suppressed) carrier, not the frequency of the mark signal. Review your MCP's manual to determine the tones used and calculate an appropriate display frequency. For example, to operate on 14,083 kHz with a 2125-Hz AFSK mark frequency, the SSB radio display (suppressed-carrier) frequency should be 14,083 kHz + 2.125 kHz = 14,085.125 kHz.

## Receiving Baudot

Surplus Baudot-encoded teletypewriters (TTY, sometimes called the "green keys") were the mainstay of amateur RTTY from 1946 through around 1977. There are still some mechanical-TTY aficionados, but most operators use computer-based terminals.

Some of the first popular home computers (VIC-20, Commodore 64, Apple II) were adapted to read signals from "terminal units" or "TUs" required by TTYs. TUs translated receiver AF output into 20-mA current-loop signals to drive a polar relay in a TTY. An interface would translate the current-loop signals (or sometimes the receiver AF) to levels appropriate for the computer. Software, unique to each computer, would then decode the stream of marks and spaces into text. This technology was convoluted in that it required many different interfaces and software packages to suit the computers in use. Thankfully, it was soon replaced by multi-mode communications processors.

MCPs accept AF signals from a radio and translate them into common ASCII text or graphics file formats (see **Fig 12.23**). Because the basic interface is via ASCII, MCPs are compatible with virtually any PC running a simple terminal program. There may be compatibility problems with graphics formats, but those are fairly well standardized. Many MCPs handle CW, RTTY, ASCII, AMTOR, packet, fax and SSTV—multimode indeed!

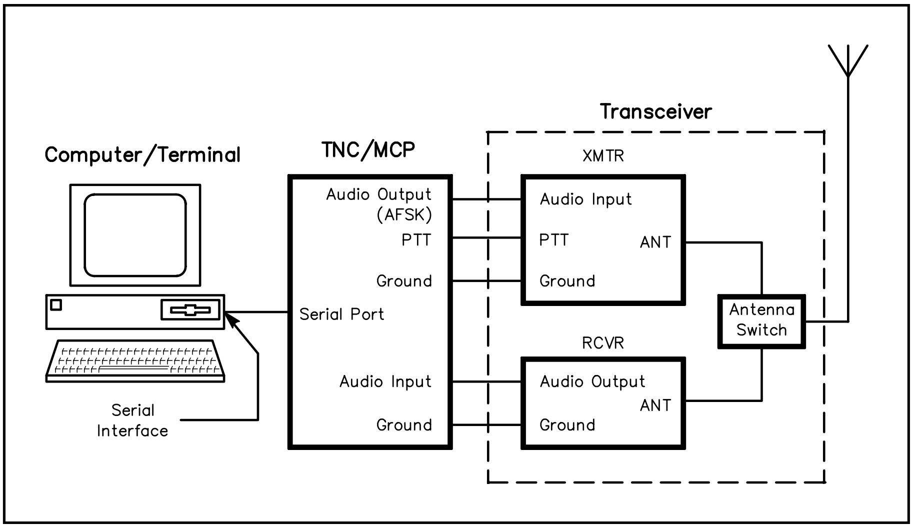



**Fig 12.23—A typical MCP station. MCPs can do all available data modes as well as SSTV and fax.**

### *AFSK Demodulators*

An AFSK demodulator takes the shifting tones from the audio output of a receiver and produces TTY keying pulses. FM is a common AFSK demodulation method. The signal is first band pass filtered to remove out-of-band interference and noise. It is then limited to remove amplitude variations. The signal is demodulated in a discriminator or a PLL. The detector output low pass filtered to remove noise at frequencies above the keying rate. The result is fed to a circuit that determines whether it is a mark or a space.

AM (limiterless) detectors, when properly designed, permit continuous copy even when the mark or space frequency fades out completely. At 170-Hz shift, however, the mark and space frequencies tend to fade at the same time. For this reason, FM and AM demodulators are comparable at 170-Hz shift.

At wider shifts (say 425 Hz and above), the independently fading mark and space can be used to achieve an in-band frequency-diversity effect if the demodulator is capable of processing it. To conserve spectrum, it is generally desirable to stay with 170-Hz shift for 45-baud Baudot and forego the possible in-band frequency-diversity gain. Keep the in-band frequency-diversity gain in mind, however, for higher signaling rates that would justify greater shift.

### *Diversity Reception*

Another type of diversity can be achieved by using two antennas, two receivers and a dual demodulator. This setup is not as far fetched as it may sound; some amateurs are using it with excellent results. One of the antennas would be the normal station antenna for that band. The second antenna could be either another antenna of the same polarization located at least $^3/_8$-wavelength away, or an antenna of the opposite polarization located at the first antenna or anywhere nearby. A problem is to get both receivers on the same frequency without carefully tuning each one. Some RTTY diversity enthusiasts have located slaved receivers on the surplus market. ICOM produced the IC-7072 Transceiver Unit, which slaves an IC-720(A) transceiver to an IC-R70 receiver. Other methods could include a computer controlling two receivers so that both would track.

Two demodulators are needed for this type of diversity. Also, some type of diversity combiner or selector is needed. Many commercial or military RTTY demodulators are equipped for diversity reception.

The payoff for using diversity is a worthwhile improvement in copy. Depending on fading conditions, adding diversity may be equivalent to raising transmitter power sevenfold (8 dB).

### Baudot RTTY Bibliography

Contact information for suppliers named here appears in the **References** chapter Address List.


Ford, *ARRL's HF Digital Handbook* (Newington, CT: ARRL, 1999).

*DATACOM*, British Amateur Radio Teleprinter Group (BARTG).

Henry, "Getting Started in Digital Communications," Part 3, *QST*, May 1992.

Hobbs, Yeomanson and Gee, *Teleprinter Handbook*, Radio Society of Great Britain.

Ingram, *RTTY Today*, Universal Electronics, Inc.

Kretzman, *The New RTTY Handbook*, Cowan Publishing Corp.

Nagle, "Diversity Reception: an Answer to High Frequency Signal Fading," *Ham Radio*, Nov 1979, pp 48-55.

***RTTY Journal.***

Schwartz, "An RTTY Primer," *CQ* magazine, Aug 1977, Nov 1977, Feb 1978, May 1978 and Aug 1978.

Tucker, *RTTY from A to Z*, Cowan Publishing Corp.


### GLOSSARY OF DIGITAL COMMUNICATIONS TERMINOLOGY

**ACK**—Acknowledgment, the control signal sent to indicate the correct receipt of a transmission block.

**Address**—A character or group of characters that identifies a source or destination.

**AFSK**—Audio frequency-shift keying.

**ALOHA**—A channel-access technique wherein each packet-radio station transmits without first checking to see if the channel is free; named after early packet-radio experiments at the University of Hawaii.

**AMICON**—AMSAT International Computer Network—Packet-radio operation on SSC L1 of AMSAT-OSCAR 10 to provide networking of ground stations acting as gateways to terrestrial packet-radio networks.

**AMRAD**—Amateur Radio Research and Development Corporation, a nonprofit organization involved in packet-radio development.

**AMTOR**—Amateur teleprinting over –radio, an amateur radioteletype transmission technique employing error correction as specified in several CCIR documents 476-2 through 476-4 and 625. CCIR Rec. 476-3 is reprinted in the *Proceedings* of the Third ARRL Amateur Radio Computer Networking Conference, available from ARRL Hq.

**ANSI**—American National Standards Institute

**Answer**—The station intended to receive a call. In modem usage, the called station or modem tones associated therewith.

**ARQ**—Automatic repeat request, an error-sending station, after transmitting a data block, awaits a reply (ACK or NAK) to determine whether to repeat the last block or proceed to the next.

**ASCII**—American National Standard Code for Information Interchange, a code consisting of seven information bits.

**AX.25**—Amateur packet-radio link-layer protocol. Copies of protocol specification are available from ARRL Hq.

**Backwave**—An unwanted signal emitted between the pulses of an on/off-keyed signal.

**Balanced**—A relationship in which two stations communicate with one another as equals; that is, neither is a primary (master) or secondary (slave).

**Baud**—A unit of signaling speed equal to the number of discrete conditions or events per second. (If the duration of a pulse is 20 ms, the signaling rate is 50 bauds or the reciprocal of 0.02, abbreviated Bd).

**Baudot code**—A coded character set in which five bits represent one character. Used in the US to refer to ITA2.

**Bell 103**—A 300-baud full-duplex modem using 200-Hz-shift FSK of tones centered at 1170 and 2125 Hz.

**Bell 202**—A 1200-baud modem standard with 1200-Hz mark, 2200-Hz space, used for VHF FM packet radio.

**BER**—Bit error rate.

**BERT**—Bit-error-rate test.

**Bit stuffing**—Insertion and deletion of 0s in a frame to preclude accidental occurrences of flags other than at the beginning and end of frames.

**Bit**—Binary digit, a single symbol, in binary terms either a one or zero.

**BLER**—Block error rate.

**BLERT**—Block-error-rate test.

**Break-in**—The ability to hear between elements or words of a keyed signal.

**Byte**—A group of bits, usually eight.

**Carrier detect (CD)**—Formally, received line signal detector, a physical-level interface signal that indicates that the receiver section of the modem is receiving tones from the distant modem.

**CCIR Rec 476-4**—The CCIR Recommendation used as the basis of AMTOR and incorporated by reference into the FCC Rules.

**CCIR**—International Radio Consultative Committee, an International Telecommunication Union (ITU) agency.

**CCITT**—International Telegraph and Telephone Consultative Committee, an ITU agency. CCIR and CCITT recommendations are available from the UN Bookstore.

**Chirp**—Incidental frequency modulation of a carrier as a result of oscillator instability during keying.

**Collision**—A condition that occurs when two or more transmissions occur at the same time and cause interference to the intended receivers.

**Connection**—A logical communication channel established between peer levels of two packet-radio stations.

**Contention**—A condition on a communications channel that occurs when two or more stations try to transmit at the same time.

**Control field**—An 8-bit pattern in an HDLC frame containing commands or responses, and sequence numbers.

**CRC**—Cyclic redundancy check, a mathematical operation. The result of the CRC is sent with a transmission block. The receiving station uses the received CRC to check transmitted data integrity.

**CSMA**—Carrier sense multiple access, a channel access arbitration scheme in which packet-radio stations listen on a channel for the presence of a carrier before transmitting a frame.

**CTS**—clear to send, a physical-level interface circuit generated by the DCE that, when on, indicates the DCE is ready to receive transmitted data (abbreviated CTS).

**Cut numbers**—In Morse code, shortening of codes sent for numerals.

**DARPA**—Defense Advanced Research Projects Agency; formerly ARPA, sponsors of ARPANET.

**Data set**—Modem.

**Datagram**—A mode of packet networking in which each packet contains complete addressing and control information. (Compare virtual circuit.)

**DCE**—Data circuit-terminating equipment, the equipment (for example, a modem) that provides communication between the DTE and the line radio equipment.

**Destination**—In packet radio, the station that is the intended receiver of the frame sent over a radio link either directly or via a repeater.

**Digipeater**—A link-level gateway station capable of repeating frames. The term "bridge" is used in industry.

**Domain**—In packet radio, the combination of a frequency and a geographical service area.

**DTE**—Data terminal equipment, for example a VDU or teleprinter.

**DXE**—In AX.25, Data switching equipment, a peer (neither master nor slave) station in balanced mode at the link layer.

**EASTNET**—A series of digipeaters along the US East Coast.

**EIA**—Electronic Industries Association.

**EIA-232-C**—An EIA standard physical-level interface between DTE (terminal) and DCE (modem), using 25-pin connectors.

**Envelope-delay distortion**—In a complex waveform, unequal propagation delay for different frequency components.

**Equalization**—Correction for amplitude-frequency and/or phase-frequency distortion.

**Eye pattern**—An oscilloscope display in the shape of one or more eyes for observing the shape of a serial digital stream and any impairments.

**FADCA**—Florida Amateur Digital Communications Association.

**FCS**—Frame check sequence. (See CRC.)

**FEC**—Forward error correction, an error-control technique in which the transmitted data is sufficiently redundant to permit the receiving station to correct some errors.

**Field**—In packet radio, at the link layer, a subdivision of a frame, consisting of one or more octets.

**Flag**—In packet switching, a link-level octet (01111110) used to initiate and terminate a frame.

**Frame**—In packet radio, a transmission block consisting of opening flag, address, control, information, frame-check-sequence and ending flag fields.

**FSK**—Frequency-shift keying.

**Gateway**—In packet radio, an interchange point.

**HDLC**—High-level data link control procedures as specified in ISO 3309.

**Host**—As used in packet radio, a computer with applications programs accessible by remote stations.

**IA5**—International Alphabet No. 5, a 7-bit coded character set, CCITT version of ASCII.

**Information field**—Any sequence of bits containing the intelligence to be conveyed.

**ISI**—Intersymbol interference; slurring of one symbol into the next as a result of multipath propagation.

**ISO**—International Organization for Standardization.

**ITA2**—International Telegraph Alphabet No. 2, a CCITT 5-bit coded character set commonly called the Baudot or Murray code.

**Jitter**—Unwanted variations in amplitude or phase in a digital signal.

**Key clicks**—Unwanted transients beyond the necessary bandwidth of a keyed radio signal.

**LAP**—Link access procedure, CCITT X.25 unbalanced-mode communications.

**LAPB**—Link access procedure, balanced, CCITT X.25 balanced-mode communications.

**Layer**—In communications protocols, one of the strata or levels in a reference model.

**Level 1**—Physical layer of the OSI reference model.

**Level 2**—Link layer of the OSI reference model.

**Level 3**—Network layer of the OSI reference model.

**Level 4**—Transport layer of the OSI reference model.

**Level 5**—Session layer of the OSI reference model.

**Level 6**—Presentation layer of the OSI reference model.

**Level 7**—Application layer of the OSI reference model.

**Loopback**—A test performed by connecting the output of a modulator to the input of a demodulator.

**LSB**—Least-significant bit.

**Mode A**—In AMTOR, an automatic repeat request (ARQ) transmission method.

**Mode B**—In AMTOR, a forward error correction (FEC) transmission method.

**Modem**—Modulator-demodulator, a device that connects between a data terminal and communication line (or radio). Also called data set.

**MSB**—Most-significant bit.

**MSK**—Frequency-shift keying where the shift in Hz is equal to half the signaling rate in bits per second.

**NAK**—Negative acknowledge (opposite of ACK).

**NAPLPS**—ANSI X3.110-1983 Videotex/Teletext Presentation Level protocol syntax.

**NBDP**—Narrow-band direct-printing telegraphy.

**NEPRA**—New England Packet Radio Association.

**Node**—A point within a network, usually where two or more links come together, performing switching, routine and concentrating functions.

**NRZI**—Nonreturn to zero. A binary baseband code in which output transitions result from data 0s but not from 1s. Formal designation is NRZ-S (nonreturn-to-zero—space).

**Null modem**—A device to interconnect two devices both wired as DCEs or DTEs; in EIA RS-232-C interfacing, back-to-back DB25 connectors with pin-for-pin connections except that Received Data (pin 3) on one connector is wired to Transmitted Data (pin 3) on the other.

**Octet**—A group of eight bits.

**OOK**—On-off keying.

**Originate**—The station initiating a call. In modem usage, the calling station or modem tones associated therewith.

**OSI-RM**—Open Systems Interconnection Reference Model specified in ISO 7498 and CCITT Rec X.200.

**Packet radio**—A digital communications technique involving radio transmission of short bursts (frames) of data containing addressing, control and error-checking information in each transmission.

**PACSAT**—AMSAT packet-radio satellite with store-and-forward capability.

**PAD**—Packet assembler/disassembler, a device that assembles and disassembles packets (frames). It is connected between a data terminal (or computer) and a modem in a packet-radio station (see also TNC).

**Parity check**—Addition of noninformation bits to data, making the number of ones in a group of bits always either even or odd.

**PID**—Protocol identifier. Used in AX.25 to specify the network-layer protocol used.

**PPRS**—Pacific Packet Radio Society.

**Primary**—The master station in a master-slave relationship; the master maintains control and is able to perform actions that the slave cannot. (Compare secondary.)

**Protocol**—A formal set of rules and procedures for the exchange of information within a network.

**PSK**—Phase-shift keying.

**RAM**—Random access memory.

**Router**—A network packet switch. In packet radio, a network-level relay station capable of routing packets.

**RS-232-C**—See EIA-232-C.

**RTS**—Request to send, physical-level signal used to control the direction of data transmission of the local DCE.

**RTTY**—Radioteletype.

**RxD**—Received data, physical-level signals generated by the DCE are sent to the DTE on this circuit.

**Secondary**—The slave in a master-slave relationship. Compare primary.

**SOFTNET**—An experimental packet-radio network at the University of Linkoping, Sweden.

**Source**—In packet radio, the station transmitting the frame over a direct radio link or via a repeater.

**SOUTHNET**—A series of digipeaters along the US Southeast Coast.

**SSID**—Secondary station identifier. In AX.25 link-layer protocol, a multipurpose octet to identify several packet-radio stations operating under the same call sign.

**TAPR**—Tucson Amateur Packet Radio Corporation, a nonprofit organization involved in packet-radio development.

**Teleport**—A radio station that acts as a relay between terrestrial radio stations and a communications satellite.

**TNC**—Terminal node controller, a device that assembles and disassembles packets (frames); sometimes called a PAD.

**TR switch**—Transmit-receive switch to allow automatic selection between receive and transmitter for one antenna.

**TTY**—Teletypewriter.

**TU**—Terminal unit, a radioteletype modem or demodulator.

**Turnaround time**—The time required to reverse the direction of a half-duplex circuit, required by propagation, modem reversal and transmit-receive switching time of transceiver.

**TxD**—Transmitted data, physical-level data signals transferred on a circuit from the DTE to the DCE.

**UI**—Unnumbered information frame.

**V.24**—A CCITT standard defining physical-level interface circuits between a DTE (terminal) and DCE (modem), equivalent to EIA RS-232-C.

**V.28**—A CCITT standard defining electrical characteristics for V.24 interface.

**VADCG**—Vancouver Amateur Digital Communications Group.

**VDT**—Video-display terminal.

**VDU**—Video display unit, a device used to display data, usually provided with a keyboard for data entry.

**Videotex**—A presentation-layer protocol for two-way transmission of graphics.

**Virtual circuit**—A mode of packet networking in which a logical connection that emulates a point-to-point circuit is established. (Compare Datagram.)

**WESTNET**—A series of digipeaters along the US West Coast.

**Window**—In packet radio at the link layer, the range of frame numbers within the control field used to set the maximum number of frames that the sender may transmit before it receives an acknowledgment from the receiver.

**X.25**—CCITT packet-switching protocol.

# ASCII

The American National Standard Code for Information Interchange (ASCII) is a coded character set used for information-processing systems, communications systems and related equipment. Current FCC regulations provide that amateur use of ASCII shall conform to ASCII as defined in ANSI Standard X3.4-1977. Its international counterparts are ISO 646-1983 and International Alphabet No. 5 (IA5) as specified in CCITT Rec V.3.

ASCII uses 7 bits to represent letters, figures, symbols and control characters. Unlike ITA2 (Baudot), ASCII has both upper- and lower-case letters. A table of ASCII characters is presented as "ASCII Character Set" in the **References** chapter.

In the international counterpart code, £ replaces #, and the international currency sign ¤ may replace $ by agreement of the sender and recipient. Without such agreement, neither £, ¤ nor $ represent the currency of any particular country.

### *Parity*

While not strictly a part of the ASCII standard, an eighth bit (P) may be added for parity checking. FCC rules permit optional use of the parity bit. The applicable US and international standards (ANSI X3.16-1976; CCITT Rec V.4) recommend an even parity sense for asynchronous and odd parity sense for synchronous data communications. The standards, however, generally are not observed by hams.

### *Code Extensions*

By sacrificing parity, the eighth bit can be used to extend the ASCII 128-character code to 256 characters. Work is underway to produce an international standard that includes characters for all written languages.

## ASCII Serial Transmission

Serial transmission standards for ASCII (ANSI X3.15 and X3.16; CCITT Rec V.4 and X.4) specify that the bit sequence shall be least-significant bit (LSB) first to most-significant bit (MSB), that is b0 through b6 (plus the parity bit, P, if used).

Serial transmission may be either synchronous or asynchronous. In synchronous transmissions, only the information bits (and optional parity bit) are sent, as shown in **Fig 12.24A**.

Asynchronous serial transmission adds a start pulse and a stop pulse to each character. The start pulse length equals that of an information pulse. The stop pulse may be one or two bits long. There is some variation, but one stop bit is the convention, except for 110-baud transmissions with mechanical teletypewriters.

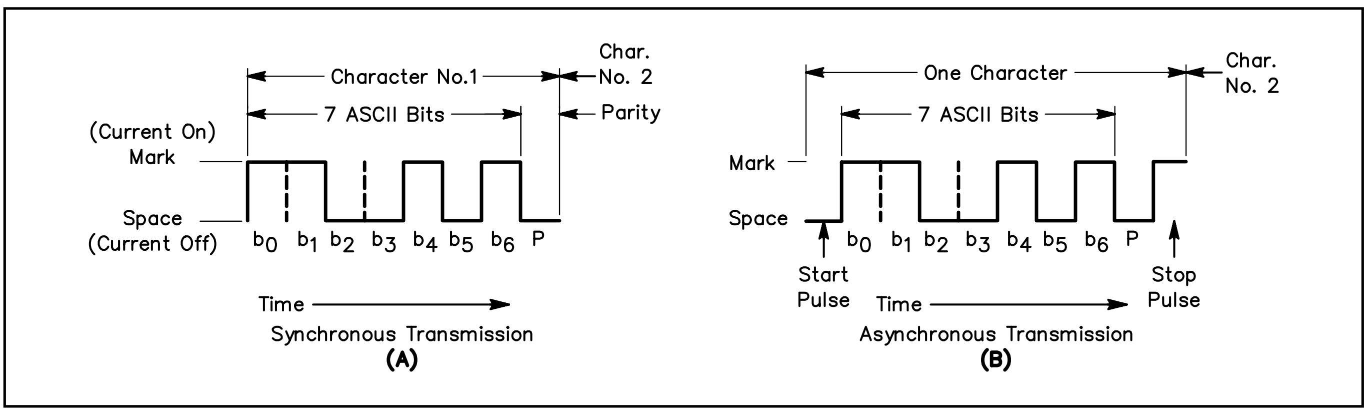



**Fig 12.24—Typical serial synchronous and asynchronous timing for the ASCII character S.**

## ASCII-over-AMTOR

The superior weak signal performance of AMTOR, compared to RTTY and HF packet radio, has made it a popular mode for HF data networks. AMTOR BBS systems are popular for passing long-haul traffic. Traffic from VHF and UHF packet networks is converted into AMTOR (and more recently PACTOR and CLOVER) by specially equipped HF BBS stations. This system combines the best attributes of several different data transfer modes: The convenience and short-range high data rate of VHF/UHF packet is combined with the high reliability of AMTOR/PACTOR/CLOVER for long-range HF data transfer.

There was a problem with AMTOR relays, however. The CCIR-476 and CCIR-625 AMTOR symbol set has no lower-case letters and lacks many punctuation symbols common in VHF/UHF packet radio. Therefore, messages routed via AMTOR can differ from the original in format and appearance. Differences in the header text of AMTOR vs packet messages can be particularly troublesome to automated data-transfer systems.

In late 1991, G3PLX (Peter Martinez) and W5SMM (Vic Poor) devised an extended AMTOR character set that contains all of the printable ASCII symbols (ASCII control characters are not supported). Using this scheme, AMTOR-delivered messages are indistinguishable from those delivered via ASCII-based modes (such as packet radio). This “ASCII-over-AMTOR” system uses the generally unused “blank” character code (“00000” in Baudot, “1101010” in AMTOR) to toggle between the standard AMTOR character set and the new “Blank Code Extension” character set, which includes lower-case letters and ASCII punctuation symbols.

When two ASCII-over-AMTOR equipped stations first link, both controllers are set to the standard CCIR-476/625 character set; upper-case letters are sent and the FIGS code (AMTOR “0110110”) switches between letters and numbers. When the first “blank” character is sent, both stations switch to the new character set. Any following AMTOR letter codes are assumed to be lower-case letters and FIGS codes are translated into the new punctuation symbol set. A second instance of the “blank” code switches both stations back to the standard AMTOR character set. The expanded ASCII-over-AMTOR character set is shown in the Table.

### ASCII-over-AMTOR Blank-Code Extension Symbol Set

| | ***Standard CCIR-476*** | | ***Blank Code Extension*** | |
|---|---|---|---|---|
| *Bit Code* | *Ltrs* | *Figs* | *Ltrs* | *Figs* |
| 1000111 | A | - | a | _ |
| 1110010 | B | ? | b | |
| 0011101 | C | : | c | ; |
| 1010011 | D | WRU | d | WRU |
| 1010110 | E | 3 | e | |
| 0011011 | F | % | f | ‘ |
| 0110101 | G | @ | g | } |
| 1101001 | H | # | h | { |
| 1001101 | I | 8 | i | |
| 0010111 | J | * | j | |
| 0011110 | K | ( | k | [ |
| 1100101 | L | ) | l | ] |
| 0111001 | M | . | m | > |
| 1011001 | N | , | n | < |
| 1110001 | O | 9 | o | ~ |
| 0101101 | P | 0 | p | |
| 0101110 | Q | 1 | q | ! |
| 1010101 | R | 4 | r | $ |
| 1001011 | S | ‘ | s | “ |
| 1110100 | T | 5 | t | |
| 1001110 | U | 7 | u | & |
| 0111100 | V | = | v | \| |
| 0100111 | W | 2 | w | |
| 0111010 | X | / | x | \ |
| 0101011 | Y | 6 | y | ^ |
| 1100011 | Z | + | z | |
| 1111000 | CR | CR | CR | CR |
| 1101100 | LF | LF | LF | LF |
| 1011010 | LTRS | LTRS | LTRS | LTRS |
| 0110110 | FIGS | FIGS | FIGS | FIGS |
| 1011100 | SP | SP | SP | SP |
| 1101010 | BLNK | BLNK | BLNK | BLNK |

CR = carriage return
LF = line feed
LTRS = shift to letter characters
FIGS = shift to figure characters
SP = space
BLNK = toggle between CCIR-476 and Blank Code Extension sets.

*Notes:*

1. The logic state “1” represents the Mark or “Z” condition, the higher radiated radio frequency.
2. Certain FIGS-case symbols follow CCIR-476 and common European usage, differing from the “US TTY” symbols shown in the ITA2 Codes table in the **References** chapter. These differences are necessary to assure international compatibility.
3. The signal “BELL” is not supported because it is generally a nuisance to operation of otherwise silent automated message relay stations. If BELL is required, use FIGS-J in the Blank Extension set.

The ASCII-over-AMTOR extended symbol set is supported by most commercially available AMTOR controllers and popular BBS software, such as APLINK and AMTOR MBO. The symbol set is backward compatible with stations that do not have the extended capability. A station that is not equipped with ASCII-over-AMTOR will notice very few differences when receiving these signals except that all letters will appear to be upper-case and the standard punctuation symbols will be printed.

The ASCII-over-AMTOR extension is remarkably efficient. If no nonBaudot characters are sent, there is no additional overhead to the transmission. Even if the extended set is sent, far fewer bits are transmitted than if ASCII were transmitted.

This technique, however, requires an error-correcting code such as AMTOR. The concept would not work with standard Baudot RTTY because a noise "hit" on a Blank character would result in printing from the wrong symbol set. The AMTOR error-correcting code is not infallible, but on-the-air use of ASCII-over-AMTOR has demonstrated that case-errors are very rare. The system works well and is in daily use by AMTOR BBS stations throughout the world.

## ASCII Data Rates

Data-communication signaling rates depend largely on the medium and the state of the art when the equipment was selected. Numerous national and international standards that recommend different data rates, are listed in **Table 12.4**. The most-used rates tend to progress in 2:1 steps from 300 to 9600 bits/s and in 8 kbits/s increments from 16 kbits/s upward (see **Table 12.5**). For Amateur Radio, serial ASCII transmissions data rates of 75, 110, 150, 300, 600, 1200, 2400, 4800, 9600, 16000, 19200 and 56000 bits/s are suggested.

**Table 12.4**
**Data Transmission Signaling-Rate Standards**

| *Standard* | *Signaling Rates (bit/s)* | *Tolerance* |
|---|---|---|
| ***CCIT*** | | |
| V.5 | 600, 1200, 2400, 4800 | ±0.01% |
| V.6 | Preferred: 600, 1200, 2400, 3600, 4800, 7200, 9600<br>Supplementary: 1800, 3000, 4200, 5400, 6000, 6600 7800, 8400, 9000, 10200, 10800 | ±0.01% |
| V.21 | 110, 150,<br>300 (where possible) | ≤200 bit/s<br>≤300 bit/s |
| V.23 | 600<br>1200<br>75 (backward channel) | ≤600 bit/s<br>≤1200 bit/s<br>≤75 bits |
| V.34 | 28800, 26400, 24000, 21600, 19200, 16800 or 14400 | |
| V.35 | Preferred: 48000 | |
| V.36 | Recommended for international use: 48000<br>Certain applications: 56000, 64000, 72000 | |
| X.3 | Packet assembly/disassembly speeds: 50, 75, 100, 134.5, 150, 200, 300, 600, 1200, 1200/75, 1800, 2400, 4800, 9600, 19200, 48000, 56000, 64000 | |
| ***ANSI*** | | |
| X3.1 | Serial: 75, 150, 300, 600, 1200, 2400, 4800, 7200, 9600<br>Parallel: 75, 150, 300, 600, 900, 1200 | |
| X3.36 | Above 9600 bit/s, signaling rates shall be in integral multiples of 8000 bit/s.<br>Selected standard rates: 16000, 56000, 1344000 and 1544000<br>Recognized for international use: 48000 | |
| ***EIA*** | | |
| RS-269-B | (Same as ANSI X3.1) | |
| ***FED STD*** | | |
| -1001 | (Same as ANSI X3.36) For foreign communications: 64000 | |
| -1041 | 2400, 4800, 9600 | |

### *Bauds vs Bits Per Second*

The "baud" is a unit of signaling speed equal to one discrete condition or event per second. In single-channel transmission, such as the FCC prescribes for Baudot transmissions, the signaling rate in bauds equals the data rate in bits per second. However, the FCC does not limit ASCII to single-channel transmission. Some digital modulation systems have more than two (mark and space) states. In *dibit* (pronounced die-bit) modulation, two ASCII bits are sampled

**Table 12.5**
**ASCII Asynchronous Signaling Rates**

| *Bits per Second* | *Data Pulse (ms)* | *Stop Pulse (ms)* | *CPS* | *WPM* |
|---|---|---|---|---|
| 110 | 9.091 | 18.182 | 10.0 | 100 |
| 150 | 6.667 | 6.667 | 15.0 | 150 |
| 300 | 3.333 | 3.333 | 30.0 | 300 |
| 600 | 1.667 | 1.667 | 60.0 | 600 |
| 1200 | 0.8333 | 0.8333 | 120 | 1200 |
| 2400 | 0.4167 | 0.4167 | 240 | 2400 |
| 4800 | 0.2083 | 0.2083 | 480 | 4800 |
| 9600 | 0.1041 | 0.1041 | 960 | 9600 |
| 19200 | 0.0520 | 0.0520 | 1920 | 19200 |

$$\text{CPS} = \text{characters per second} = \frac{1}{\text{START} + 7(\text{DATA}) + \text{PARITY} + \text{STOP}}$$

$$\text{WPM} = \text{words per minute} = \frac{\text{CPS}}{6} \times 60 = \text{number of 5-letter-plus-space groups per minute}$$

at a time. The four possible states for a dibit are 00, 01, 10 and 11. In four-phase modulation, each state is assigned an individual phase of 0º, 90º, 180º and 270º respectively. For dibit phase modulation, the signaling speed in bauds is half the information-transfer rate in bits/s. As the FCC specifies the digital sending speed in bauds, amateurs may transmit ASCII at higher information rates by using digital modulation systems that encode more bits per signaling element. This technology is open for exploration by Amateur Radio experimenter. One such example is Clover II.

### Amateur ASCII RTTY Operations

On April 17, 1980 the FCC first permitted ASCII in the Amateur Radio Service. US amateurs have been slow to abandon Baudot in favor of asynchronous serial ASCII.

One cause for resistance is the reasoning that asynchronous ASCII has two (or three with a parity bit added) more bits than asynchronous Baudot and is usually sent at higher speeds. Thus, it is felt that the greater data rates and increased bandwidth needed for ASCII would make its reliability less than that of Baudot. This is true as far as it goes, but does not exhaust the theoretical possibilities, which will be discussed below.

On the practical side, some amateurs tried ASCII on the air and experienced poor results. In some cases, this can be traced to the use of modems that were optimized for 45-baud operation. At 110 or 300 bauds, the 45-baud mark and space filters are too narrow.

On the HF bands, speeds above 50 or 75 bauds are subject to intersymbol interference (ISI, slurring one pulse into the next) from multipath propagation. Multiple paths can be avoided by operating at the maximum usable frequency (MUF), where there is only one ray path. The amount of multipath delay varies according to operating frequency with respect to the MUF and path distance. Paths in the 600- to 5000-mile range are generally less subject to multipath than shorter or longer ones. Paths of 250 miles or less are difficult from a multipath standpoint. As a result, successful operation at the higher ASCII speeds depends on using the highest frequency possible as well as having suitable modems at both ends of the circuit.

Returning to the theoretical comparison of Baudot and ASCII, recall that the FCC requires asynchronous (start-stop) transmission of Baudot. This means that the five information pulses must be sent with a start pulse and a stop pulse, usually of 1.42 times the length of the information pulse. Thus, an asynchronous Baudot transmitted character requires 7.42 units. In contrast, 7 bits of ASCII plus a parity bit, a start and a two-unit stop pulse has 11 units.

However, it is possible to send only the 7 ASCII information bits synchronously (without start and stop pulses), making the number of units that must be transmitted (7 vs 7.42) slightly smaller for ASCII than for Baudot. Or, it is possible to synchronously transmit 8 bits (7 ASCII bits plus a parity bit) and take advantage of the error-detection capability of parity. Also, there is nothing to prevent ASCII from being sent at a lower speed such as 50 or 75 bauds, to make it as immune to multipath as is 45- or 50-baud Baudot RTTY. So it is easy to see that ASCII can be as reliable as Baudot RTTY, if care is used in system design.

While 45- or 50-baud RTTY circuits can provide reliable communications, this range of signaling speeds does not make full use of the HF medium. Speeds ranging from 75 to 1200 bauds can be achieved on HF with error-detection and error-correction techniques similar to those used in AMTOR. Reliable transmission at higher speeds can be accomplished by means of more sophisticated modes, which are described later in this chapter.

### ASCII Bibliography


ANSI X3.4-1977, "Code for Information Interchange," American National Standards Institute.

ANSI X3.15-1976, "Bit sequencing of the American National Standard Code for Information Interchange in Serial-by-Bit Data Transmission."

ANSI X3.16-1976, "Character Structure and Character Parity Sense for Serial-by-Bit Data Communication Information Interchange."

ANSI X3.25-1976, "Character Structure and Character Parity Sense for Parallel-by-Bit Communication in American National Standard Code for Information Interchange."

Bemer, "Inside ASCII," *Interface Age*, May, June and July 1978.

CCITT V.3, "International Alphabet No. 5," International Telegraph and Telephone Consultative Committee, CCITT volumes with recommendations prefixed with the letters V and X are available from United Nations Bookstore.

CCITT V.4, "General Structure of Signals of International Alphabet No. 5 Code for Data Transmission over the Public Telephone Network."

ISO 646-1973 (E), "7-bit Coded Character Set for Information Processing Interchange," International Organization for Standardization, available from ANSI.

Mackenzie, *Coded Character Sets, History and Development*, Addison-Wesley Publishing Co, 1980.

## AMTOR

RTTY circuits are plagued with problems of fading and noise unless something is done to mitigate these effects. Frequency, polarization and space diversity are methods of providing two or more simultaneous versions of the transmission to compare at the receiving station. Another method of getting more than one opportunity to see a given transmission is time diversity. The same signal sent at different times will experience different fading and noise conditions. Time diversity is the basis of AMTOR or Amateur Teleprinting Over Radio.

AMTOR always uses two forms of time diversity in either Mode A (ARQ, automatic repeat request) or Mode B (FEC, forward error correction). In Mode A, a repeat is sent only when requested by the receiving station. In Mode B, each character is sent twice. In both Mode A or Mode B, the second type of time diversity is supplied by the redundancy of the code itself.

Since 1983, AMTOR has been part of the US Amateur Radio rules. The rules recognize several documents that define AMTOR, from 476-2 (1978) to CCIR Rec 476-4 and Rec 625 (1986). Anyone interested in the design aspects of AMTOR should refer to these recommendations. You may obtain a complete reprint of Rec 476-3 as part of the *Proceedings* of the Third ARRL Amateur Radio Computer Networking Conference, available from ARRL Hq.

### Overview

AMTOR is based on SITOR, a system devised in the Maritime Mobile Service as a means of improving communications between RTTYs using the ITA2 (Baudot) code. The system converts the 5-bit code to a 7-bit code for transmission such that there are 4 mark and 3 space bits in every character.

The constant mark/space ratio limits the number of usable combinations to 35. ITA2 takes up 32 of the combinations; the 3 remaining are service information signals—α, β and RQ in **Table 12.6**. The table also shows several other service signals that are borrowed from the 32 combinations that equate to ITA2. They are not confused with the message characters because they are sent only by the receiving station.

#### *Mode B (FEC)*

When transmitting to no particular station (for example calling CQ, net operation or bulletin transmissions) there is no (one) receiving station to request repeats. Even if one station were selected, its ability to receive properly may not be representative of others desiring to copy the signal.

Mode B uses a simple forward-error-control (FEC) technique: it sends each character twice. Burst errors are virtually eliminated by delaying the repetition for a period thought to exceed the duration of most noise bursts. In AMTOR, groups of five characters are sent (DX) and then repeated (RX). At 70 ms per character, there is 280 ms between the first and second transmissions of a character.

**Table 12.6**
**CCIR Rec 625 Service Information Signals[1]**

| *Mode A (ARQ)* | *Bit No. 6543210* | *Mode B (FEC)* |
|---|---|---|
| Control signal 1 (CS1) | 1100101 | |
| Control signal 2 (CS1) | 1101010 | |
| Control signal 3 (CS3) | 1011001 | |
| Control signal 4 (CS4) | 0110101 | |
| Control signal 5 (CS5) | 1101001 | |
| Idle signal β | 0110011 | Idle signal β |
| Idle signal α | 0001111 | Phasing signal 1, idle signal α |
| Signal repetition (RQ) | 1100110 | Phasing signal 2 |

[1] 1 represents the mark condition (shown as B in CCIR recommendations), which is the higher emitted radio frequency for FSK, the lower audio frequency for AFSK. 0 represents the space condition (shown as Y in CCIR recommendations). Bits are numbered 0 (LSB) through 6 (MSB). The order of bit transmission is LSB first, MSB last.

The receiving station tests for the constant 4/3 mark/space ratio and prints only unmutilated DX or RX characters. If both are mutilated, an error symbol or space prints.

The Information Sending Station (ISS) transmitter must be capable of 100% duty-cycle operation for Mode B. Thus, it may be necessary to reduce power level to 25% to 50% of full rating.

The Sounds of Amateur Radio **Listen to calling CQ on AMTOR.**

### *Mode A (ARQ)*

This synchronous system, transmits blocks of three characters from the Information Sending Station (ISS) to the Information Receiving Station (IRS). After each block, the IRS either acknowledges correct receipt (based on the 4/3 mark/space ratio), or requests a repeat. This cycle repeats as shown in **Fig 12.25**.

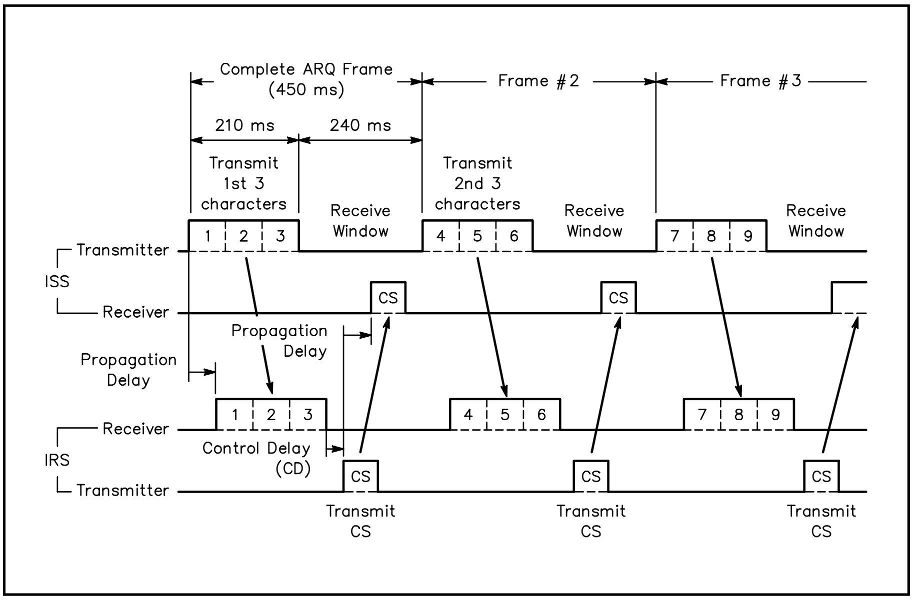



**Fig 12.25—Typical AMTOR timing. Dark arrows indicate the signal path from the ISS to the IRS and vice versa. Note the propagation delays; they determine the minimum and maximum communcations distances.**

The station that initiates the ARQ protocol is known as the Master Station (MS). The MS first sends the selective call of the called station in blocks of three characters, listening between blocks. Four-letter AMTOR calls are normally derived from the first character and the last three letters of the station call sign. For example, W1AW's AMTOR call would be WWAW. The Slave Station (SS) recognizes its selective call and answers that it is ready. The MS now becomes the ISS and will send traffic as soon as the IRS says it is ready.

When an ISS is done sending, it can enable the other station to become the ISS by sending the three-character sequence FIGS Z B. A station ends the contact by sending an "end of communication signal,'' three Idle Signal Alphas.

On the air, AMTOR Mode A signals have a characteristic "chirp-chirp" sound. Because of the 210/240-ms on/off timing, Mode A can be used with some transmitters at full power levels.

The W1AW AMTOR Mode B transmission follows the Baudot and ASCII bulletins. A W1AW schedule appears in the **References** chapter.

## AMTOR Bibliography


S. Ford, *ARRL's HF Digital Handbook* (Newington, CT: ARRL, 1999).

CCIR, Recommendation 476-3, "Direct-Printing Telegraph Equipment in the Maritime Mobile Service." Reprint available from ARRL Hq as part of the *Proceedings* of the Third ARRL Amateur Radio Computer Networking Conference.

*DATACOM*, British Amateur Radio Teleprinter Group (BARTG).

Henry, "Getting Started in Digital Communications," Part 4, *QST*, June 1992.

Martinez, "AMTOR, An Improved RTTY System Using a Microprocessor," *Radio Communication*, RSGB, Aug 1979.
Martinez, "AMTOR, The Easy Way," *Radio Communication*, RSGB, June/July 1980.
Martinez, "AMTOR—a Progress Report," *Radio Communication*, RSGB, Sep 1981, p 813.
Meyn, "Operating with AMTOR," Technical Correspondence, *QST*, July 1983, pp 40-41.
Newland, "An Introduction to AMTOR," *QST*, July 1983.
Newland, "A User's Guide to AMTOR Operation," *QST*, Oct 1985.

# PACKET RADIO

*Data communications* is telecommunications between computers. *Packet switching* is a form of data communications that transfers data by subdividing it into "packets," and *packet radio* is packet switching using the medium of radio. This description was written by Steve Ford, WB8IMY.

Packet radio has its roots in the Hawaiian Islands, where the University of Hawaii began using the mode in 1970 to transfer data to its remote sites dispersed throughout the islands. Amateur packet radio began in Canada after the Canadian Department of Communications permitted amateurs to use the mode in 1978. (The FCC permitted amateur packet radio in the US in 1980.)

In the first half of the 1980s, packet radio was the habitat of experimenters and those few communicators who did not mind communicating with a limited number of potential fellow packet communicators. In the second half of the decade, packet radio "took off" as the experimenters built a network that increased the potential number of packet stations that could intercommunicate and thus attracted tens of thousands of communicators who wanted to take advantage of this potential. Today, packet radio is one of the most popular modes of Amateur Radio communications, because it is very effective.

It provides error-free data transfer. The receiving station receives information exactly as the transmitting station sends it, so you do not waste time deciphering communication errors caused by interference or changes in propagation.

It uses time efficiently, since packet bulletin-board systems (PBBSs) permit packet operators to store information for later retrieval by other amateurs.

It uses the radio spectrum efficiently, since one radio channel may be used for multiple communications simultaneously or one radio channel may be used to interconnect a number of packet stations to form a "cluster" that provides for the distribution of information to all of the clustered stations. The popular *DX PacketClusters* are typical examples (see **Fig 12.26**).

Each local channel may be connected to other local channels to form a network that affords interstate and international data communications. This network can be used by interlinked packet bulletin-board systems to transfer information, messages and third-party traffic via HF, VHF, UHF and satellite links.

It uses other stations efficiently, since any packet-radio station can use one or more other packet-radio stations to relay data to its intended destination.

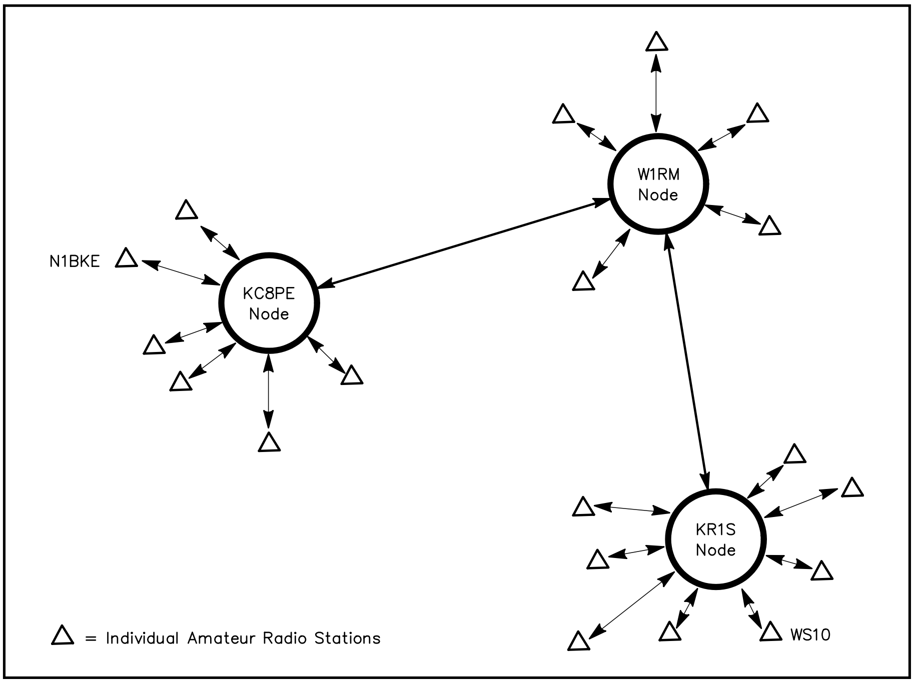



**Fig 12.26—*DX PacketClusters* are networks comprised of individual nodes and stations with an interest in DXing and contesting. In this example, N1BKE is connected to the KC8PE node. If he finds a DX station on the air, he'll post a notice—otherwise known as a *spot*—which the KC8PE node distributes to all its local stations. In addition, KC8PE passes the information along to the W1RM node. W1RM distributes the information and then passes it to the KR1S node, which does the same. Eventually, WS1O—who is connected to the KR1S node—sees the spot on his screen. Depending on the size of the network, WS1O will receive the information within minutes after it was posted by N1BKE.**

It uses current station transmitting and receiving equipment efficiently, since the same equipment used for voice communications may be used for packet communications. The outlay for the additional equipment necessary to make your voice station a packet-radio station may be as little as $100. It also allows you to use that same equipment as an alternative to costly landline data communications links for transferring data between computers.

## The TNC

The terminal node controller—or *TNC*—is at the heart of every packet station. A TNC is actually a computer unto itself. It contains the AX.25 packet protocol firmware along with other enhancements depending on the manufacturer. The TNC communicates with you through your computer or data terminal. It also allows you to communicate with other hams by feeding packet data to your transceiver.

The TNCs accepts data from a computer or data terminal and assembles it into packets (see **Fig 12.27**). In addition, it translates the digital packet data into audio tones that can be fed to a transceiver. The TNC also functions as a receiving device, translating the audio tones into digital data a computer or terminal can understand. The part of the TNC that performs this tone-translating function is known as a *modem* (see **Fig 12.28**).

If you're saying to yourself, "These TNCs sound a lot like telephone modems," you're pretty close to the truth! The first TNCs were based on telephone modem designs. If you're familiar with so-called *smart* modems, you'd find that TNCs are very similar.

You have plenty of TNCs to choose from. The amount of money you'll spend depends directly on what you want to accomplish. Most TNCs are designed to operate at 300 and 1200 bit/s, or 1200 bit/s exclusively (see **Fig 12.29**). There are also TNCs dedicated to 1200 and 9600 bit/s operation, or 9600 bit/s exclusively. Many of these TNCs include convenient features such as personal

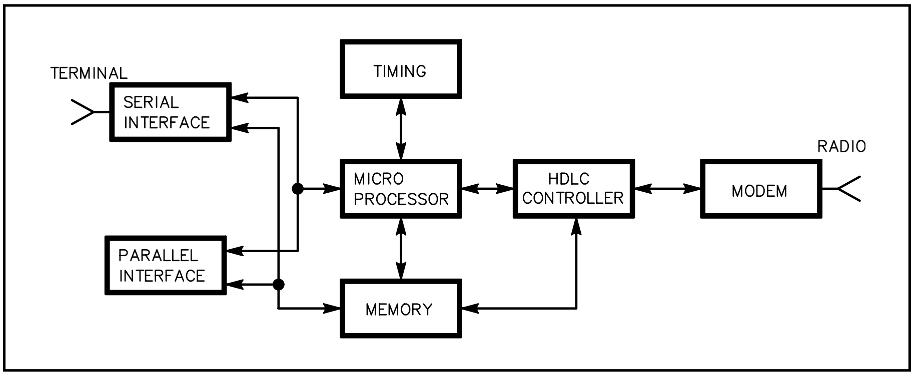



**Fig 12.27—The functional block diagram of a typical TNC.**

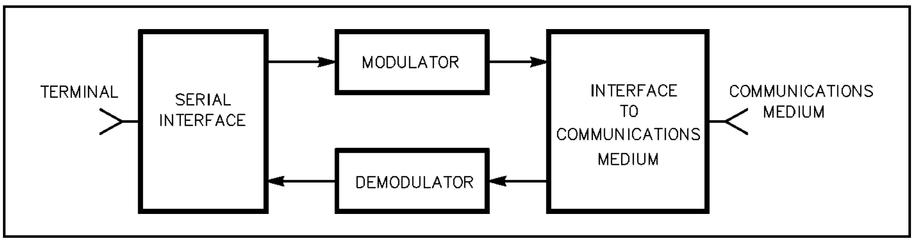



**Fig 12.28—A block diagram of a typical modem.**


**Fig 12.29—Four popular 1200 bit/s packet TNCs: (clockwise, from bottom left) the MFJ-1270C, AEA PK-88, Kantronics KPC-3 and the DRSI DPK-2.**

packet mailboxes where friends can leave messages when you're not at home. Some TNCs also include the ability to easily disconnect the existing modem and substitute another. This feature is very important if you wish to experiment at different data rates. For example, a 1200 bit/s TNC with a *modem disconnect header* can be converted to a 9600 bit/s TNC by disconnecting the 1200 bit/s modem and adding a 9600 bit/s modem.

If you're willing to spend more money, you can buy a complete *multimode communications processor*, or *MCP*. These devices not only offer packet, they also provide the capability to operate RTTY, CW, AMTOR, PACTOR, FAX and other modes. In other words, an MCP gives you just about every digital mode in one box.

### *TNC Emulation and Internal TNCs*

TNC-emulation systems exist for IBM PCs and compatibles. One is known as *BayCom,* which uses the PC to emulate the functions of a TNC/terminal while a small external modem handles the interfacing. BayCom packages are available in kit form for roughly half the price of a basic TNC.

PC owners also have the option of buying full-featured TNCs that mount *inside* their computers. TNC *cards* are available on the market. They are complete TNCs that plug into card slots inside the computer cabinet. No TNC-to-computer cables are necessary. Connectors are provided for cables that attach to your transceiver. In many cases, specialized software is also provided for efficient operation.

## Transceiver Requirements

Packet activity on the HF bands typically takes place at 300 bit/s using common SSB transceivers. The transmit audio is fed from the TNC to the microphone jack or auxiliary audio input. Receive audio is obtained from the external speaker jack or auxiliary audio output. Tuning is critical for proper reception; a visual tuning indicator—available on some TNCs and all MCPs—is recommended.

These simple connections also work for 1200 bit/s packet, which is common on the VHF bands (2 m in particular). Almost any FM transceiver can be made to work with 1200 bit/s packet by connecting the transmit audio to the microphone jack and taking the receive audio from the external speaker (or earphone) jack.

At data rates beyond 1200 bit/s, transceiver requirements become more rigid. At 9600 bit/s (the most popular data rate above 1200 bit/s), the transmit audio must be injected at the modulator stage of the FM transceiver. Receive audio must be tapped at the discriminator. Most 9600 bit/s operators use modified Amateur Radio transceivers or commercial radios. The Motorola *Mitrek* transceiver is a popular choice.

In the mid '90s amateur transceiver manufacturers began incorporating data ports on some FM voice rigs. The new "data-ready" radios are not without problems, however. Their IF filter and discriminator characteristics leave little room for error. If you're off frequency by a small amount, you may not be able to pass data. In addition, the ceramic discriminator coils used in some transceivers have poor group delay, making it impossible to tune them for wider bandwidths. With this in mind, some amateurs prefer to make the leap to 9600 bit/s and beyond using *dedicated* amateur data radios such as those manufactured by Tekk and Kantronics (see Address List in **References** chapter), among others.

Regardless of the transceiver used, setting the proper deviation level is extremely critical. At 9600 bit/s, for example, optimum performance occurs when the maximum deviation is maintained at 3 kHz. Deviation adjustments involve monitoring the transmitted signal with a deviation meter or service monitor. The output level of the TNC is adjusted until the proper deviation is achieved.

The Sounds of Amateur Radio

**Listen to a 9600 baud packet transmission.**

## Packet Networking

### *Digipeaters*

A digipeater is a packet-radio station capable of recognizing and selectively repeating packet frames. An equivalent term used in industry is *bridge*. Virtually any TNC can be used as a single-port digipeater, because the digipeater function is included in the AX.25 Level 2 protocol firmware. Although the use of digipeaters is waning today as network nodes take their place, the digipeater function is handy when you need a relay and no node is available, or for on-the-air testing.

### *NET/ROM*

Ron Raikes, WA8DED, and Mike Busch, W6IXU, developed new firmware for the TNC 2 (and TNC-2 clones) that supports Levels 3 and 4, the Network and Transport layers of the packet-radio network. NET/ROM replaces the TNC-2 EPROM (that contains the TAPR TNC-2 firmware) and converts the TNC into a *network node controller (NNC)* for use at wide- and medium-coverage digipeater sites. Since it is so easy to convert an off-the-shelf TNC into an NNC via the NET/ROM route, NET/ROM has become the most popular network implementation in the packet-radio world and has been installed at most dedicated digipeater stations, thus propelling the standard AX.25 digipeater into packet-radio history.

The NET/ROM network user no longer has to be concerned with the digipeater path required to get from one point to another. All you need to know is the local node of the station you wish to contact. NET/ROM knows what path is required, and if one path is not working or breaks down for some reason, NET/ROM will switch to an alternative path, if one exists. You can be assured that NET/ROM is on top of things, because each NET/ROM node automatically updates its node list periodically, and whenever a new node comes on the air, the other NET/ROM nodes become aware of the new node's existence. In addition to automatic route updating, routing information may also be updated manually by means of a terminal keyboard or remotely using a packet-radio connection.

Once you are connected to another station via the NET/ROM network, most of your packets get through because node-to-node packet acknowledgment is used rather than end-to-end acknowledgment. Besides offering node-to-node acknowledgment, NET/ROM also allows you to build cross-frequency or cross-band multiport nodes. This is done by installing NET/ROM in two TNCs and connecting their serial ports together. In addition to providing these sophisticated NNC functions, NET/ROM also provides the standard AX.25 digipeater function.

### *ROSE*

Several years ago, the Radio Amateur Telecommunications Society (RATS) developed a networking protocol known as *R*ATS *O*pen *S*ystem *E*nvironment, or *ROSE*. Like networks based on *NET/ROM* nodes, the objective of ROSE is to let the network do the work when you're trying to connect to another station.

Using a ROSE network is similar to using the telephone. ROSE nodes are frequently referred to as *switches*, and each switch has its own address based on the telephone area code and the first 3 digits of the local exchange. A ROSE switch in one area of Connecticut, for example, may have an address of 203555. 203 is the area code and 555 is the local telephone exchange. The ROSE network uses this addressing system to create reliable routes for packets (see **Fig 12.30**).

Unless you wish to set up a ROSE switch of your own, you won't need special equipment or software to use the network. You can access a ROSE network today if a switch is available in your area. All you need to know is the call sign of your local switch and the ROSE address of the switch nearest to any stations to want to contact.

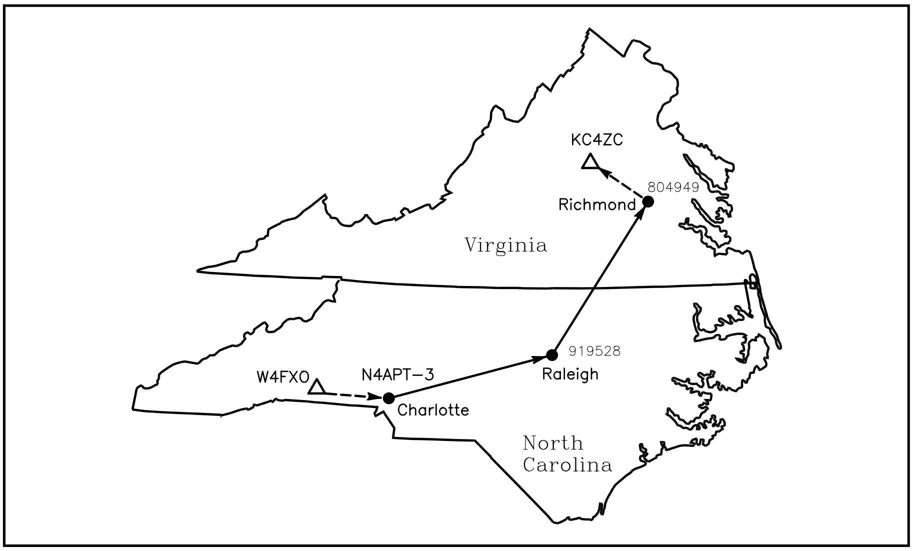



**Fig 12.30—In this hypothetical example, W4FXO, near Charlotte, North Carolina, uses the ROSE network to establish a connection to KC4ZC northwest of Richmond, Virginia. All that W4FXO has to do is issue a connect request that includes his local ROSE switch (N4APT-3) and the ROSE address of the switch nearest KC4ZC (804949). When the request is sent, the network takes over. In this example, the connection to KC4ZC is established by using a ROSE switch in Raleigh.**

ROSE networks are appearing in many areas of the country. They are especially popular in the southeast and midAtlantic states. ROSE addresses and system maps are available from RATS (see **References** chapter Address List). Send a business-sized SASE with your request.

### *TexNet*

TexNet is a high speed, centralized packet networking system developed by the Texas Packet Radio Society (TPRS). Designed for local and regional use, TexNet provides AX.25-compatible access on the 2-m band at 1200 bit/s. This allows packeteers to use TexNet without investing in additional equipment or software. The node-to-node backbones operate in the 70-cm band with data moving through the network at 9600 bit/s. Telephone links are also used to bridge some gaps in the system.

The network offers a number of services to its users. Two conference levels are available by simply connecting to the proper node according to its SSID. By connecting to W5YR-2, for example, you'll join the first conference level. Connecting to W5YR-3 places you in the second level. When you connect to a conference, you can chat with anyone else on the network in roundtable fashion.

Every TexNet network is served by a single PBBS. By using only one PBBS, the network isn't bogged down with constant mail forwarding. Even if you're some distance from the PBBS, with the speed and efficiency of TexNet you'll hardly notice the delay.

## *TCP/IP*

If you're an active packeteer, sooner or later someone will bring up the subject of TCP/IP—Transmission Control Protocol/Internet Protocol. Of all the packet networking alternatives discussed so far, TCP/IP is the most popular. In fact, many packeteers believe that TCP/IP may someday become the standard for amateur packet radio.

Despite its name, TCP/IP is more than two protocols; it's actually a set of several protocols. Together they provide a high level of flexible, "intelligent" packet networking. At the time of this writing, TCP/IP networks are local and regional in nature. For long-distance mail handling, TCP/IP still relies on traditional AX.25 *NET/ROM* networks. Even so, TCP/IP enthusiasts see a future when the entire nation, and perhaps the world, will be linked by high-speed TCP/IP systems using terrestrial microwave and satellites.

Maintaining a packet connection on a *NET/ROM* network can be a difficult proposition—especially if the station is distant. You can only hope that all the nodes in the path are able to relay the packets back and forth. If the one of the nodes becomes unusually busy, your link to the other station could collapse. Even when the path is maintained, your packets are in direct competition with all the other packets on the network. With randomly calculated transmission delays, collisions are inevitable. As a result, the network bogs down, slowing data throughput for everyone.

TCP/IP has a unique solution for busy networks. Rather than transmitting packets at randomly determined intervals, TCP/IP stations automatically *adapt* to network delays as they occur. As network throughput slows down, active TCP/IP stations sense the change and lengthen their transmission delays accordingly. As the network speeds up, the TCP/IP stations shorten their delays to match the pace. This kind of intelligent network sharing virtually guarantees that all packets will reach their destinations with the greatest efficiency the network can provide.

With TCP/IP's adaptive networking scheme, you can chat using the *telnet* protocol with a ham in a distant city and rest assured that you're not overburdening the system. Your packets simply join the constantly moving "freeway" of data. They might slow down in heavy traffic, but they *will* reach their destination eventually. (This adaptive system is used for *all* TCP/IP packets, no matter what they contain.)

TCP/IP excels when it comes to transferring files from one station to another. By using the TCP/IP *file transfer protocol* (ftp), you can connect to another station and transfer computer files—including software. As you can probably guess, transferring large files can take time. With TCP/IP, however, you can still send and receive mail (using the *SMTP* protocol) or talk to another ham *while* the transfer is taking place.

When you attempt to contact another station using TCP/IP, all network routing is performed automatically according to the TCP/IP address of the station you're trying to reach. In fact, TCP/IP networks are transparent to the average user.

On conventional *NET/ROM* networks, access to backbone links is restricted. This isn't true on TCP/IP. Not only are you allowed to use the backbones, you're actually *encouraged* to do so. If you have the necessary equipment to communicate at the proper frequencies and data rates, you can tap into the high-speed TCP/IP backbones directly. By doing so, you'll be able to handle data at much higher rates. This benefits you and everyone else on the network.

To operate TCP/IP, all you need is a computer (it must be a computer, not a terminal), a 2-m FM transceiver and a TNC with *KISS* capability. As you might guess, the heart of your TCP/IP setup is software. The TCP/IP software set was written by Phil Karn, KA9Q, and is called *NOSNET* or just *NOS*.

There are dozens of *NOS* derivatives available today. All are based on the original *NOSNET*. The programs are available primarily for IBM-PCs and compatibles and Macintoshes. You can obtain NOS software from on-line sources such as the CompuServe *HAMNET* forum libraries, Internet ftp sites, Amateur Radio-oriented BBSs and elsewhere. *NOS* takes care of all TCP/IP functions, using your "KISSable" TNC to communicate with the outside world. The only other item you need is your own IP address. Individual IP Address Coordinators assign addresses to new TCP/IP users.

## PACTOR

PACTOR (PT) is an HF radio transmission system developed by German amateurs Hans-Peter Helfert, DL6MAA, and Ulrich Strate, DF4KV. It combines the best of AMTOR and packet to make a system that is superior to both. This description was adapted from PACTOR specifications by the *Handbook* Editor. PACTOR is much faster than AMTOR, yet improves on AMTOR's error-correction scheme. It performs well under both weak-signal and high-noise conditions. PACTOR/AMTOR BBS stations operating in the US and other countries are used by amateurs all over the world. The BBSs respond automatically to both PACTOR and AMTOR calls. PACTOR carries binary data, so it can transfer binary files, ASCII and other symbol sets.

Packet-radio style CRCs (two per packet, 16 bits each) and "ARQ Memory" enable the PT system to reconstruct defective packets by overlaying good and damaged data from different transmissions, which reduces repeats and transmission time. PT's overhead is much less than that of AMTOR. PACTOR uses complete call signs for addressing. The mark/space convention is unnecessary and frequency-shift independent.

### Transmission Formats

#### *Information Blocks*

All packets have the basic structure shown in **Fig 12.31**, and their timing is as shown in **Table 12.7**:

*Header:* contains a fixed bit pattern to simplify repeat requests, synchronization and monitoring. The header is also important for the Memory ARQ function. In each packet carrying new information the bit pattern is inverted.

*Data:* any binary information. The format is specified in the status word. Current choices are 8-bit ASCII or 7-bit ASCII (with Huffman encoding). Characters are not broken across packets. ASCII RS (hex 1E) is used as an IDLE character in both formats.

*Status word:* see **Table 12.8**

*CRC:* The CRC is calculated according to the CCITT standard, for the data, status and CRC.

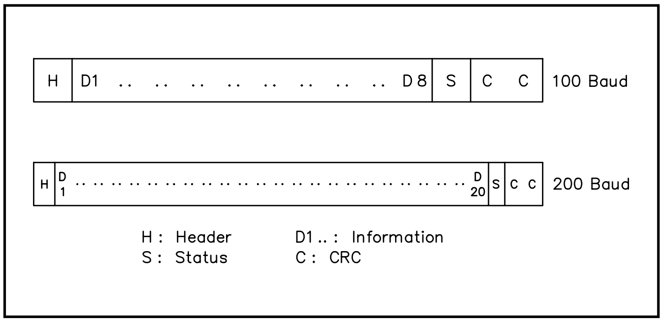



**Fig 12.31—PACTOR data packet format.**

**Table 12.7**
**PACTOR Timing**

| *Object* | *Length (seconds)* |
|---|---|
| Packet | 0.96 (200 bd: 192 bits; 100 bd: 96 bits) |
| CS receive time | 0.29 |
| Control signals | 0.12 (12 bits at 10 ms each) |
| Propagation delay | 0.17 |
| Cycle | 1.25 |

**Table 12.8**
**PACTOR Status Word**

| *Bit* | *Meaning* |
|---|---|
| 0 | Packet count (LSB) |
| 1 | Packet count (MSB) |
| 2 | Data format (LSB) |
| 3 | Data format (MSB) |
| 4 | Not defined |
| 5 | Not defined |
| 6 | Break-in request |
| 7 | QRT request |

***Data Format Bits***

| *Format* | *bit 3* | *bit 2* |
|---|---|---|
| ASCII 8 bit | 0 | 0 |
| Huffman code | 0 | 1 |
| Not defined | 1 | 0 |
| Not defined | 1 | 1 |

Bits 0 and 1 are used as a packet count; successive packets with the same value are identified by the receiver as repeat packets. A modulus-4 count helps with unrecognized control signals, which are unlikely in practice.

### *Acknowledgment Signals*

The PACTOR acknowledgment signals are similar to those used in AMTOR, except for CS4 (see **Table 12.9**). Each of the signals is 12 bits long. The characters differ in pairs in 8 bits (Hamming offset) so that the chance of confusion is reduced. (One of the most common causes of errors in AMTOR is the small CS Hamming offset of 4 bits.)

If the CS is not correctly received, the TX reacts by repeating the last packet. The request status can be uniquely recognized by the 2-bit packet number so that wasteful transmissions of pure RQ blocks are unnecessary.

### *Timing*

The receiver pause between two blocks is 0.29 s. After deducting the CS lengths, 0.17 s remain (just as in AMTOR) for switching and propagation delays so that there is adequate reserve for DX operation.

## Contact Flow

### *Listening*

In the listen mode, the receiver scans any received packets for a CRC match. This method uses a lot of computer processing resources, but it's flexible.

### *CQ*

A station seeking contacts transmits CQ packets in a FEC mode, without pauses for acknowledgment between packets. The transmit time length number of repetitions and speed are the transmit operator's choice. (This mode is also suitable for bulletins and other group traffic.) Once a listening station has copied the call, the listener assumes the TX station role and initiates a contact. Thus, the station sending CQ initially takes the RX station role. The contact begins as shown in **Table 12.10**

### *Speed Changes*

With good conditions, PT's normal signaling rate is 200 baud (for a 600-Hz bandwidth), but the system automatically changes from 200 to 100 baud and back, as conditions demand. In addition, Huffman coding can further increase the throughput by a factor of 1.7. There is no loss of synchronization speed changes; only one packet is repeated.

When the RX receives a bad 200-baud packet, it

**Table 12.9**
**PACTOR Control Signals**

| *Code* | *Chars* (hex) | *Function* |
|---|---|---|
| CS1 | 4D5 | Normal acknowledge |
| CS2 | AB2 | Normal acknowledge |
| CS3 | 34B | Break-in (forms header of first packet from RX to TX) |
| CS4 | D2C | Speed change request |

All control signals are sent only from RX to TX.

**Table 12.10**
**PACTOR Initial Contact**

***Master Initiating Contact***

| | | | |
|---|---|---|---|
| *Size (bytes)* | *1* | *8* | *6* |
| Content | /Header/ | SLAVECAL/ | SLAVECAL/ |
| Speed (bauds) | 100 | 100 | 200 |

***Slave Response***

The receiving station detects a call, determines mark/space polarity, decodes 100-bd and 200-bd call signs. It uses the two call signs to determine if it is being called and the quality of the communication path. The possible responses are:

| | |
|---|---|
| First call sign does not match slave's (Master not calling this slave) | none |
| Only first call sign matches slave's (Master calling this slave, poor communications) | CS1 |
| First and second call signs both match the slaves (good circuit, request speed change to 200 bd) | CS4 |

can acknowledge with CS4. TX immediately assembles the previous packet in 100-baud format and sends it. Thus, one packet is repeated in a change from 200 to 100 baud.

The RX can acknowledge a good 100-baud packet with CS4. TX immediately switches to 200 baud and sends the next packet. There is no packet repeat in an upward speed change.

### *Change of Direction*

The RX station can become the TX station by sending a special change-over packet in response to a valid packet. RX sends CS3 as the first section of the changeover packet. This immediately changes the TX station to RX mode to read the data in that packet and responds with CS1 and CS3 (acknowledge) or CS2 (reject).

### *End of Contact*

PACTOR provides a sure end-of-contact procedure. TX initiates the end of contact by sending a special packet with the QRT bit set in the status word and the call of the RX station in byte-reverse order at 100 baud. The RX station responds with a final CS.

## PACTOR II

This new protocol is a significant improvement over PACTOR; yet it is fully compatible with the older mode. Invented in Germany, PACTOR uses 16PSK to transfer up to 800 bits/s at a 100-baud rate. This keeps the bandwidth less than 500 Hz. Users believe that PACTOR II is faster and more robust than CLOVER.

PACTOR II uses a DSP with Nyquist waveforms, Huffman *and* Markov compression, and powerful Viterbi decoding to increase transfer rate and sensitivity into the noise level. The effective transfer rate of text is over 1200 bits/s. Features of PACTOR II include:

- Frequency agility—It can automatically adjust or lock two signals together over a ±100-Hz window.
- Powerful data reconstruction based upon computer power—with over 2 MB of available memory.
- Cross correlation—applies analog Memory ARQ to acknowledgment frames and headers.
- Soft decision making—Uses artificial intelligence (AI) as well as digital information received to determine frame validity.
- Extended data block length—When transferring large files under good conditions, the data length is doubled to increase the transfer rate.
- Automatic recognition of PACTOR I, PACTOR II and so on, with automatic mode switching.
- Intermodulation products are canceled by the coding system.
- Two long-path modes extend frame timing for long-path terrestrial and satellite propagation paths.

This is a fast, robust mode—possibly the most powerful in the ham bands. It has excellent coding gain as well. It can also communicate with all earlier PACTOR I systems. Like packet and AMTOR stations, PACTOR II stations acknowledge each received data block. Unlike those modes, PACTOR II employs computer logic as well as received data to reassemble defective data blocks into good frames. This reduces the number of transmissions and increases the throughput of the data.

The Sounds of Amateur Radio

**Listen to a PACTOR II QSO in progress.**

## G-TOR

This brief description has been adapted from "A Hybrid ARQ Protocol for Narrow Bandwidth HF Data Communication" by Glenn Prescott, WB0SKX, Phil Anderson, W0XI, Mike Huslig, KB0NYK, and Karl Medcalf, WK5M (May 1994 *QEX*, pp 12-19).

G-TOR is short for Golay-TOR, an innovation of Kantronics, Inc. It's a new HF digital-communication mode for the Amateur Service. G-TOR was inspired by HF Automatic Link Establishment (ALE) concepts and is structured to be compatible with ALE systems when they become available.

The purpose of the G-TOR protocol is to provide an improved digital radio communication capability for the HF bands. The key features of G-TOR are:

- Standard FSK tone pairs (mark and space)
- Link-quality-based signaling rate: 300, 200 or 100 baud
- 2.4-s transmission cycle
- Low overhead within data frames
- Huffman data compression—two types, on demand
- Embedded run-length data compression
- Golay forward-error-correction coding
- Full-frame data interleaving
- CRC error detection with hybrid ARQ
- Error-tolerant "Fuzzy" acknowledgments

The primary benefit of these innovations is increased throughput—that is, more bits communicated in less time. This is achieved because the advanced processing features of G-TOR provide increased resistance to interference and noise and greatly reduce multipath-induced data errors.

The G-TOR protocol is straightforward and relatively easy to implement on existing multimode TNCs.

### Propagation Problems

The miserable propagation conditions characteristic of the HF bands make effective data communication a nightmare. Received signals are often weak and subject to multipath fading; ever-present interference can impair reception. With digital communication, the human brain cannot help interpret the signal. Therefore, we need to incorporate great ingenuity into the receiving system. G-TOR uses modern communication signal processing to help us transmit error-free data via the inherently poor HF communication medium.

Worldwide HF communication may experience interference, multipath fading, random and burst noise. For data communication over the HF bands, three factors dominate: available bandwidth, signaling rate and the dynamic time behavior of the channel.

### ... and Answers

Transmission bandwidths of 500 Hz or less minimize the effects of multipath propagation and man-made interference. G-TOR transmits at 300 baud or less, with maximum separation of 200 Hz, for a bandwidth just slightly greater than 500 Hz.

The FCC does not currently permit symbol rates greater than 300 symbols per second (baud) on most HF bands. This is a reasonable limit because multipath propagation can become a serious problem with faster rates.

The HF channel has a characteristic dynamic time behavior: Conditions can change significantly in a few seconds. This indicates an optimum data-transmission length (usually 1 s or less). G-TOR transmissions are nearly 2 s long because the signal-processing techniques can overcome some propagation change.

### The G-TOR Protocol

Since one of the objectives of this protocol is ease of implementation in existing TNCs, the modulation format consists of standard tone pairs (FSK), operating at 300, 200 or 100 baud, depending upon channel conditions. (G-TOR initiates contacts and sends ACKs only at 100 baud.) FSK was chosen for economy and simplicity, but primarily because many hams already have FSK equipment.

The G-TOR waveform consists of two phase-continuous tones (BFSK) spaced 200 Hz apart (mark = 1600 Hz, space = 1800 Hz); however, the system can still operate at the familiar 170-Hz shift (mark = 2125 Hz, space = 2295 Hz), or with any other convenient tone pairs. The optimum spacing for 300-baud transmission is 300 Hz, but we trade some performance for a narrower bandwidth.

Each transmission consists of a synchronous ARQ 1.92-s frame and a 0.48-s interval for propagation and ACK transmissions (2.4 s cycle). All advanced protocol features are implemented in the signal-processing software.

Synchronous operation increases the system throughput during multipath fading and keeps overhead to a minimum. Synchronization is performed using the received data and precise timing.

#### *Frame Structures*

***Data Frames***—The basic G-TOR frame structure (see **Fig 12.32**) uses multiple 24-bit (triple-byte) words for compatibility with the Golay encoder. Data frames are composed of 72 (300 baud), 48 (200 baud) or 24 (100 baud) data bytes, depending upon channel conditions.

A single byte before the CRC carries command and status information:

status bits 7 and 6: Command
- 00 - data
- 01 - turnaround request
- 10 - disconnect
- 11 - connect

status bits 5 and 4: Unused
- 00 - reserved

status bits 3 and 2: Compression
- 00 - none
- 01 - Huffman (A)
- 10 - Huffman (B)
- 11 - reserved

status bits 1 and 0: Frame no. ID

The error-detection code transmitted with each frame is a 2-byte cyclic redundancy check (CRC) code—the same used in AX.25. A CRC calculation determines if error correction is needed, and another tests the result.

The connect and disconnect frames are essentially identical in structure to the data frame and contain the call signs of both stations.

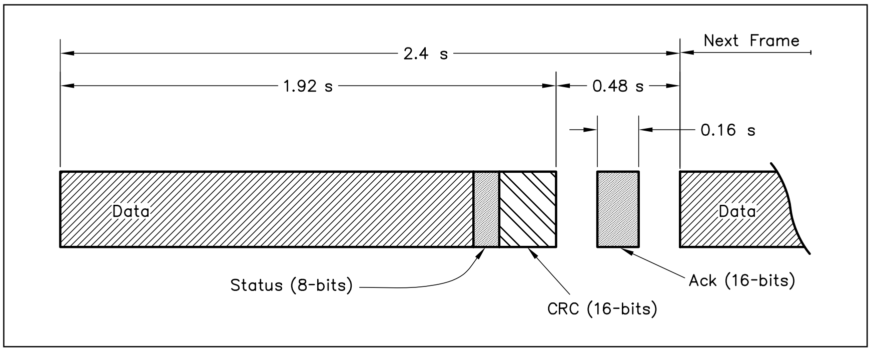



**Fig 12.32—G-TOR ARQ system timing and frame structure before interleaving. The data portion may be 69 (300 baud), 45 (200 baud) or 21 (100 baud) bytes depending on the channel quality.**

***ACK Frames***—G-TOR ACK frames are not interleaved and do not contain error-correction (parity) bits. There are five different ACK frames:

- Frame received correctly (send next data frame)
- Frame error detected (please repeat)
- Speed-up
- Speed-down
- Changeover

The ACK codes are composed of multiple cyclic shifts of a single 15-bit pseudorandom noise (PN) sequence (plus an extra 0 bit to fill 16 bits). PN sequences have powerful properties that facilitate identification of the appropriate ACK code, even in the presence of noise and interference. We refer to this concept as a “fuzzy” ACK, in that it tolerates 3 bit errors within a received ACK frame.

*Change-over frames* are essentially data frames in which the first 16 bits of data is the ACK changeover PN code.

### Data Compression

Data compression is used to remove redundancy from source data. Therefore, fewer bits are needed to convey any given message. This increases data throughput and decreases transmission time—valuable features for HF. G-TOR uses run-length coding and two types of Huffman coding during normal text transmissions. Run-length coding is used when more than two repetitions of an 8-bit character are sent. It provides an especially large savings in total transmission time when repeated characters are being transferred.

The Huffman code works best when the statistics of the data are known. G-TOR applies Huffman A coding with the upper- and lower-case character set, and Huffman B coding with upper-case-only text. Either type of Huffman code reduces the average number of bits sent per character. In some situations, however, there is no benefit from Huffman coding. The encoding process is then disabled. This decision is made on a frame-by-frame basis by the information-sending station.

### Golay Coding

The real power of G-TOR resides in the properties of the (24,12) extended Golay error-correcting code, which permits correction of up to three random errors in three received bytes. The (24,12) extended Golay code is a half-rate error-correcting code: Each 12 data bits are translated into an additional 12 parity bits (24 bits total). Further, the code can be implemented to produce separate input-data and parity-bit frames.

The extended Golay code is used for G-TOR because the encoder and decoder are simple to implement in software. Also, Golay code has mathematical properties that make it an ideal choice for short-cycle synchronous communication:

- The rare property of self-duality makes the code “invertible”; that is, the original data can be recovered by simply recoding the parity bits.
- Because of the linear block code structure of the Golay code, the encoder and decoder can be implemented using a simple table look-up procedure. An alternative decoder implementation uses the well-known Kasami decoding algorithm, which requires far less memory than the look-up table.

Error-correction coding inserts some redundancy into each (triple-byte) word so that errors occurring in the receiving process can be corrected. However, most error-correcting codes are effective at correcting only random errors. Burst errors from lightning or interference exceed the capabilities of most error-correcting codes.

## Interleaving

The conventional solution is called “interleaving.” Interleaving (the very last operation performed before transmission and first performed upon reception) rearranges the bit order to randomize the effects of long error bursts.

The interleaving process reads 12-bit words into registers by columns and reads 48-bit words out by rows; see **Fig 12.33**. The deinterleaver simply performs the inverse, reading the received data bits into the registers by row and extracting the original data sequence by reading the columns. If a long burst of errors occurs—say, 12 bits in length—the errors will be distributed into 48 separate 12-bit words before error correction is applied, thus effectively nullifying the long burst. Both data and parity frames are completely interleaved.

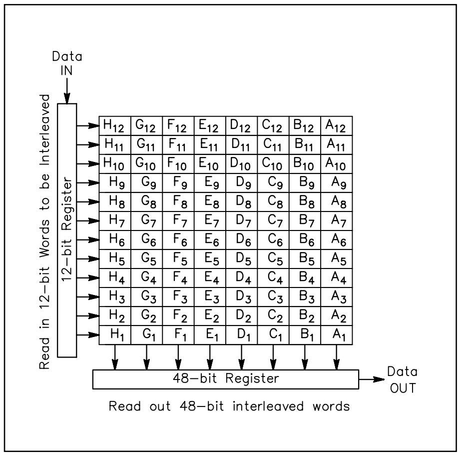



**Fig 12.33—Interleaving the bits to be transmitted.**

## Hybrid ARQ

G-TOR combines error detection and forward error correction with ARQ. Hybrid-ARQ uses a CRC to check for errors in every frame. Only when errors are found; does G-TOR use forward error correction (a relatively slow process) to recover the data.

The half-rate invertible Golay code provides an interesting dimension to the hybrid-ARQ procedure. With separate data and parity frames, both of which can supply the complete data, G-TOR frames alternate between data and parity frames.

When the receiver detects an error and requests a retransmission, the sending station sends the complementary portion of the frame (data or parity).

When the complementary frame arrives, it is processed and checked for errors. If it checks, the data is accepted and a new frame is requested. If it fails the CRC check, the two frames are combined, corrected and checked.

Using this scheme, two transmissions provide three independent chances to correct any errors. If this process still fails, a retransmission is requested.

## G-TOR Performance

Initial testing with G-TOR was conducted during January 1994, between Lawrence, Kansas, and Laguna Niguel, California. During these tests, TRACE was set ON at each station, enabling the raw data display of frames received with and without the aid of forward error correction and interleaving. The results were somewhat surprising. While PACTOR often dropped in transmission speed from 200 to 100 bauds, G-TOR nearly always operated at 300 bauds. Enough frames were corrected to keep the system running at maximum speed, regardless of man-made interference and mild multipath conditions. Transfer duration for the entire test files varied from 12 to 27 minutes for PACTOR, but only 5.5 to 7.5 minutes for all but one G-TOR transfer. G-TOR simply maintained its highest pace better than PACTOR, resulting in a substantial increase in average throughput.

On-air tests have shown G-TOR to have the ability to “hang in there” when channel conditions get tough. The time required to send a given binary file tends to be much less for G-TOR than for PACTOR.

This protocol should continue to be valuable when DSP-based TNCs become widely available. G-TOR has the essential characteristics to be a useful protocol for years to come.

See “A Comparison of HF Digital Protocols” in Jul 1996 *QST* for an overview of performance tradeoffs between the numerous competing protocols available.

## CLOVER-II

The desire to send data via HF radio at high data rates and the problems encountered when using AX.25 packet radio on HF radio led Ray Petit, W7GHM, to develop a unique modulation waveform and data transfer protocol that is now called "CLOVER-II." Bill Henry, K9GWT, supplied this description of the Clover-II system. CLOVER modulation is characterized by the following key parameters:

- Very low base symbol rate: 31.25 symbols/second (all modes).
- Time-sequence of amplitude-shaped pulses to provide a very narrow frequency spectra. Occupied bandwidth = 500 Hz at 50 dB below peak output level.
- Differential modulation between pulses.
- Multilevel modulation.

The low base symbol rate is very resistant to multipath distortion because the time between modulation transitions is much longer than even the worst-case time-smearing caused by summing of multipath signals. By using a time-sequence of tone pulses, Dolph-Chebychev "windowing" of the modulating signal and differential modulation, the total occupied bandwidth of a CLOVER-II signal is held to 500 Hz.

### The CLOVER Waveform

Multilevel tone, phase and amplitude modulation give CLOVER a large selection of data modes that may be used (see **Table 12.11**). The adaptive ARQ mode of CLOVER senses current ionosphere conditions and automatically adjusts the modulation mode to produce maximum data throughput. When using the "Fast" bias setting, ARQ throughput automatically varies from 11.6 bytes/s (1.7 times AMTOR) to 70 bytes/s (10.5 times AMTOR).

The CLOVER-II waveform uses four tone pulses that are spaced in frequency by 125 Hz. The time and frequency domain characteristics of CLOVER modulation are shown in **Figs 12.34**, **12.35** and **12.36**. The time-domain shape of each tone pulse is intentionally shaped to produce a very compact frequency spectra. The four tone pulses are spaced in time and then combined to produce the composite output shown. Unlike other modulation schemes, the CLOVER modulation spectra is the same for all modulation modes.

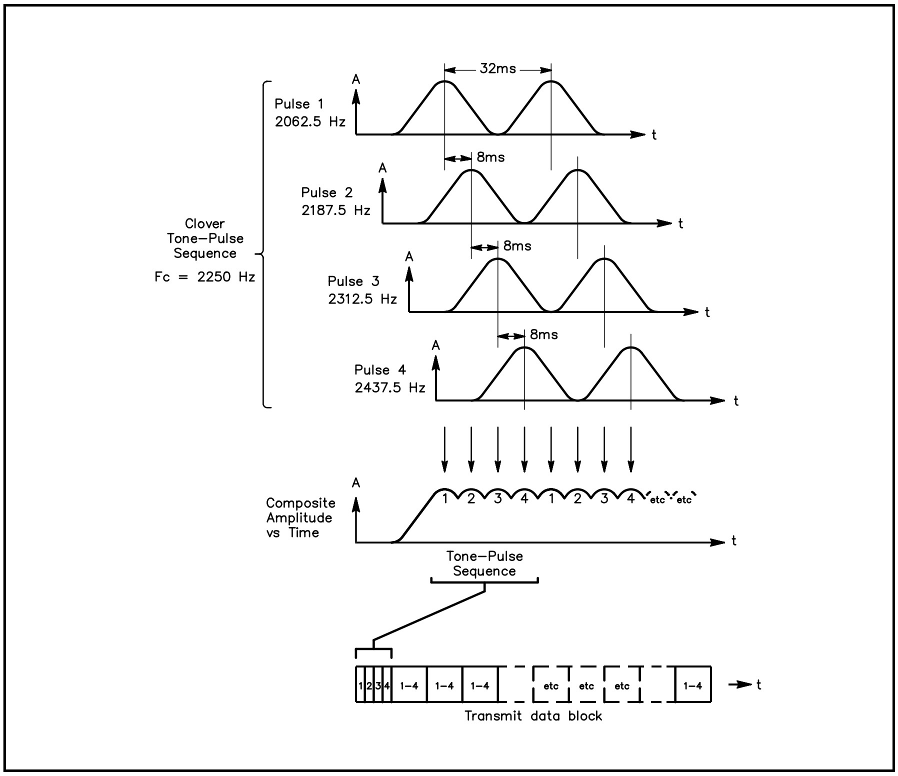
#### *Modulation*

Data is modulated on a CLOVER-II signal by varying the phase and/or amplitude of the tone pulses. Further, all data modulation is differential on the same tone pulse; data is represented by the phase (or amplitude) difference from one pulse to the next. For example, when binary phase modulation is used, a data change from "0" to "1" may be represented by a change in the phase of tone pulse 1 by 180° between the first and second occurrence of that pulse. Further, the phase state is changed only while the pulse amplitude is zero. Therefore, the wide frequency spectra normally associated with PSK of a continuous carrier is

**Table 12.11**
**CLOVER-II Modulation Modes**

As presently implemented, CLOVER-II supports a total of 7 different modulation formats: 5 using PSM and 2 using a combination of PSM and ASM (Amplitude Shift Modulation).

| *Name* | *Description* | *In-Block Data Rate* |
|---|---|---|
| 16P4A | 16 PSM, 4-ASM | 750 bps |
| 16PSM | 16 PSM | 500 bps |
| 8P2A | 8 PSM, 2-ASM | 500 bps |
| 8PSM | 8 PSM | 375 bps |
| QPSM | 4 PSM | 250 bps |
| BPSM | Binary PSM | 125 bps |
| 2DPSM | 2-Channel Diversity BPSM | 62.5 bps |



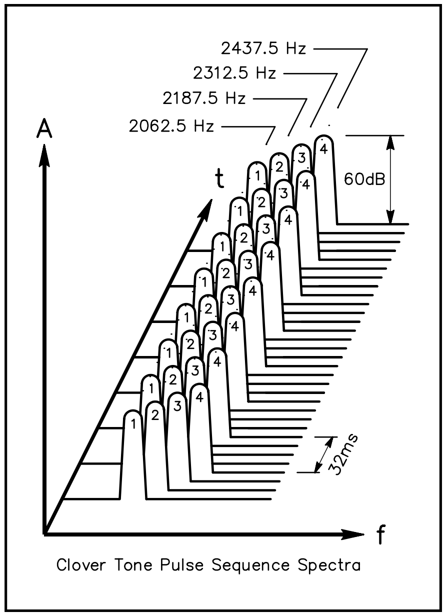




**Fig 12.35—A frequency-domain plot of a CLOVER-II waveform.**

**Fig 12.34—Ampitude vs time plots for CLOVER-II's four-tone waveform.**

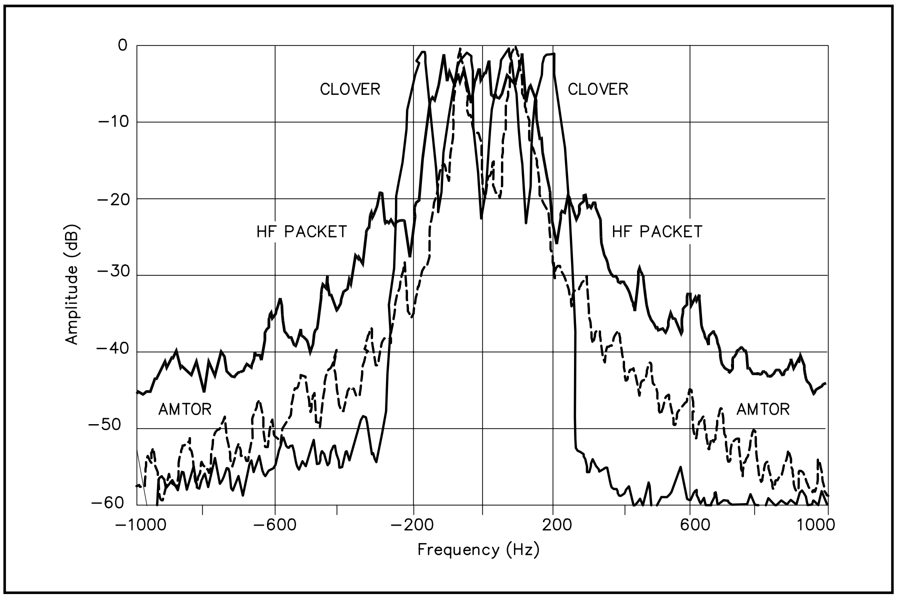



**Fig 12.36—Spectra plots of AMTOR, HF packet-radio and CLOVER-II signals.**

avoided. This is true for all CLOVER-II modulation formats. The term "phase-shift modulation" (PSM) is used when describing CLOVER modes to emphasize this distinction.

### *Coder Efficiency Choices*

CLOVER-II has four "coder efficiency" options: 60%, 75%, 90% and 100% ("efficiency" being the approximate ratio of real data bytes to total bytes sent). "60% efficiency" corrects the most errors but has the lowest net data throughput. "100% efficiency" turns the encoder off and has the highest throughput but fixes no errors. There is therefore a tradeoff between raw data throughput vs the number of errors that can be corrected without resorting to retransmission of the entire data block.

Note that while the "In Block Data Rate" numbers listed in the table go as high as 750 bps, overhead reduces the net throughput or overall efficiency of a CLOVER transmission. The FEC coder efficiency setting and protocol requirements of FEC and ARQ modes add overhead and reduce the net efficiency.

Table 12.12 and Table 12.13 detail the relationships between block size, coder efficiency, data bytes per block and correctable byte errors per block.

## Multilevel Digital Modulation Waveforms

Digital waveforms discussed so far have all used either on/off keying (OOK, that is Morse code, or CW) or frequency-shift keying (FSK, RTTY, AMTOR, PACTOR and packet radio). Both OOK and FSK are "simple" digital modulation waveforms; they have only two binary states that are represented by two radio-frequency states. In Morse code, the states are key-down = logical "1" and key-up = logical "0." In RTTY, AMTOR, PACTOR and packet radio, one frequency is "1" state, another is the "0" state.

More efficient use may be made of the spectrum by using multilevel modulation, in which one change in the transmitted signal may represent two or more bits of data. A simple example of multi-level modulation is quadrature phase-shift keying, known as QPSK. The simplest QPSK signal transmits a continuous carrier at a single frequency. Digital information is modulated on this carrier by changing the phase shift in 90° increments. Since there are four possible 90°-increment states (0°, 90°, 180° and 270°), four different modulation states may be signaled. Put another way, each phase state may be used to represent two bits of binary data. Examples of four common PSK modes are shown in **Fig A**.

The Sounds of Amateur Radio

**Listen to 1200-baud PSK packet transmissions from OSCAR 16.**

Note that the phase of the transmitter carrier may be changed from any given state to any other state. Thus when using QPSK, if two bits of data change from "00" to "11," only one change to the transmitter carrier phase is required—from 0° to 270°. This observation illustrates the very important difference between modulation symbol rate (bauds) and data throughput rate (in bits-per-second, bits/s). In QPSK, bauds = 0.5 × bits/s (100 baud = 200 bits/s). This concept can be extended to 8PSK which has 8 phase states that represent 3 bits of data (bits/s = 3 × bauds). Carried further, each phase state in 16PSK modulation represents 4 bits of binary data and the throughput is 4 times the base symbol rate (bit/s = 4 × bauds).

Higher-level phase shift modulation schemes have been used (32PSK and 64PSK for example), but these systems require much more complex demodulator design. In particular, demodulator sensitivity to noise and distortion increases greatly as the number of possible phase states is increased. Consider the relatively simple QPSK example. The design-center phase states of 0°, 90°, 180°, and 270° represent the four possible modulation conditions. Ionosphere propagation, multipath signal reflections, and transmission distortion all conspire to insert phase "jitter" or uncertainty in the received signal. In QPSK, signals with a phase shift between 45° and 135° can be assumed to represent the 90° state, 135° to 225° for the 180° state and so on. The margin for error or "phase margin" for QPSK is ±45°. A similar calculation for 16PSK shows that its phase margin is just ±12.25°. If we consider use of a 10.000 MHz carrier with 16PSK, the period of the

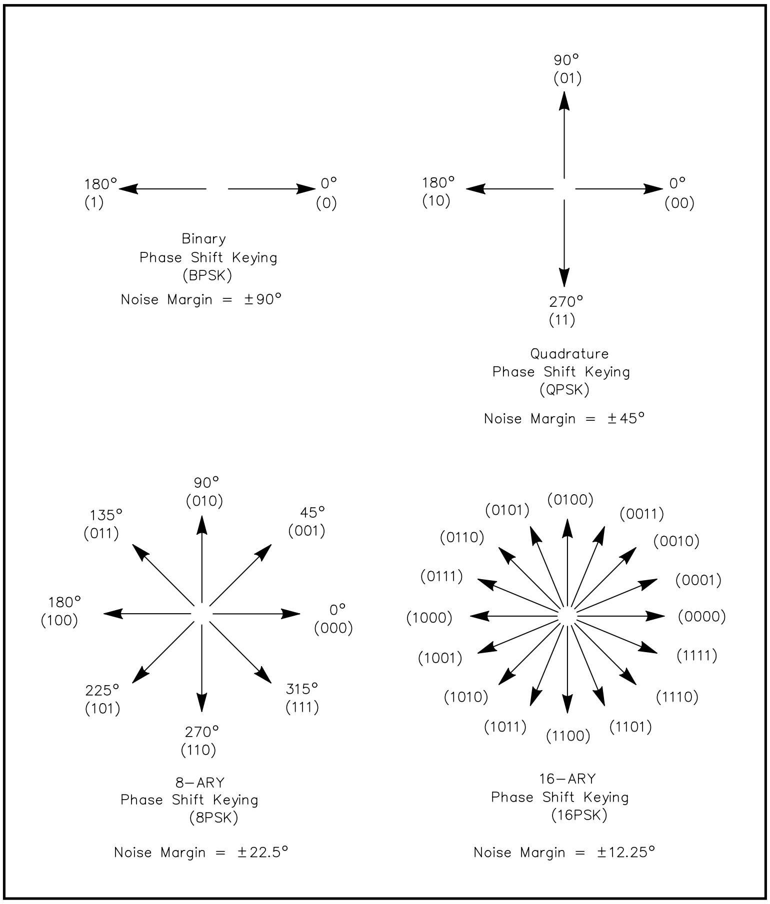


**Fig A**

carrier sine wave is 0.100 microsecond and the allowable phase jitter corresponds to a time uncertainty of only ±0.003403 ms, or ±3.403 ns. Obviously, very stable phase references must be used in a 16PSK system and it does not take very much distortion or noise to make correct data detection impossible. However, such systems are commonly used in telephone-line modems.

Telephone modems carry the multilevel concept one step further and use amplitude-level modulation (amplitude-shift keying, ASK) in addition to PSK modulation. If two-level ASK is used with 16PSK, a total of 32 states may be sent. Similarly, use of 4ASK and 16PSK gives 64 unique states for each modulation change. This is commonly called "QAM" for Quadrature Amplitude Modulation and is the modulation used by most 9600-baud telephone modems. Each modulation change can represent the state of 6 bits of binary data; the data throughput is 6 times the base modulation symbol rate (bits/s = 6 × bauds). As noted above, complex multilevel modulation schemes require very complex and expensive demodulators that are very susceptible to noise and distortion. Fortunately, modern telephone lines are relatively noise-free and stable. By use of error correction and line distortion equalization, high "speed" data transmission via telephone line is now in common use.

Unfortunately, these same techniques cannot be directly applied to radio data transmission, particularly to HF signals. Long-range HF signals are propagated via the ionosphere, which is not stable or well-defined from instant to instant. Ionosphere reflection height and signal attenuation varies widely with time of day, geographic location, and solar activity. Moreover, noise levels on HF vary considerably with location as well as time of day.

With multipath propagation, multiple copies of an original signal are summed at the receiving antenna. Since each signal travels via a different path, the propagation delays are different. Multiple signals therefore arrive at the receiver at slightly different times and the "mark-to-space" transition time is different for each signal. This causes "smearing" of the exact transition times. Multipath distortion occurs commonly on HF when both single-hop and multiple-hop signals arrive at the receiving antenna with similar strengths. Multipath distortion is also common on VHF and UHF signals in highly populated areas where large buildings provide reflecting surfaces.

The HF environment is therefore complicated and hostile to data transmission. Modulation techniques that work well on stable and predictable telephone lines may also be usable for VHF and UHF radio systems, but they may seldom be directly applied to HF data radio systems. Further, data format protocols that were devised for the stable phone-line environment are generally not optimum for use on HF data radio. For example, both the FSK modulation and the protocol used for AX.25 packet radio lead to serious problems when used on HF signals.

**Table 12.12**
**Data Bytes Transmitted Per Block**

| | *Reed-Solomon Encoder Efficiency* | | | |
|---|---|---|---|---|
| *Block Size* | *60%* | *75%* | *90%* | *100%* |
| 17 | 8 | 10 | 12 | 14 |
| 51 | 28 | 36 | 42 | 48 |
| 85 | 48 | 60 | 74 | 82 |
| 255 | 150 | 188 | 226 | 252 |

**Table 12.13**
**Correctable Byte Errors Per Block**

| | *Reed-Solomon Encoder Efficiency* | | | |
|---|---|---|---|---|
| *Block Size* | *60%* | *75%* | *90%* | *100%* |
| 17 | 1 | 1 | 0 | 0 |
| 51 | 9 | 5 | 2 | 0 |
| 85 | 16 | 10 | 3 | 0 |
| 255 | 50 | 31 | 12 | 0 |

## CLOVER FEC

All modes of CLOVER-II use Reed-Solomon forward error correction (FEC) data encoding which allows the receiving station to correct errors without requiring a repeat transmission. This is a very powerful error correction technique that is not available in other common HF data modes such as AX.25 packet radio or AMTOR ARQ mode.

## CLOVER ARQ

Reed-Solomon data coding is the primary means by which errors are corrected in CLOVER "FEC" mode (also called "broadcast mode"). In ARQ mode, CLOVER-II employs a three-step strategy to combat errors. First, channel parameters are measured and the modulation format is adjusted to minimize errors and maximize data throughput. This is called the "Adaptive ARQ Mode" of CLOVER-II. Second, Reed-Solomon encoding is used to correct a limited number of byte errors per transmitted block. Finally, only those data blocks in which errors exceed the capacity of the Reed-Solomon decoder are repeated (selective block repeat). Unlike AX.25 packet radio, CLOVER-II does not repeat blocks which have been received correctly.

With seven different modulation formats, four data block lengths (17, 51, 85 or 255 bytes) and four Reed-Solomon coder efficiencies (60%, 75%, 90% and 100%), there are 112 (7 × 4 × 4) different waveform modes that could be used to send data via CLOVER. Once all of the determining factors are considered, however, there are 8 different waveform combinations which are actually used for FEC and/or ARQ modes.

## CLOVER vs AMTOR vs Packet

**Fig 12.36** shows the modulator output spectra of CLOVER-II, AMTOR and HF packet radio. Nearly all of the CLOVER-II signal energy is concentrated within ±250 Hz of the center frequency. Therefore, CLOVER-II signals can be spaced as closely as 500 Hz from any data-mode signal with very little cochannel interference. Tests show that "cross-talk" between two 500-Hz spaced CLOVER-II signals is less than 50 dB. This is much better than the common spacing of AMTOR (1000 Hz) or HF packet signals (2000 Hz).

**Fig 12.37** shows throughput vs S/N for AMTOR and various modes of CLOVER-II. For all values of S/N and all modes of CLOVER, the data throughput obtainable using CLOVER-II is higher than that achievable when using AMTOR. In addition, CLOVER may be used to send full 8-bit computer data whereas AMTOR is restricted to either the Baudot RTTY characters set (CCIR-476/625) or the printable subset of ASCII (ASCII-over-AMTOR).

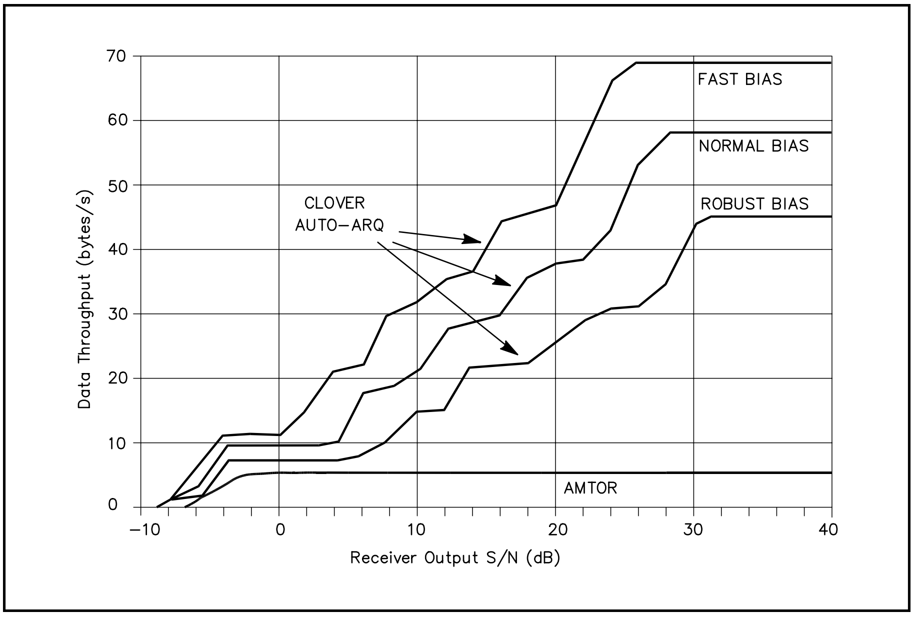



**Fig 12.37—ARQ-mode data throughput vs receiver S/N ratio for AMTOR and three different CLOVER-II configurations.**

RTTY has better automatic receive decoding performance than Morse code and is relatively inexpensive, but offers no automatic error correction. AMTOR includes error correction, has good performance under weak signal conditions and is relatively inexpensive. However, its maximum data throughput rate is low and it cannot support transmission of 8-bit data files.

AX.25 packet radio is inexpensive but its performance on HF is typically very poor. This is due both to the popular choice of modulation (200 Hz shift, 300 baud FSK) and the AX.25 protocol which was not designed to handle the burst-type errors that are common to HF propagation. The MIL-188/110A (now proposed Federal Standard pFS-1052) "Serial, Single-Tone" waveform works well on HF and can pass error-corrected 8-bit data with a throughput of up to 2400 baud. However, modems for this mode are presently very expensive, the occupied bandwidth of 3000 Hz is very wide, and ARQ or adaptive ARQ modes are still under development.

In comparison, CLOVER-II modems are moderately expensive but will adaptively match existing signal conditions and provide high data throughput rates when conditions permit. CLOVER-II will pass full 8-bit data and a CLOVER signal is the most bandwidth efficient of all modes considered.

### *How do They Compare?*

An extensive comparison of digital modes was written by Tim Riley; Dennis Bodson, W4PWF; Stephen Rieman and Teresa Sparkman. See "A Comparison of HF Digital Protocols," *QST*, July 1996, page 35.

## CLOVER-2000

CLOVER-2000 is a faster version of CLOVER (about four times faster) that uses eight tone pulses, each of which is 250-Hz wide, spaced at 250-Hz centers, contained within the 2-kHz bandwidth between 500 and 2500 Hz. The eight tone pulses are sequential, with only one tone being present at any instant and each tone lasting 2 ms. Each frame consists of eight tone pulses lasting a total of 16 ms, so the base modulation rate of a CLOVER-2000 signal is always 62.5 symbols per second (regardless of the type of modulation being used). CLOVER-2000's maximum raw data rate is 3000 bits per second. Allowing for overhead, CLOVER-2000 can deliver error-corrected data over a standard HF SSB radio channel at up to 1994 bits per second, or 249 charcaters (8-bit bytes) per second. These are the uncompressed data rates; the maximum throughput is typically doubled for plain text if compression is used. The effective data throughput rate of CLOVER-2000 can be even higher when binary file transfer mode is used with data compression.

The binary file transfer protocol used by HAL Communications operates with a terminal program explained in the HAL E2004 engineering document listed under references. Data compression algorithms tend to be context sensitive—compression that works well for one mode (e.g. text), may not work well for other data forms (graphics, etc.). The HAL terminal program uses the PK-WARE compression algorithm which has proved to be a good general-purpose compressor for most computer files and programs. Other algorithms may be much more efficient for some data formats, particularly for compression of graphic image files and digitized voice data. The HAL Communications CLOVER-2000 modems can be operated with other data compression algorithms in the users' computers.

CLOVER-2000 is similar to the previous version of CLOVER, including the transmission protocols and Reed-Solomon error detection and correction algorithm. The original descriptions of the CLOVER Control Block (CCB) and Error Correction Block (ECB) still apply for CLOVER-2000, except for the higher data rates inherent to CLOVER-2000. Just like CLOVER, all data sent via CLOVER-2000 is encoded as 8-bit data bytes and the error-correction coding and modulation formatting processes are transparent to the data stream—every bit of source data is delivered to the receiving terminal without modification. Control characters and special "escape sequences" are not required or used by CLOVER-2000. Compressed or encrypted data may therefore be sent without the need to insert (and filter) additional control characters and without concern for data integrity. Five different types of modulation may be used in the ARQ mode—BPSM (Binary Phase Shift Modulation), QPSM (Quadrature PSM), 8PSM (8-level PSM), 8P2A (8PSM + 2-level Amplitude-Shift Modulation), and 16P4A (16 PSM plus 4 ASM).

The same five types of modulation used in ARQ mode are also available in Broadcast (FEC) mode, with the addition of 2-Channel Diversity BPSM (2DPSM). Each CCB is sent using 2DPSM modulation, 17-byte block size, and 60% bias. The maximum ARQ data throughput varies from 336 bits per second for BPSM to 1992 bits per second for 16P4A modulation. BPSM is most useful for weak and badly distorted data signals while the highest format (16P4A) needs extremely good channels, with high SNRs and almost no multipath.

Most ARQ protocols designed for use with HF radio systems can send data in only one direction at a time. For example, when using CCIR-476/625 (SITOR) or PACTOR, one station sends all of its data, ending the transmission with an "OVER" command. The second station may then send its information. Because CLOVER-2000 does not need an "OVER" command, data may flow in either direction at any time. The CLOVER ARQ time frame automatically adjusts to match the data volume to be sent in either or both directions. When first linked, both sides of the ARQ link exchange information using six bytes of the CCB. When one station has a large volume of data buffered and ready to send, ARQ mode automatically shifts to an expanded time frame during which one or more 255 byte data blocks are sent. If the second station also has a large volume of data buffered and ready to send, its half of the ARQ frame is also expanded. Either or both stations will shift back to CCB level when all buffered data has been sent.

This feature provides the benefit of full-duplex data transfer but requires use of only simplex frequencies and half-duplex radio equipment. This two-way feature of CLOVER can also provide a back-channel order-wire capability. Communications may be maintained in this “chat” mode at 55 words per minute, which is more than adequate for real-time keyboard-to-keyboard communications.

Two different CLOVER-2000 modems are available from HAL Communications, the PCI-4000/2K and the DSP-4100/2K. The PCI-4000/2K is for use inside dedicated desk-top personal computers. The PCI-4000/2K may be installed in any IBM-compatible personal computer that uses an 80386 or faster microprocessor (386, 486, Pentium, etc.) and supports the ISA PC plug-in card bus. The DSP-4100/2K is for connection to a laptop or non-IBM PC, since it is a stand-alone DSP modem that may be used with any computer or data terminal having an RS-232 port.

## PSK31

Peter Martinez, G3PLX, who was instrumental in bringing us AMTOR, developed PSK31 for real time keyboard-to-keyboard QSOs. This section was adapted from an article in *RadCom*, Jan 1999. The name derives from the modulation type (phase shift keying) and the data rate, which is actually 31.25 bauds. PSK31 is a robust mode for HF communications that features the 128 ASCII (Internet) characters and the full 256 ANSI character set. This mode works well for two-way QSOs and for nets. Time will tell if PSK31 will replace Baudot RTTY on the amateur HF bands.

Morse code uses a single carrier frequency keyed on and off as dits and dahs to form characters. RTTY code shifts between two frequencies one for *mark* (1) the other for *space* (0). Sequences of marks and spaces comprise the various characters.

Martinez devised a new variable-length code for PSK31 that combines the best of Morse and RTTY. He calls it *Varicode* because a varying number of bits are used for each character (see **Fig A**). Much like the Morse code, the more commonly used letters have shorter codes.

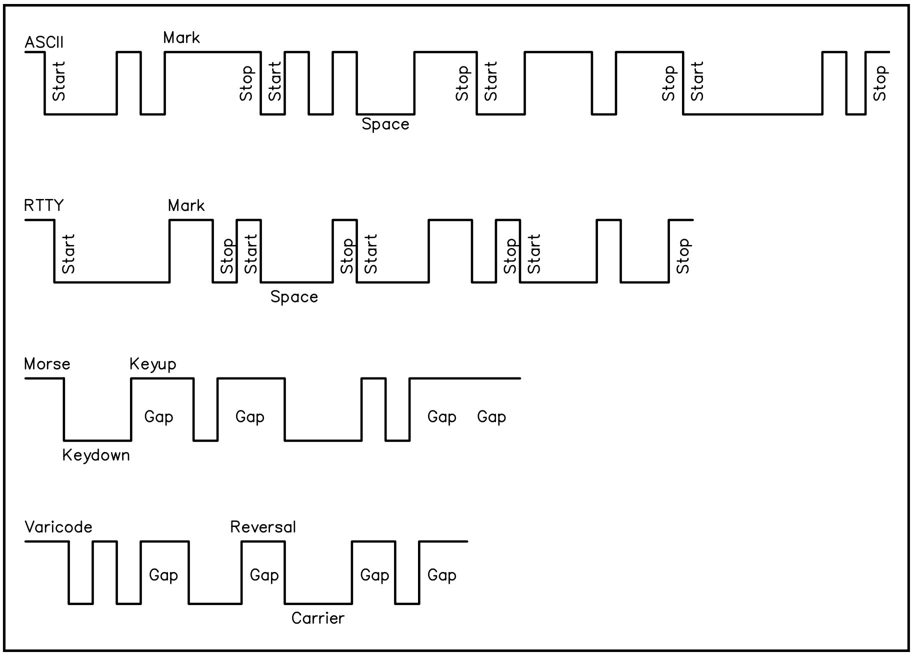



**Fig A—Codes for the word "ten" in ASCII, Baudot, Morse and Varicode.**

As with RTTY, there is a need to signal the gaps between characters. The Varicode does this by using "00" to represent a gap. The Varicode is structured so that two zeros never appear together in any of the combinations of 1s and 0s that make up the characters. In on-the-air tests, Martinez has verified that the unique "00" sequence works significantly better than RTTY's stop code for keeping the receiver synchronized.

With Varicode, a typing speed of about 50 words per minute requires a 32 bit/s transmission rate. Martinez chose 31.25 bit/s because it can be easily derived from the 8-kHz sample rate used in many DSP systems.

The shifting carrier phase generates sidebands 31.25 Hz from the carrier. These are used to synchronize the receiver with the transmitter. The required bandwidth is less than that for the FSK signal of 100 baud Baudot RTTY, as shown in **Fig B**.

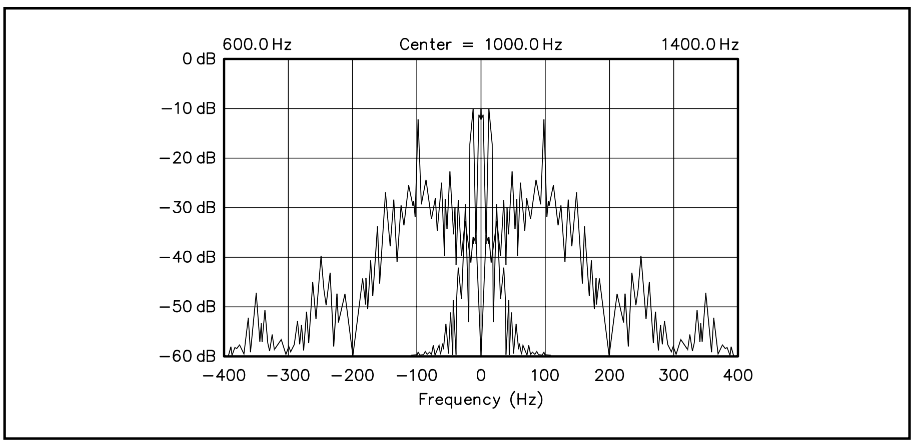



**Fig B—The spectrum of a PSK31 signal compared to that of a 100 baud, 200-Hz-shift FSK signal.**

## ERROR CORRECTION

Martinez has added error correction to PSK31 by using QPSK (quatenary phase shift keying) and a *convolutional encoder* to generate one of four different phase shifts that corresond to patterns of five successive data bits. At the receiving end, a Viterbi decoder is used to correct errors. There are 32 possible sequences for five bits. The Viterbi decoder tracks these possibilities while discarding the least likely and retaining the most likely sequences. Retained sequences are given a score that is based on the running total. The most accurate sequence is reported, and thus errors are corrected.

Operating PSK31 in the QPSK mode should result in 100% copy under most conditions, but at a price. Tuning is twice as critical as it is with BPSK. An accuracy of less than 4 Hz is required for the Viterbi decoder to function properly.

## GETTING STARTED

In addition to a transceiver and antenna, you only need a computer with a Windows operating system and a 16-bit sound card to receive and transmit PSK31. Additional information and software is available for free download over the Web. Use a search engine to find PSK31 information and links to downloads.

# Image Modes

## FACSIMILE

This section, by Dennis Bodson, W4PWF, and Steven Karty, N5SK, covers several facsimile systems in most common Amateur Radio use today. For further information on the area of facsimile, its history, and the development of related standards associated with this mode, refer to *FAX: Facsimile Technology and Systems*.[1] The subject of Weather fax, while of interest to many amateurs, is not a primary activity of the Amateur Radio Service. Information on this subject is contained in the *Weather Satellite Handbook*.[2]

### FACSIMILE OVERVIEW

Facsimile (fax) is a method for transmitting very high resolution still pictures using voice bandwidth radio circuits. The narrow bandwidth of the fax signal, equivalent to SSTV, provides the potential for worldwide communications on the HF bands. Fax is the oldest of the image-transmitting technologies and has been the primary method of transmitting newspaper photos and weather charts. Fax is also used to transmit high-resolution cloud images from both polar-orbit and geostationary satellites. Many of these images are retransmitted using fax on the HF bands.

The resolution of typical fax images greatly exceeds what can be obtained using SSTV or even conventional television (typical images will be made up of 800 to 1600 scanning lines). This high resolution is achieved by slowing down the rate at which the lines are transmitted, resulting in image transmission times of 4 to 10 minutes. Prior to the advent of digital technology, the only practical way to display such images was to print each line directly to paper as it arrived. The mechanical systems for accomplishing this are known as facsimile *recorders* and are based on either photographic media (a modulated light source exposing film or paper) or various types of direct printing technologies including electrostatic and electrolytic papers.

Modern desktop computers have virtually eliminated bulky fax recorders from most amateur installations. Now the incoming image can be stored in computer memory and viewed on a standard TV monitor or a high-resolution computer graphics display. The use of a color display system makes it entirely practical to transmit color fax images when band conditions permit. The same computer-based system that handles fax images is often capable of SSTV operation as well, blurring what was once a clear distinction between the two modes. The advent of the personal computer has provided amateurs with a wide range of options within a single imaging installation. SSTV images of low or moderate resolution can be transmitted when crowded band conditions favor short frame transmission times. When band conditions are stable and interference levels are low, the ability to transmit very high resolution fax images is just a few keystrokes away!

### HARDWARE AND SOFTWARE

Electromechanical fax equipment has been replaced by personal computer hardware and software. The computer allows reception and transmission of various line per minute rates and indices of cooperation by simply pressing a key or by pointing and clicking a mouse. Many fax programs are available as either commercial software or shareware. Usually, the shareware packages (and often trial versions of the commercial packages) are available by downloading from the Internet.

A good starting point is the ARRL software repositories. To get to them, set your browser to the *ARRL Web* and go to the FTP

1 McConnell, Ken, Bodson, Dennis, and Urban, Steve, *FAX: Facsimile Technology and Systems*, 3rd Ed., Artech House, 1999,

2 Taggart, R.E., *Weather Satellite Handbook*, 5th Ed. (Newington: ARRL, 1994).

(files) link in the site index. You can use any commercial search site to look for "fax" AND "software." Examplesof several fax programs are as follows:

*JVFAX* is a very popular fax program. It is DOS-based program with a large number of options for installation. It can receive and transmit several fax formats, black-and-white and color. Your computer's serial port, connected to a very simple interface, provides the connection to your transceiver.

The *FAX 480* software program can also be used with fax as well as SSTV. For more information on this program and others including website addresses, see the July 1998 *QST* article "FAX 480 and SSTV Interfaces and Software" page 32. A copy for downloading of the free software program *vester_n.zip* for *FAX 480* can be found online at the Oakland University FTP site (see the **References** chapter). This program also uses a simple interface almost identical to that for *JVFAX*.

*Weatherman* is a DOS-based program, using a SoundBlaster (or compatible) card as the interface. The program is shareware and provides receive-only capability. A single, shielded wire from your receiver audio output to the computer audio input is the only connection needed.

*WXSat* operates under *Windows 3.X*. While specifically set up to decode and store weather-satellite APT pictures, it can also be used for HF-fax reception.

Both *Weatherman* and *WXSat* are samples of what you can find during a search on the Internet. Often, programs are offered and then either withdrawn or improved over the versions previously distributed—to get the latest and greatest you have to periodically search and see what comes up. If you use an online service such as CompuServe or AOL, they are another source of fax software. Check their ham forums or sections for listings.

Many commercial multimode controllers either contain software to receive and transmit fax, or are compatible with PC-hosted software. Available controller suppliers include MFJ, Timewave, and Kantronics; additional software may be required for the Kam Plus. Check the advertising pages of *QST* for the latest units available.

One well-known fax page on the Internet, complete with downloadable software, is posted and maintained by Marius Rensen; it contains listings of commercial fax transmissions for you to test your software or just SWL for interest. See the **References** chapter for the URL. Before using a program taken from any Internet source, check other sources for newer versions. It is not uncommon to have older versions posted on one place and newer versions in another. It is a good idea to virus check the software before and after unzipping.

Image transmission using voice bandwidth is a trade-off between resolution and time. In the section on slow-scan television, standards are described that permit 240-line black-and-white images to be transmitted in about 36 seconds while color images of similar resolution require anywhere from 72 to 188 seconds, depending on the color format. In terms of resolution, 240-line SSTV images are roughly equivalent to what you would obtain with a standard broadcast TV signal recorded on a home VCR. This is more than adequate for routine video communication, but there are many situations that demand images of higher resolution.

HAL Communications Corporation has developed an interesting system which enables a standard fax machine (Group 3 or G3) to send commercial fax images over HF radio. HAL Communications accomplishes this with just two small ancillary devices, which connect between a standard fax machine and an ordinary HF radio transceiver (see **Fig 12.38**). This method is frequently referred to as "G3 fax over radio." Any G3 fax machine can be connected to the HAL FAX-4100 controller with just a standard RJ-11 modular connector. The FAX-4100 controller connects directly to the HAL CLOVER-2000 (DSP-4100) radio data modem, which in turn connects to the HF transceiver. This entire setup is duplicated at the opposite end of the link.

A "call" is initiated from the fax machine keypad just as if the fax machine were connected to a phone line. The FAX-4100 controller includes a built-in 9600-baud G3 modem which emulates the telephone system: The controller at the initiating end answers the ring from the originating fax machine, establishes

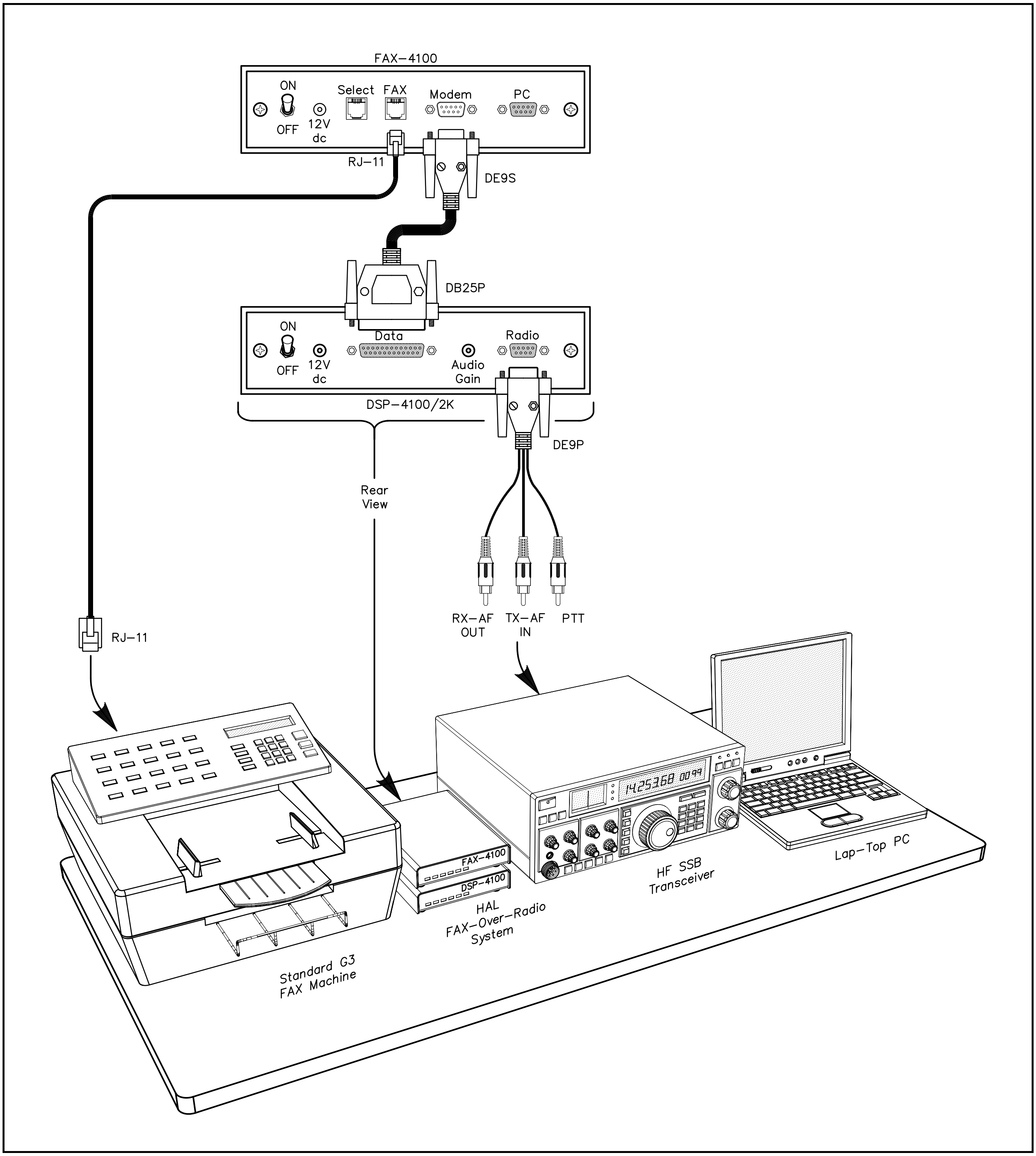



**Fig 12.38—Set up of a G3 fax machine connected to a HAL FAX-4100 controller and a HAL CLOVER-2000 (DSP-4100) radio data modem, which in turn connects to an HF transceiver.**

the HF radio link (based on the "phone number"), and handshakes with the controller at the other end to start the receiving faxmachine. Fax image data then passes from the fax machine into the controller's memory at the originating end. The controller also establishes a data link between the CLOVER-2000 modems at both ends, then passes the fax data through them and the controller at the receiving end, and finally into the receiving G3 fax machine. HAL has automated the HF radio operating procedures. To the user, sending a fax over HF radio is a simple three-step process:

1. Lay the page(s) on the fax machine.
2. Enter the ID number of the other station.
3. Push GO on the fax machine.

Housekeeping control functions and indications are also automated, feeding messages back to the fax machine whenever possible (link failed, other station not available, etc.). A full page can be sent in 2 to 6 minutes, depending upon ionospheric conditions and density of the page to be transmitted. The entire link set up and maintenance procedure is transparent to the fax operator, who need not know nor care that an HF radio system is part of the fax link. It all works just like a standard fax telephone transmission. An additional piece of equipment is available from HAL to enable the same fax machine to be shared between HF radio and conventional telephone lines. The HAL LI-4100 Line Interface is a "smart switch" that can be connected between the fax machine, the FAX-4100 controller, and up to two telephone lines.

## SLOW-SCAN TELEVISION (SSTV)

An ancient Chinese proverb states: “A picture is worth a thousand words.” It’s still true today. Sight is our highest bandwidth sense and the primary source of information about the world around us. What would you think about a TV news program without pictures about the stories? Would you enjoy reading the comics if there were no drawings with the text? Do you close your eyes when talking to someone in person? Many hams feel the same way about conversing with Amateur Radio: sending images is a wonderful way to enhance communication. This material was written by John Langner, WB2OSZ.

For decades only a dedicated few kept SSTV alive. The little commercial equipment was very expensive and home brewing was much too complicated for most people. Early attempts at computer-based systems were rather crude and frustrating to use.

The situation has changed dramatically in recent years. There is now a wide variety of commercial products and home-brew projects to fit every budget, and SSTV activity is experiencing rapid growth. There is even software that uses the popular Sound Blaster computer sound card for SSTV.

The early SSTV 8-second transmission standard is illustrated in **Fig 12.49**. Audio tones in the 1500 to 2300-Hz range represent black, white, and shades of gray. A short 1200-Hz burst separates the scan lines, and a longer 1200-Hz tone signals the beginning of a new picture.

### *Color SSTV Evolution*

The early experimenters weren’t content with only black and white (B&W) images and soon devised a clever way to send color pictures with B&W equipment. The transmitting station sends the same image three times, one each with red, green and blue filters in front of the TV camera lens. The receiving operator took three long-exposure photographs of the screen, placing red, green and blue filters in front of the film camera’s lens at the appropriate times. This was known as the “frame sequential” method.

In the 1970s, it became feasible to save these three images in solid-state memory and simultaneously display them on an ordinary color TV. But, the frame-sequential method had some drawbacks. As the first frame was received you’d see a red and black image. During the second frame, green and yellow would appear. Blue, white, and other colors wouldn’t show up until the final frame. Any noise (QRM or QRN) could ruin the image registration (the overlay of the frames) and spoil the picture.

The next step forward was the “line sequential” method. Each line is scanned 3 times: once each for the red, green, and blue picture components. Pictures could be seen in full color as they were received and registration problems were reduced. The Wraase SC-1 modes are examples of early line-sequential color transmission. They have a horizontal sync pulse for each of the color component scans. The major weakness here is that if the receiving end gets out of step, it won’t know which scan represents which color.

Rather than sending color images with the usual RGB (red, green, blue) components, Robot Research used luminance and chrominance signals for their 1200C modes. The first half or two thirds of each scan line contains the luminance information which is a weighted average of the R, G and B components. The remainder of each line contains the chrominance signals with the color information. Existing B&W equipment could display the B&W-compatible image on the first part of each scan line and the rest

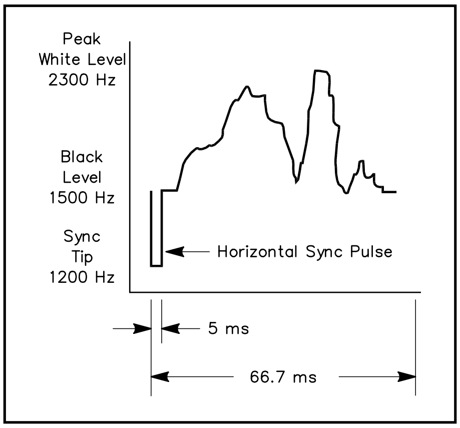



**Fig 12.49—Early SSTV operators developed a basic 8-second black and white transmission format. The sync pulses are often called “blacker than black.” A complete picture would have 120 lines (8 seconds at 15 ms per line). Horizontal sync pulses occur at the beginning of every line; a 30 ms vertical sync pulse precedes each frame.**

would go off the edge of the screen. This compatibility was very beneficial when most people still had only B&W equipment.

The luminance-chrominance encoding made more efficient use of the transmission time. A 120-line color image could be sent in 12 s, rather than the usual 24 s. Our eyes are more sensitive to details in changes of brightness than color, so the time could be used more efficiently by devoting more time to luminance than chrominance. The NTSC and PAL broadcast standards also take advantage of this vision characteristic and use less bandwidth for the color part of the signal.

The 1200C introduced another innovation: it encodes the transmission mode in the vertical sync signal. By using narrow FSK encoding around the sync frequency, compatibility was maintained. This new signal just looked like an extra-long vertical sync to older equipment. (See the sidebar "Examining Robot's Vertical-Interval-Signaling (VIS) Code" for more details.)

The luminance-chrominance encoding offers some benefits but image quality suffers. It is acceptable for most natural images but looks bad for sharp, high-contrast edges, which are more and more common as images are altered via computer graphics. As a result, all newer modes have returned to RGB encoding.

The Martin and Scottie modes are essentially the same except for the timings. They have a single horizontal sync pulse for each set of RGB scans. Therefore, the receiving end can easily get back in step if synchronization is temporarily lost. Although they have horizontal sync, some implementations ignore them on receive. Instead, they rely on very accurate time bases at the transmitting and receiving stations to keep in step. The advantage of this "synchronous" strategy is that missing or corrupted sync pulses won't disturb the received image. The disadvantage is that even slight timing inaccuracies produce slanted pictures.

In the late 1980s, yet another incompatible mode was introduced. The AVT mode is different from all the rest in that it has *no horizontal sync*. It relies on very accurate oscillators at the sending and receiving stations to maintain synchronization. If the beginning-of-frame sync is missed, it's all over. There is no way to determine where a scan line begins. However, it's much harder to miss the 5-s header than the 300-ms VIS code. Redundant information is encoded 32 times and a more powerful error-detection scheme is used. It's only necessary to receive a small part of the AVT header in order to achieve synchronization. After this, noise can wipe out parts of the image, but image alignment and colors remain correct. **Table 12.15** lists characteristics of common modes.

### Scan Converters

A scan converter is a device that converts signals from one TV standard to another. In this particular case we are interested in converting between SSTV, which can be sent through audio channels, and fast

#### Examining Robot's Vertical-Interval-Signaling (VIS) Code

The original 8-second black-and-white SSTV-image standard used a 30-millisecond, 1200-Hz pulse to signal the beginning of a new frame. In the Robot 1200C, Robot Research increased the vertical sync period by a factor of 10, encoded 8 bits of digital data into it and called it *vertical-interval signaling* (VIS). VIS is composed of a start bit, 7 data bits, an even parity bit, and a stop bit, each 30 milliseconds long. (See Fig A).

Since then, inventors of new SSTV modes (Martin, Scottie, AVT, etc) have adopted Robot's scheme and assigned codes to their particular mode that are unused by the Robot modes. So, each of the SSTV transmission modes has a unique VIS code. This allows new equipment to automatically select any of the new SSTV modes while maintaining compatibility with the older equipment.—*WB2OSZ*

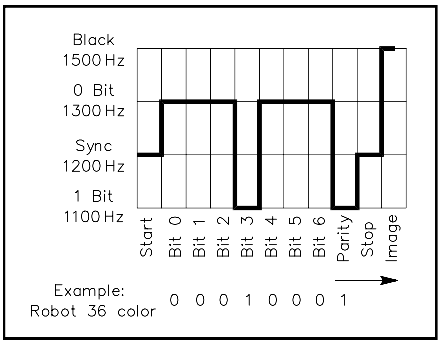



**Fig A—Composition of the vertical interval signaling (VIS) code.**

**Table 12.15**
**SSTV Transmission Characteristics**

| *Mode* | *Designator* | *Color Type* | *Scan Time (sec)* | *Scan Lines* | *Notes* |
|---|---|---|---|---|---|
| AVT | 24 | RGB | 24 | 120 | D |
| | 90 | RGB | 90 | 240 | D |
| | 94 | RGB | 94 | 200 | D |
| | 188 | RGB | 188 | 400 | D |
| | 125 | BW | 125 | 400 | D |
| Martin | M1 | RGB | 114 | 240 | B |
| | M2 | RGB | 58 | 240 | B |
| | M3 | RGB | 57 | 120 | C |
| | M4 | RGB | 29 | 120 | C |
| HQ | HQ1 | YC | 90 | 240 | G |
| | HQ2 | YC | 112 | 240 | G |
| Pasokon TV | P3 | RGB | 203 | 16+480 | |
| | P5 | RGB | 305 | 16+480 | |
| | P7 | RGB | 406 | 16+480 | |
| Robot | 8 | BW | 8 | 120 | A,E |
| | 12 | BW | 12 | 120 | E |
| | 24 | BW | 24 | 240 | E |
| | 36 | BW | 36 | 240 | E |
| | 12 | YC | 12 | 120 | |
| | 24 | YC | 24 | 120 | |
| | 36 | YC | 36 | 240 | |
| | 72 | YC | 72 | 240 | |
| Scottie | S1 | RGB | 110 | 240 | B |
| | S2 | RGB | 71 | 240 | B |
| | S3 | RGB | 55 | 120 | C |
| | S4 | RGB | 36 | 120 | C |
| | DX | RGB | 269 | 240 | B |
| Wraase SC-1 | 24 | RGB | 24 | 120 | C |
| | 48 | RGB | 48 | 240 | B |
| | 96 | RGB | 96 | 240 | B |
| Wraase SC-2 | 30 | RGB | 30 | 128 | |
| | 60 | RGB | 60 | 256 | |
| | 120 | RGB | 120 | 256 | |
| | 180 | RGB | 180 | 256 | |
| Pro-Skan | J120 | RGB | 120 | 240 | |
| WinPixPro | GVA 125 | BW | 125 | 480 | |
| | GVA 125 | RGB | 125 | 240 | |
| | GVA 250 | RGB | 250 | 480 | |
| JV Fax | JV Fax Color | RGB | variable | variable | F |
| FAX480 | Fax 480 | BW | 138 | 480 | |
| | Truscan | BW | 128 | 480 | H |
| | Colorfax 480 | RGB | 384 | 480 | I |

***Notes***

RGB—Red, green and blue components sent separately.
YC—Sent as Luminance (Y) and Chrominance (R-Y and B-Y).
BW—Black and white.
A—Similar to original 8-second black & white standard.
B—Top 16 lines are gray scale. 240 usable lines.
C—Top 8 lines are gray scale. 120 usable lines.
D—AVT modes have a 5-second digital header and no horizontal sync.
E—Robot 1200C doesn't really have B&W mode but it can send red, green or blue memory separately. Traditionally, just the green component is sent for a rough approximation of a b&w image.
F—JV Fax Color mode allows the user to set the number of lines sent, the maximum horizontal resolution is slightly less than 640 pixels. This produces a slow but very high resolution picture. SVGA graphics are required.
G—Available only on Martin 4.6 chipset in Robot 1200C.
H—Vester version of FAX480 (with VIS instead of start signal and phasing lines).
I—Trucolor version of Vester Truscan.

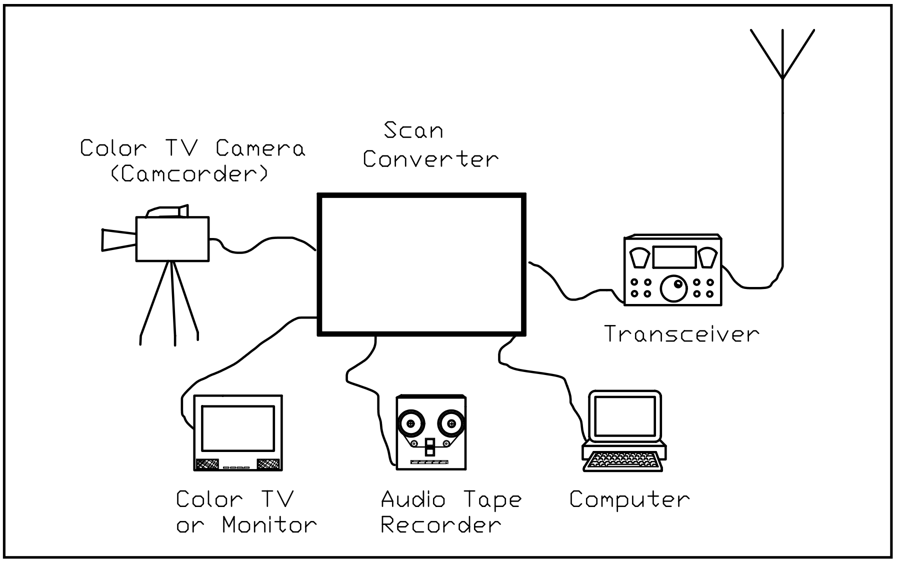



**Fig 12.50—Diagram of an SSTV station based on a scan converter.**

### Courteous SSTV Operating

- Recommended frequencies: 3.845, 7.171, 14.230, 14.233, 21.340, 28.680, 145.5 MHz.
- 14.230 is the most active
- Make contact by voice before sending SSTV.
- Not all systems recognize the VIS code, so it is good manners to announce the mode before transmitting.

scan (broadcast or ATV), so we can use ordinary camcorders and color televisions to generate and display pictures. From about 1985 to 1992, the Robot 1200C was king.

**Fig 12.50** shows a typical SSTV station built around a scan converter such as the Robot 1200C or a SUPERSCAN 2001. The scan converter has circuitry to accept a TV signal from a camera and store it in memory. It also generates a display signal for an ordinary television set. The interface to the radio is simply audio in, audio out and a push-to-talk (PTT) line. In the early days, pictures were stored on audio tape, but now computers store them on disks. Once a picture is in a computer, it can be enhanced with paint programs.

This is the easiest approach. Just plug in the cables, turn on the power and it works. Many people prefer special dedicated hardware, but most of the recent growth of SSTV has been from these lower cost PC-based systems.

## SSTV with a Computer

There were many attempts to use early home computers for SSTV. Those efforts were hampered by very small computer memories, poor graphics capabilities and poor software development tools.

Surprisingly, little was available for the ubiquitous IBM PC until around 1992, when several systems appeared in quick succession. By this time, all new computers had a VGA display, which is required for this application. Most new SSTV stations look like **Fig 12.51**. Some sort of interface is used to get audio in and out of the computer. These can be external interfaces connected to a serial or printer port, an internal card specifically for SSTV or even a peripheral audio card. IBM-type PC compatible computers with VGA video display monitors can also be used with their existing SoundBlaster-compatible sound boards for the interfaces, if software such as WinSkan and WinPix Pro are used. Most of the work is done in software. System updates are performed by reading a floppy disk instead of changing EPROMs or other components. Most of these software programs are based on the work of Ben Vester, K3BC. These computer programs include Vester Truscan, Pasokon TV Lite, ProScan, JVFAX, and HamComm; they all use a simple "clipper" hardware interface, which can easily be built with less than $15 worth of RadioShack parts, because the computer program does all of the processing work previously done by more expensive hardware. See the July 1998 issue of *QST* for "FAX 480 and SSTV Interfaces and Software" on page 32. The URL for downloading Vester's software is **ftp://oak.oakland.edu/pub/hamradio/arrl/bbs/programs**.

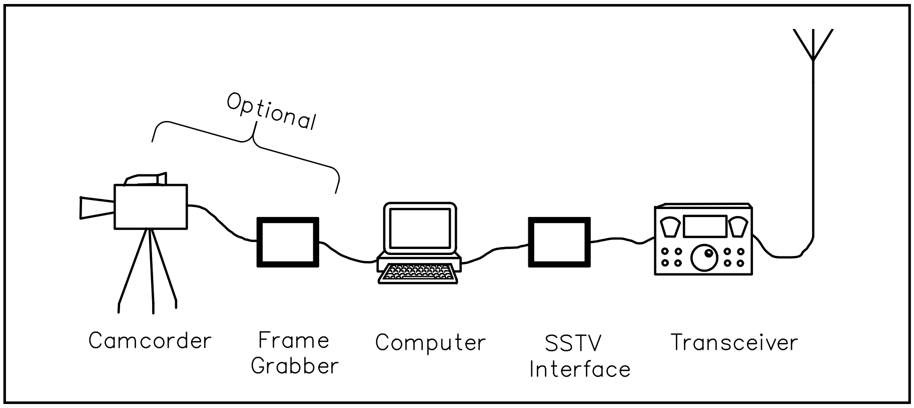



**Fig 12.51—A modern, PC-based SSTV station.**

### *How It Works*

Transmitting SSTV images with a computer is quite simple. All you need to do is generate fairly accurate tones and change them at the proper pixel rate. Tones in the range of 1500 to 2300 Hz correspond to the pixel intensities, and most modes use 1200-Hz sync pulses. A very low-cost system could even use the computer's built-in tone generator for transmitting, but the tones must be pure with little distortion in order to produce an acceptable RF signal via AFSK (see "AFSK" under Baudot section of this chapter).

SSTV reception is a little more difficult. First you must somehow measure the frequency of the incoming tone. You can't simply count the number of cycles in a second, or even 0.01 second, because the frequency is changing thousands of times each second. **Fig 12.52** illustrates one way of rapidly measuring the incoming tone's frequency. Two filters are designed to have maximum outputs a little beyond the ends of the frequency range of interest. The output of one filter is rectified to become a positive voltage; the output of the other is rectified to become a negative voltage; then the voltages are summed. A low-pass filter, with a 1-kHz cutoff, removes the audio carrier ripple while passing the slower video signal. With careful design, the result is a voltage that is fairly proportional to the input frequency. Finally, an analog to digital (A/D) converter processes the signal for the computer.

Another frequency-measuring approach uses digital circuitry to measure the period of each audio cycle (see **Fig 12.53**). When the signal amplitude crosses zero, a counter is reset. It then proceeds to count

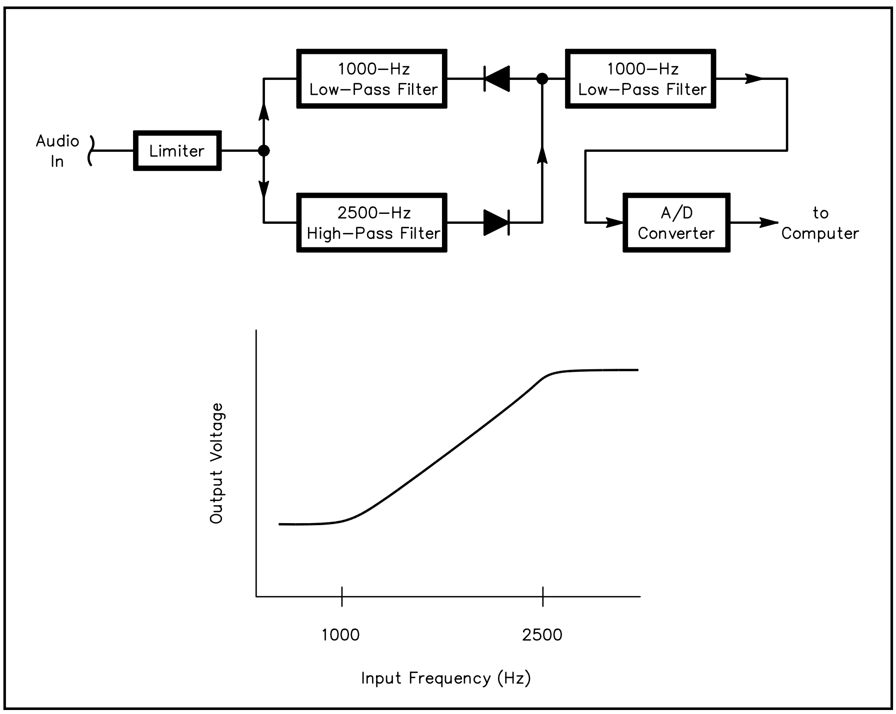



**Fig 12.52—Block diagram of an analog SSTV demodulator.**

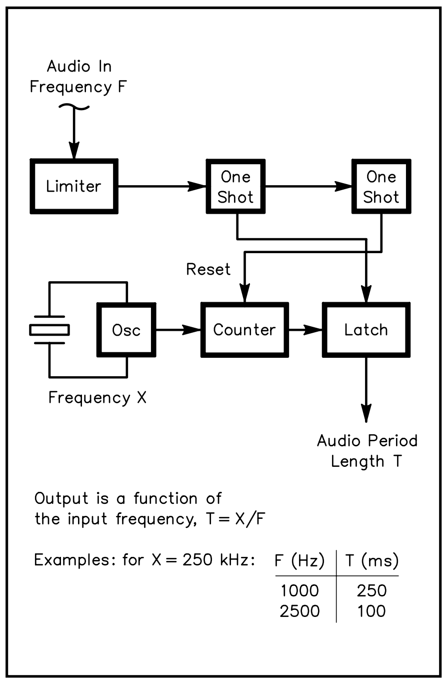



**Fig 12.53—Block diagram of a digital SSTV demodulator.**

## SSTV Glossary

**ATV**—Amateur Television. Sending pictures by Amateur Radio. You'd expect this abbreviation to apply equally to fast-scan television (FSTV), slow-scan television (SSTV) and facsimile (fax), but it's generally applied only to FSTV.

**AVT**—Amiga Video Transceiver. 1) Interface and software for use with an Amiga computer, developed by Ben Blish-Williams, AA7AS, and manufactured by Advanced Electronic Applications (AEA); 2) a family of transmission modes first introduced with the AVT product.

**Back porch**—The blank part of a scan line immediately following the horizontal sync pulse.

**Chrominance**—The color component of a video signal. NTSC and PAL transmit color images as a black-and-white compatible luminance signal along with a color subcarrier. The subcarrier phase represents the hue and the subcarrier's amplitude is the saturation. Robot color modes transmit pixel values as luminance (Y) and chrominance (R-Y [red minus luminance] and B-Y [blue minus luminance]) rather than RGB (red, green, blue).

**Demodulator**—For SSTV, a device that extracts image and sync information from an audio signal.

**Field**—Collection of top to bottom scan lines. When interlaced, a field does not contain adjacent scan lines and there is more than one field per frame.

**Frame**—One complete scanned image. The Robot 36-second color mode has 240 lines per frame. NTSC has 525 lines per frame with about 483 usable after subtracting vertical sync and a few lines at the top containing various information.

**Frame Sequential**—A method of color SSTV transmission which sent complete, sequential frames of red, then green and blue. Now obsolete.

**Front porch**—The blank part of a scan line just before the horizontal sync.

**FSTV**—Fast-Scan TV. Same as common, full-color, motion commercial broadcast TV.

**Interlace**—Scan line ordering other than the usual sequential top to bottom. For example, NTSC sends a field with just the even lines in $^1/_{60}$ second, then a field with just the odd lines in $^1/_{60}$ second. This results in a complete frame 30 times a second. AVT "QRM" mode is the only SSTV mode that uses interlacing.

**Line Sequential**—A method of color SSTV transmission that sends red, green, and blue information for *each sequential scan line.* This approach allows full-color images to be viewed during reception.

**Luminance**—The brightness component of a video signal. Usually computed as Y (the luminance signal) = 0.59 G (green) + 0.30 R (red) + 0.11 B (blue).

**Martin**—A family of amateur SSTV transmission modes developed by Martin Emmerson, G3OQD, in England.

**NTSC**—National Television System Committee. Television standard used in North America and Japan.

**PAL**—Phase alteration line. Television standard used in Germany and many other parts of Europe.

**Pixel**—Picture element. The dots that make up images on a computer's monitor.

**P7 monitor**—SSTV display using a CRT having a very-long-persistence phosphor.

**RGB**—Red, Green, Blue. One of the models used to represent colors. Due to the characteristics of the human eye, most colors can be simulated by various blends of red, green, and blue light.

**Robot**—(1) Abbreviation for Robot 1200C scan converter; (2) a family of SSTV transmission modes introduced with the 1200C.

**Scan converter**—A device that converts one TV standard to another. For example, the Robot 1200C converts SSTV to and from FSTV.

**Scottie**—A family of amateur SSTV transmission modes developed by Eddie Murphy, GM3SBC, in Scotland.

**SECAM**—Sequential color and memory. Television standard used in France and the Commonwealth of Independent States.

**SSTV**—Slow Scan Television. Sending still images by means of audio tones on the MF/HF bands using transmission times of a few seconds to a few minutes.

**Sync**—That part of a TV signal that indicates the beginning of a frame (vertical sync) or the beginning of a scan line (horizontal sync).

**VIS**—Vertical Interval Signaling. Digital encoding of the transmission mode in the vertical sync portion of an SSTV image. This allows the receiver of a picture to automatically select the proper mode. This was introduced as part of the Robot modes and is now used by all SSTV software designers.

**Wraase**—A family of amateur SSTV transmission modes first introduced with the Wraase SC-1 scan converter developed by Volker Wraase, DL2RZ, of Wraase Electronik, Germany.

pulses from a crystal controlled oscillator. At the end of the audio cycle, the counter content is snatched, the counter is reset and the process starts all over again.

The digital approach offers a few advantages over the analog approach. A single chip can contain the counter and handle several other functions as well. The analog approach requires a handful of op amps, resistors, capacitors, diodes and an analog to digital (A/D) converter. The digital approach has crystal controlled accuracy and no adjustments are required. The frequency-to-voltage transfer function of the analog version isn't exactly linear and can change with temperature, power-supply variations and component aging.

Digital Signal Processing (DSP) is an exciting possibility for SSTV demodulators. With DSP, a high speed A/D converter is used to sample the audio input. After that, it's all software. DSP can be used to construct filters that are more flexible, accurate, stable and reproducible than their analog counterparts.

Once you have the tone-frequency information, the real work begins. The next step is to separate the composite signal into the sync and video components. To reduce the effects of noise, the sync pulses are cleaned up with a low pass filter and Schmitt trigger. Then, sync is used to control the timing of pixel sampling. **Fig 12.54** contains a high level outline of a program used to receive an 8-s B&W picture. Receiving colors isn't much more difficult. For nonRobot modes, gather the R, G, and B scans for each line, combine them and display a line in color. Robot modes require considerably more calculation to undo their encoding.

## AN INEXPENSIVE SSTV SYSTEM

Here is a color SSTV/FAX480/weatherfax (**Fig 12.55**) system for IBM PCs and compatibles that is essentially 99% software! (It most recently appeared in July 1998 *QST,* pp 23-26.) And this system *transmits,* too! The software is available from *ARRLWeb*. See page viii.

Ben Vester's, K3BC, work is aimed at the experimentally inclined, so if you're not familiar with BASIC programming, be prepared to learn a little about it if you want to maximize the utility of this system.

### Hardware

**Fig 12.56** shows a simple circuit used for receiving and transmitting. Connect the output of T2 to the phone patch input (often labeled LINE INPUT) of your transceiver. If you already have a phone patch, you can eliminate T2, and connect the line directly to the patch's phone-line terminals. Nearly all patches employ transformer isolation, but a simple ohmmeter check will verify that is true of *your* patch. (Avoid using the transceiver's mike input because of possible RF feedback problems.) R3 is set to the proper level for the audio going to the transmitter. SSTV has a 100% duty cycle signal, so you must set the audio signal to the transceiver at a level it can handle without overheating.

There is no low-pass filtering in the audio line between the computer output and transmitter audio input. On-the-air checks with many stations reveal that no additional external filtering is required when using SSB transmitters equipped with mechanical or crystal filters. If you intend to use this circuit with an AM or phasing-type

```
Set line number, L, to 1
Repeat:
  Wait for sync
  Wait for end of sync
  If it was vertical sync, set L = 1

  Gather 128 pixels
  Display pixels on line L
  Increment L
  If L > 120, set L = 1
```

**Fig 12.54—An outline of typical software written to display an SSTV frame from received digital picture information.**


**Fig 12.55—An example SSTV image.**

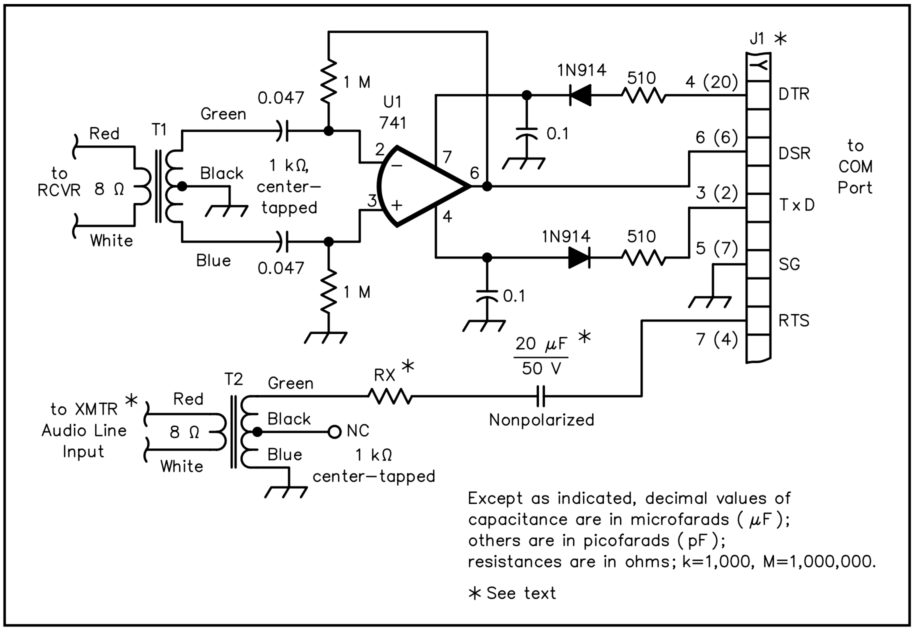



**Fig 12.56—Schematic of the simple SSTV receive and transmit circuit. This circuit appears on page 34 of July 1998 *QST*. T1 and T2 are RadioShack 273-1380 audio-output transformers; the 20-µF, 50-V capacitor is a parallel combination of two RadioShack 272-999 10-µF, 50-V nonpolarized capacitors; equivalent parts can be substituted. Unless otherwise specified, resistors are 1/4-W, 5%-tolerance carbon composition or film units. An optional low-pass filter can be used between the output of the computer and the transmitter's audio-input line (see text). At J1, numbers in parentheses are for 25-pin connectors; other numbers are for 9-pin connectors.**

SSB rig (or with VHF/UHF FM transmitters), add audio filtering to provide the required spectral purity. An elliptical low-pass filter such as described by Campbell[1] should be adequate for most cases.

Circuit component values aren't critical nor is the circuit's physical construction. *Do* use a socket for the IC. A PC board is available from FAR Circuits,[2] but perf-board construction employing short leads works fine.

## The Computer

The most important piece of hardware is the computer, which should have an 80286 (or better) microprocessor; a '386 machine running at 16 or 33 MHz definitely gives better results. You need a VGA color monitor that can provide a 640×480, 256-color noninterlaced display and a VGA (usually identified as SVGA) video adapter card that offers a 640×480×256-color mode.[3] The software directly addresses six of the most common SVGA chip types and also includes a VESA standard choice. If your video adapter card doesn't match one of the six, you'll need a VESA driver for your specific card. If you have trouble finding a driver, try checking on the Internet.

## Software

*GWBASIC* is the programming tool. Although the guts of the program are contained in assembly language code (.ASM files), this code is available to the program (and you) through BASIC. All of the modifications to the core programs (.ASM files) that adapt them to the multitude of SSTV/FAX modes are accomplished using BASIC POKEs. This allows experimenters with even a limited knowledge of BASIC programming to make modifications that add other modes, and so on. In deference to a few friends who complained about learning any BASIC, the programs include a system configuration list.The program uses this list to determine which POKEs to make. This system is strictly keyboard controlled. The software uses a unique technique to get wider color definition than is normally available with a 256-color video card.

1 R. Campbell, "High-Performance, Single-Signal Direct-Conversion Receivers," *QST,* Jan 1993, pp 32-40. See also Feedback, *QST,* Apr 1993, p 75.

2 FAR Circuits (see Address List in **References** chapter). The PC board is $4.50, plus $1.50 shipping.

3 Picture quality is degraded with an interlaced display.

## Commercial SSTV Products

All software and computer interfaces are for IBM PC with VGA display unless otherwise noted. Contact information for each of these sources appears in the Address List in the **References** chapter.

### Scan Converters

***DFM 1200 USA***—PC Boards and instructions to build a Robot 1200C clone. The builder must collect EPROM and other parts. Muneki also supplies some of the hard-to-find parts.

Donald P. Lucarell, K8SQL
Felipe Rojas
Muneki Yamafuzi, JF3GOH

***SUPERSCAN 2001***—Similar to 1200C but with many new features such as: four image memories, built-in mouse interface, on-screen help messages, and battery back up of CMOS memory to save system parameters when power is turned off. EPROMs developed by Martin Emmerson. Available assembled and in various semi-kit options.

Jad Bashour

***Tasco (TSC-70U)***—A stand-alone color slow-scan TV converter. Receives and transmits color slow-scan without a PC. For picture storage, a PC interface is an optional module.

***Replacement EPROMs***—A brand new 1200C was capable of only the "Robot" modes. Martin supplies replacement EPROMs which add the Martin, Scottie, AVT, Wraase, and fax modes and other interesting features such as an "oscilloscope" tuning indicator. Product Reviews: Jul 1991, *73 Amateur Radio Today,* p 46 (version 4.0); *IVCA Newsletter*, Fall 1991 (version 4.1); *IVCA Newsletter* Spring 1993 (version 4.2)

Martin H. Emmerson MSc, G3OQD

### Computer-based SSTV Systems

***BMK-MULTY***—Software for transmitting and receiving AMTOR, RTTY, CW, PACTOR, Audio Spectrum Analyzer, HF WEFAX, and SSTV.

Schnedler Systems, AC4IW

***MFJ-1278B***—MCP for packet radio, RTTY, AMTOR, CW and so on. It is also capable of sending and receiving most popular SSTV modes with the MultiCom software.

MFJ Enterprises Inc

***Pasokon TV***—Interface to send and receive SSTV fits inside expansion slot of computer. Software supports all popular modes, automatic receive mode selection from VIS code, up to 32k simultaneous colors on screen, graphical user interface with mouse support. Article: Jan 1993, *QST*, p 20. A free demo version, called *EZSSTV*, is available in many of the ham radio software depositories.

Absolute Value Systems

***PC SSTV 5***—Compact separate send and receive interfaces plug into a serial port. Software supports the most popular modes, reads/writes popular image file formats, built-in text generating capability.

Software Systems Consulting

***Slow Scan II***—Software to send and receive SSTV using the popular Sound Blaster (or compatible) sound card instead of interface dedicated to SSTV. Details: May 1993, *QEX*. A free demo version of the software is available on CompuServe: Go HAMNET, Library 6, search for "SSTV", "SLOWSC.ZIP".

Harlan Technologies

***SSTV Explorer***—Low cost, receive-only system for most popular modes. Compact interface plugs into serial port. Has graphical user interface with mouse support, automatic receive mode selection, super VGA support with up to 32768 colors. Product Review: April 1994, *QST*, p 80.

Radioware

***Viewport VGA***—External interface to send and receive plugs into printer port. Software (shareware by KA2PYJ) supports most popular modes. Construction article: *73*, Aug 1992.

A&A Engineering

**Accessories and Related Software**

***ART (Amiga Robot Terminal)***—Hardware interface and software to control Robot 1200C from Amiga computer. Contains paint program, multifont text, and many image processing functions. Supports Martin and Scottie EPROMs.
Thomas M. Hibben, KB9MC

***Audio Analyzer***—Software for use with the Sound Blaster. Produces frequency vs time plots of audio signals. Useful for studying SSTVsignals.
Harlan Technologies

***DFM SSTV Bandpass Filter***—A bandpass filter especially designed for SSTV.
Donald Lucarell

***GEST***—"all-in-one" SSTV utility package for the Robot 1200C. Includes paint program, text generation, special effects and image processing tools. Graphical user interface supports CGA, EGA, VGA and mouse. Controls the 1200C through parallel port.
Torontel
Royal Electronics (Canada)

***HiRes***—Paint program for use with the Robot 1200C. Has many impressive special effects and character fonts.
Tom Jenkins, N9AMR

***HiRes 32***—New version of HiRes designed specifically for use with PC-based SSTV systems. Requires VGA display adapter capable of 32768 simultaneous colors.
Tom Jenkins, N9AMR

***Robot Helper***—Robot 1200C control program for Microsoft Windows and OS/2 environments. Some features include: thumbnail previews of images on disk, dual image preview windows, fast image load and save to Robot, support for Robot or Martin EPROMs.
William Montgomery, VE3EC

***SCAN***—Software for use with Robot 1200C.
Bert Beyt, W5ZR

### *Some Program Details*

One of the common SSTV practices is to retransmit a picture you just received so other SSTVers not copying the originating station can see the image. This capability is included.

*RT.BAS* is the receive and retransmit program. On receive, you simply choose the mode from a menu, and wait for the picture transmission to complete. As of this writing, Robot 36 and 72 modes are available in either a synchronous or a line-synced mode. Other modes (all synchronous) are Scottie 1 and 2, Martin 1 and 2, AVT90, AVT94, Wraase 96, FAX480 and weatherfax.

When receiving, if you fail to get the mode selection made in time to catch the frame sync, you can go directly to copying by pressing the keyboard's spacebar. On all but the AVT modes, the next line sync is picked up and starts the picture. The AVT modes copy out of sync. Because the program allows you to scroll horizontally across the RGB color frames, you can resync after the picture has been received. A few images have nonstandard color registration, so the program can adjust color registration after the picture is received. You also can save the picture—usually after you have scrolled the picture so the CRT screen frames just the part you want to keep.

*TX.BAS* is used for transmitting any picture file. When queried, you provide the mode and the file name, and after a brief pause while the picture loads, press **G**(o) to transmit. To avoid additional switching complexity, VOX transmitter switching is used.

*VU.BAS* allows you to view a picture. It has the same adjustments available as *RT.BAS*. One feature (applicable only to the Robot modes) is the ability to "retune" the picture (in 10-Hz increments) as you view its color balance.

*SLIDESHO.BAS* gives you the vehicle to display a bunch of pictures as a slide show. Place *SLIDESHO.BAS* in a directory contained in your PATH statement so it can be called up from anywhere.

*TIFCONV.BAS* converts 640×480, 24-bit color, TIFF pictures into a format that can be transmitted by any of the supported SSTV modes except Robot. TIFF is a common format used to transfer higher-resolution pictures between programs. This program works with the Computer Eyes/RT[4] and Software Systems Consulting[5] frame grabbers. The picture output from this program can be viewed with *VU.BAS* and, of course, is bound by 320×240 with 18-bit color.

*LABEL.BAS* allows you to add call signs and other text to the SSTV pictures. It takes any black-and-white TIFF (that is, 1-bit) file and creates a mask cutout where the black is. You can superimpose the cutout over an SSTV picture either in any color you want, or transfer a cutout of any background file you find interesting. The letters will then look like they were cut out of the background picture. Obviously, you can use squares or circles in addition to fonts to transfer a piece of one file onto another one. Use a cheap hand scanner to capture interesting fonts you find. You can get a three-dimensional effect by painting a color through the mask, then moving the mask a few pixels and rerunning the data through *LABEL* with a background file or another color. Or, run several different masks through *LABEL* in sequence to obtain different colors or patterns on different letters.

Work with Ben Vester's system continues. Look at articles by Vester in the SSTV Bibliography and watch *QST* for more discussion.

## SUMMARY

For decades there was a convenient excuse for not trying SSTV: it cost kilobucks to buy a specialized piece of equipment. But you can't use that excuse anymore. There are several free programs that only require trivial interfaces to receive pictures. Once you get hooked, there are plenty of other home-brew projects and commercial products available at affordable prices. You need not be a computer wizard to install and use these systems.

SSTV is a rapidly changing area of Amateur Radio. Although it is still supported, the once-popular Robot 1200C has been discontinued. Many new products have been introduced.

### SSTV Bibliography


Abrams, C., and R. Taggart, "Color Computer SSTV," *73*, Nov and Dec 1984.


[4] ComputerEyes R/T by Digital Vision (see Address List in **References** chapter).

[5] Software Systems Consulting (see Address List in **References** chapter).

### For More Information

Contact information for each of these sources appears in the Address List in the **References** chapter.

**Weekly nets:**

Saturdays, 1500 UTC 14.230 MHz.
Saturdays, 1800 UTC 14.230 MHz.

**SSTV Newsletter:**

*VISION* from International Visual Communications Association (IVCA)

**Magazines specializing in ATV:**

*Amateur Television Quarterly*
*ATV Today!*
*CQ-TV* from British Amateur Television Club
*The SPEC-COM Journal*

**Old A5 magazine reprints:**

ESF copy services

**Handbook:**

*Slow Scan Television Explained*, by Mike Wooding, G6IQM (available from British Amateur Television Club and *Amateur Television Quarterly*).

Battles, B. and S. Ford, “Smile—You’re on Ham Radio!” *QST*, Oct 1992, p 42.

Bodson, D., W4PWF, and Karty, S., N5SK, “FAX480 and SSTV Interfaces and Software,” *QST*, Jul 1998, pp 32-36.

Cameroni, G., I2CAB, and G. Morellato, I2AED (translated by Jim Grubbs, K9EI), “Get on SSTV—with the C-64,” *Ham Radio*, Oct 1986, p 43.

Churchfield, T., K3HKR, “Amiga AVT System,” *73 Amateur Radio*, Jul 1989, p 29.

Goodman, D., WA3USG, “SSTV with the Robot 1200C Scan Converter and the Martin Emmerson EPROM Version 4.0,” *73 Amateur Radio Today*, Jul 1991, p 46.

Langner, J., WB2OSZ, “Color SSTV for the Atari ST,” *73 Amateur Radio*, Dec 1989, p 38; Jan 1990, p 41.

Langner, J., WB2OSZ, “SSTV—The AVT System Secrets Revealed,” *CQ-TV* 149 (Feb 1990), p 79.

Langner, J. WB2OSZ, “Slow Scan Television—It isn’t expensive anymore,” *QST*, Jan 1993, p 20.

Montalbano, J., KA2PYJ, “The ViewPort VGA Color SSTV System,” *73*, Aug 1992, p 8.

Schick, M., KA4IWG, “Color SSTV and the Atari Computer,” *QST*, Aug 1985, p 13.

Taggart, R., WB8DQT, “The Romscanner,” *QST*, Mar 1986, p 21.

Taggart, R., WB8DQT, “A New Standard for Amateur Radio Facsimile,” *QST*, Feb 1993, p 31.

Vester, B., K3BC "Vester SSTV/FAX80/Fax System Upgrades," Technical Correspondence, *QST*, Jun 1994, pp 77-78.

Vester, B., K3BC, "SSTV: An Inexpensive System Continues to Grow," Dec 1994 *QST*, pp 22-24.

Vester, B., K3BC, “K3BC’s SSTV Becomes TRUSCAN,” Technical Correspondence, *QST*, Jul 1996, pp66-67.

## FAST-SCAN TELEVISION

Fast-scan amateur television (FSTV or just ATV) is a wide-band mode that uses standard broadcast, or NTSC, television scan rates. It is called "fast scan" only to differentiate it from slow-scan TV. In fact, no scan conversions or encoder/decoders are necessary with FSTV. Any standard TV set can display the amateur video and audio. Standard (1 V P-P into 75 Ω) composite video from home camcorders, cameras, VCRs or computers is fed directly into an AM ATV transmitter. The audio has a separate connector and goes through a 4.5 MHz FM subcarrier generator which is mixed with the video. This section was written by Tom O'Hara, W6ORG.

Amateurs regularly show themselves in the shack, zoom in on projects, show home video tapes, computer programs and just about anything that can be shown live or by tape (see **Figs 12.57** and **12.58**). Whatever the camera "sees" and "hears" is faithfully transmitted, including color and sound information. Picture quality is about equivalent to that of a VCR, depending on video signal level and any interfering carriers. All of the sync and signal-composition information is present in the composite-video output of modern cameras and camcorders. Most camcorders have an accessory cable or jacks that provide separate video and audio outputs. Audio output may vary from one camera to the next, but usually it has been amplified from the built-in microphone to between 0.1 to 1 V P-P (into a 10-kΩ load).

ATV transmitters have been carried by helium balloons to above 100,000 ft, to the edge of space. The result is fantastic video transmissions, showing the curvature of the Earth, that have been received as far as 500 miles from the balloon. Small cameras have been put into the cockpits of R/C model airplanes to transmit a pilot's-eye view. Many ATV repeaters retransmit Space Shuttle video and audio from NASA during missions. This is especially exciting for schools involved with SAREX. ATV is used for public service events such as parades, races, Civil Air Patrol searches and remote damage assessment.

Emergency service coordinators have found that live video from a site gives a better understanding of a situation than is possible from voice descriptions alone. Weather-radar video, WEFAX, or other computer generated video has also been carried by ATV transmitters for RACES groups during significant storms. This use enables better allocation of resources by presenting real-time information about the storm track. Computer graphics and video special effects are often transmitted to dazzle the viewers.

### *How Far Does ATV Go?*

The theoretical snow-free line-of-sight distance for 10 W, given 15.8-dBd antennas and 2-dB feed-line loss at both ends, is 91 miles. (See **Table 12.16**.) However, except for temperature-inversion skip conditions, reflections, or through high hilltop repeaters, direct line-of-sight ATV contacts seldom exceed 25 miles. The RF horizon over flat terrain with a 50-ft tower is 10 miles. For best DX use low loss feed line and a broadband high-gain antenna, up as high as possible. The antenna system is the most important part of an ATV system because it affects both receive and transmit signal strength.

A snow-free, or "P5," picture rating (see **Fig 12.59**) requires at least 200 µV (–61 dBm) of signal at the input of the ATV receiver, depending

**Fig 12.57—Students enjoy using ATV to communicate between science and computer classes.**

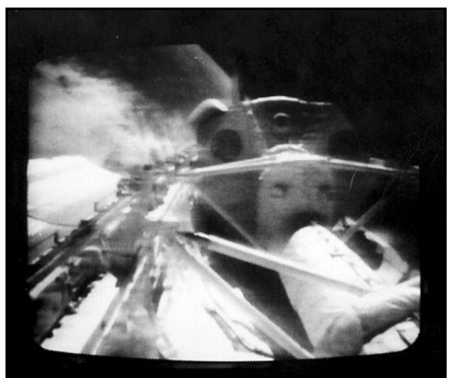


**Fig 12.58—The ATV view shows the aft end of the Space Shuttle cargo bay during mission STS-9.**

**Table 12.16**

**Line-of-Sight Snow-Free 70-cm ATV Communication Distances**

This table relates transmit and receive station antenna gains to communication distances in miles for 1/10/100 W PEP at 440 MHz. To find the possible snow-free distance under line-of-sight conditions, select the column that corresponds to transmit antenna gain and the row for the receive antenna gain. Read the distance where the row and column intersect. Multiply the result by 0.5 for 902 MHz and 0.33 for 1240 MHz. The table assumes 2 dB of feed-line loss, a 3 dB system noise figure at both ends and snow-free is greater than 40 dB picture:noise ratio (most home cameras give 40 to 45 dB picture:noise; this is used as the limiting factor to define snow-free ATV pictures). The P unit picture rating system goes down about 6 dB per unit. For instance, P4 pictures would be possible at double the distances in the table.

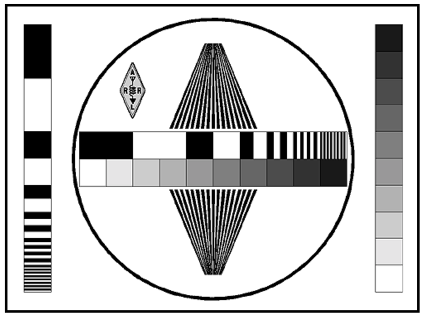
| | *TX Antenna* | | | |
|---|---|---|---|---|
| | *0 dBd* | *4 dBd* | *9 dBd* | *15.8 dBd* |
| ***RX Antenna*** | | | | |
| *0 dBd* | 0.8/2.5/8 | 1/3.5/11 | 2/7/22 | 5/15/47 |
| *4 dBd* | 1/3.5/11 | 2/6/19 | 3.5/11/34 | 7.5/23/75 |
| *9 dBd* | 2/7/22 | 3.5/11/34 | 6/19/60 | 13/42/130 |
| *15.8 dBd* | 5/15/47 | 7.5/23/75 | 13/42/130 | 29/91/290 |

on the system noise figure and bandwidth. The noise floor increases with bandwidth. Once the receiver system gain and noise figure reaches this floor, no additional gain will increase sensitivity. At 3-MHz bandwidth the noise floor is 0.8 µV (–109 dBm) at standard temperature. If you compare this to an FM voice receiver with 15 kHz bandwidth; there is a 23 dB difference in the noise floor. However the eye, much like the ear of experienced CW operators, can pick out sync bars in the noise below the noise floor. Sync lock and large well contrasted objects or lettering can be seen between 1 and 2 µV. Color and subcarrier sound come out of the noise between 2 and 8 µV depending on their injection level at the transmitter and TV-set differences.

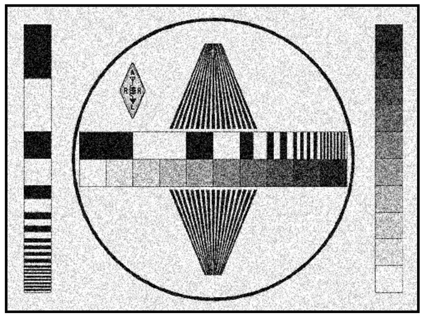
Two-meter FM is used to coordinate ATV contacts. Operators must take turns transmitting on the few available channels and the 2-m link allows full-duplex audio from many receiving stations to the ATV transmitting station, who is speaking on the sound subcarrier. This is great for interactive show and tell. Also it is much easier to monitor a squelched 2-m channel using an omni antenna rather than searching out each station with a beam. Depending on the third-harmonic relationship to the video on 70 cm, 144.34 MHz and 146.43 MHz (simplex) are the most popular frequencies; they are often mixed with the subcarrier sound on ATV repeater outputs.

### *Getting the Picture*

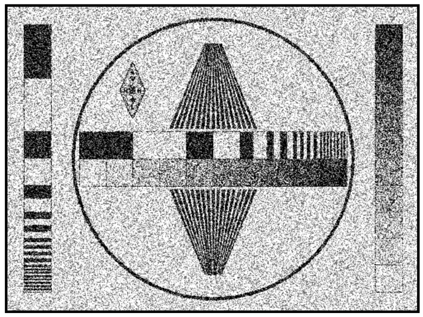
Since the 70-cm band corresponds to cable channels 57 through 61, seeing your first ATV picture may be as simple as connecting a good



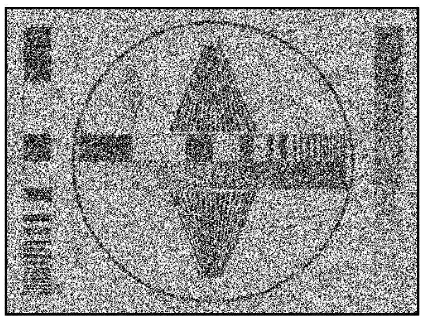
**P5 —Excellent**



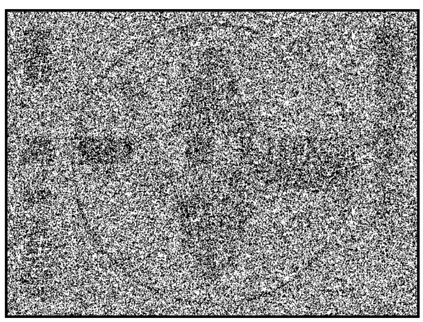
**P4—Good**



**P3—Fair**



**P2—Poor**



**P1—Barely perceptible**

**Fig 12.59—An ATV quality reporting system.**

outside 70-cm antenna (aligned for the customary local polarization) to a cable-ready TV set's antenna input jack. Cable channel 57 is 421.25 MHz, and each channel is progressively 6 MHz higher. (Note that cable and broadcast UHF channel frequencies are different.) Check the *ARRL Repeater Directory* for a local ATV repeater output that falls on one of these cable channels. Cable-ready TVs may not be as sensitive as a low-noise downconverter designed just for ATV, but this technique is well worth a try.

Most stations use a variable tuned downconverter specifically designed to convert the whole amateur band down to a VHF TV channel. Generally the 400 and 900 MHz bands are converted to TV channel 3 or 4, whichever is not used in the area. For 1200 MHz converters, channels 7 through 10 are used to get more image rejection. The downconverter consists of a low-noise preamp, mixer and tunable or crystal controlled local oscillator. Any RF at the input comes out at the lower frequencies. All signal processing is done in the TV set. A complete receiver with video and audio output would require all the TV sets circuitry, less the sweep and CRT components. There is no picture-quality gain by going direct from a receiver to a video monitor (as compared with a TV set) because IF and detector bandwidth are still the limiting factors.

A good low-noise amateur downconverter with 15 dB gain ahead of a TV set will give sensitivity close to the noise floor. A preamp located in the shack will not significantly increase sensitivity, but rather will reduce dynamic range and increase the probability of intermod interference. Sensitivity can be increased by increasing antenna-system gain: reducing feed-line loss, increasing antenna gain or adding an antenna mounted preamp (which will eliminate the coax loss plus any loss through transmit linear amplifier TR relays). Remember that each 6 dB increase in combination of transmitted power, reduced coax loss, antenna gain or receiver sensitivity can double the line-of-sight distance.

Foliage greatly attenuates the signal at UHF, so place antennas above the tree tops for the best results. Beams made for 432-MHz weak-signal work or 440-MHz FM may not have enough SWR bandwidth to cover all the ATV frequencies for transmiting, but they are okay for reception. A number of manufacturers now make ATV beam antennas to cover the whole band from 420 to 450 MHz. Use low-loss coax (such as Belden 9913: 2.5 dB/100 ft at 400 MHz) or Hardline for runs over 100 ft. All outside connectors must be weatherproofed with tape or coax sealer; any water that gets inside the coax will greatly increase the attenuation. Almost all ATV antennas use N connectors, which are more resistant to moisture contamination than other types.

Antenna polarization varies from area to area. Technically, the polarization should be chosen to give additional isolation (up to 20 dB) from other users near the channel. It is more common to find that the polarity was determined by the first local ATV operators (which antennas they had in place for other modes). Generally, those on 432 MHz SSB and weak-signal DX have horizontally polarized antennas, and those into FM, public service or repeaters will have vertical antennas. Check with local ATV operators before permanently locking down the antenna-mast clamps. Circularly polarized antennas let you work all modes, including satellites, with only 3 dB sacrificed when working a fixed polarity.

## ATV Frequencies

Standard broadcast TV channels are 6 MHz wide to accommodate the composite video, 3.58 MHz color and 4.5 MHz sound subcarriers. (See **Fig 12.60**.) Given the NTSC 525 horizontal line and 30 frames per second scan rates, the resulting horizontal resolution bandwidth is 80 lines per MHz. Therefore, with the typical TV set's 3-dB rolloff at 3 MHz (primarily in the IF filter), up to 240 vertical black lines can be seen. Color bandwidth in a TV set

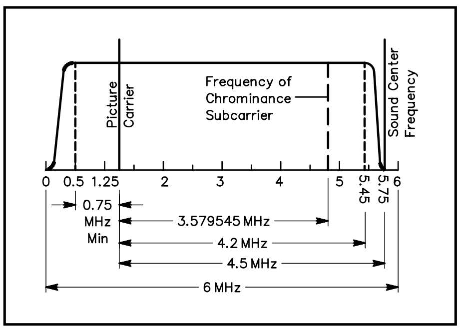



**Fig 12.60—A 6-MHz video channel with the video carrier 1.25 MHz up from the lower edge. The color subcarrier is at 3.58 MHz and the sound subcarrier at 4.5 MHz above the video carrier.**

is less than this, resulting in up to 100 color lines. Lines of resolution are often confused with the number of horizontal scan lines per frame. The video quality should be every bit as good as on a home video recorder.

The lowest frequency amateur band wide enough to support a TV channel is 70 cm (420 - 450 MHz), and it is the most popular. With transmit power, antenna gains and coax losses equal, decreasing frequency increases communication range. The 33-cm band goes half the distance that 70 cm does, but this can be made up to some extent with high-gain antennas, which are physically smaller at the higher frequency. A Technician class or higher license is required to transmit ATV on this band, and Novices can transmit ATV only in the 1270 to 1295 MHz segment of the 23-cm band.

Depending on local bandplan options, there is room for no more than two simultaneous ATV channels in the 33- and 70-cm bands without interference. Unlike cable channels, broadcast TV signals must skip a channel to keep a strong adjacent channel signal from interfering with a weaker on-channel signal. Cable companies greatly filter and equalize the signal amplitudes in order to use every channel.

Generally, because only two channels are available in the 70-cm band, an ATV repeater input on 439.25 or 434.0 MHz is shared with simplex. 421.25 MHz is the most popular in-band repeater output frequency. At least 12 MHz of separation is necessary for in-band repeaters because of filter-slope attenuation characteristics and TV-set adjacent-channel rejection. Some repeaters have their output on the 33-cm or 23-cm (the 923.25 and 1253.25 MHz output frequencies are most popular) bands which frees up a channel on 70 cm for simplex. Such cross-band repeaters also make it easier for the transmitting operator to monitor the repeated video with only proper antenna separation to prevent receiver desensitization. 426.25 MHz is used for simplex, public service and R/C models in areas with cross-band repeaters, or as a alternative to the main ATV activities on 434.0 or 439.25 MHz. Before transmitting, check with local ATV operators, repeater owners and frequency coordinators listed in the *ARRL Repeater Directory* for the coordinated frequencies used in your area.

Since a TV set receives a 6-MHz bandwidth, ATV is more susceptible to interference from many other sources than are narrower modes. Interference 40 dB below the desired signal can be seen in video. Many of our UHF (and above) amateur bands are shared with radar and other government radio positioning services. These show up as horizontal bars in the picture. Interference from amateurs who are unaware of the presence of the ATV signal (or in the absence of a technically sound and publicized local band plan) can wipe out the sound or color or put diagonal lines in the picture.

### DSB and VSB Transmission

While most ATV is double sideband (DSB) with the widest component being the sound subcarrier out plus and minus 4.5 MHz, over 90% of the spectrum power is in the first 1 MHz on both sides of the carrier for DSB or VSB (vestigial sideband). As can be seen in **Fig 12.61**, the video power density is down more than 30 dB at frequencies greater than 1 MHz from the carrier. DSB and VSB are both compatible with standard TV receivers, but the lower sound and color subcarriers are rejected in the TV IF filter as unnecessary. In the case of VSB, less than 5% of the lower sideband energy is attenuated. The other significant energy frequencies are the sound (set in the ATV transmitter at 15 dB below the peak sync) and the color at 3.58 MHz (greater than 22 dB down).

Narrow-band modes operating greater than 1 MHz above or

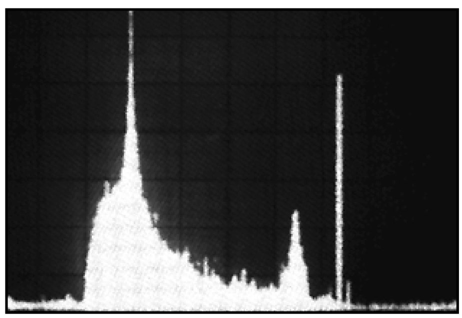


**Fig 12.61—A spectrum-analyzer photo of a color ATV signal. Each vertical division represents 10 dB; horizontal divisions are 1 MHz. Spectrum power density varies with picture content, but typically 90% of the sideband power is within the first 1 MHz.**

below the video carrier are rarely interfered with or know that the ATV transmitter is on unless the narrow-band signal is on one of the subcarrier frequencies or the stations are too near one another. If the band is full and the lower sideband color and sound subcarrier frequencies need to be used by a dedicated link or repeater, a VSB filter in the antenna line can attenuate them another 20 to 30 dB, or the opposite antenna polarization can be used for more efficient packing of the spectrum. Since all amateur linear amplifiers re-insert the lower sideband to within 10 dB of DSB, a VSB filter in the antenna line is the only cost-effective way to reduce the unnecessary lower sideband subcarrier energy if more than 1 W is used. In the more populated areas, 2-m calling or coordination frequencies are often used to work out operating time shifts, and so on, between all users sharing or overlapping the same segment of the band.

### ATV Identification

ATV identification can be on video or the sound subcarrier. A large high-contrast call-letter sign on the wall behind the operating table in view of the camera is the easiest way to fulfill the requirement. Transmitting stations fishing for DX during band openings often make up call-ID signs using fat black letters on a white background to show up best in the snow. Their city and 2-m monitoring frequency are included at the bottom of the sign to make beam alignment and contact confirmation easier.

Quite often the transmission time exceeds 10 minutes, especially when transmitting demonstrations, public-service events, space-shuttle video, balloon flights or a video tape. A company by the name of Intuitive Circuits makes a variety of boards that will overlay text on any video looped through them. Call letters and other information can be programmed into the board's non-volatile memory by on-board push buttons or an RS-232 line from a computer (depending on the version and model of the OSD board). There is even a model that will accept NMEA-0183 GPRMC data from a GPS receiver and overlay latitude, longitude, altitude, direction and speed as well as call letters on the applied camera video. This is ideal for ATV rockets, balloons and R/C vehicles. The overlaid ID can be selected to be on, off or flashed on for a few seconds every 10 minutes to automatically satisfy the ID requirement of 97.119 (see **Fig 12.62**). The PC Electronics VOR-2 board has an automatic nine minute timer, but it also has an end-of-transmission hang timer that switches to another video source for ID.

### A 20-W ATV Transceiver

Many newcomers to ATV start out by buying an inexpensive downconverter board just to check out the local simplex or repeater activity. Once they see a picture it isn't long before they want to transmit. The downconverter board can be kept separate or put in a larger chassis with transmitter boards to make one convenient package, as shown in **Fig 12.63**. All the modules shown here are available wired and tested from PC Electronics and are also functionally representative of what is available from other suppliers. **Fig 12.64** shows a block diagram of this transceiver.

The complete 20-W ATV transceiver consists of the

- *TVC-2G* downconverter (420 - 450 MHz in, TV channel 2, 3 or 4 out)
- *TXA5-70* 80 mW exciter/modulator
- *FMA5-F* 4.5-MHz sound subcarrier generator
- *PA5-70* 20-W brick linear amplifier
- *DMTR* video detector, video monitor driver and TR relay modules.


**Fig 12.62—A photo of an ATV image of the Space Shuttle interior with K6KMN's repeater ID overlaid. Automatic video overlay in the picture easily solves the 10-minute ID requirement for Space Shuttle retransmissions and other long transmissions.**


(A)

(B)

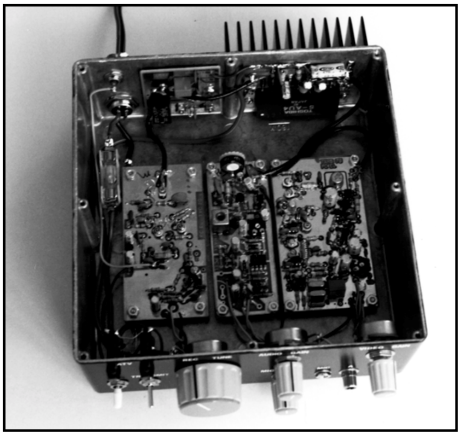


**Fig 12.63—A is the front view of a complete ATV transceiver and B is the inside view. This complete 20-W 70-cm ATV transceiver is assembled from readily available built and tested modules and mounted in a Hammond 1590F die-cast aluminum enclosure. On the box floor, left to right: TVC-2G downconverter, FMA5-F 4.5-MHz sound subcarrier generator and TXA5-70 80-mW exciter/video modulator. On the back (top left) is the downconverter-to-TV F connector and a 4-pin mic jack (which serves as the +13.8 V dc input). To the right is the DMTR TR relay board mounted to a flanged N connector. On the inside in front of the heatsink is the PA5 20-W power-amplifier module using a Mitsubishi M57716.**

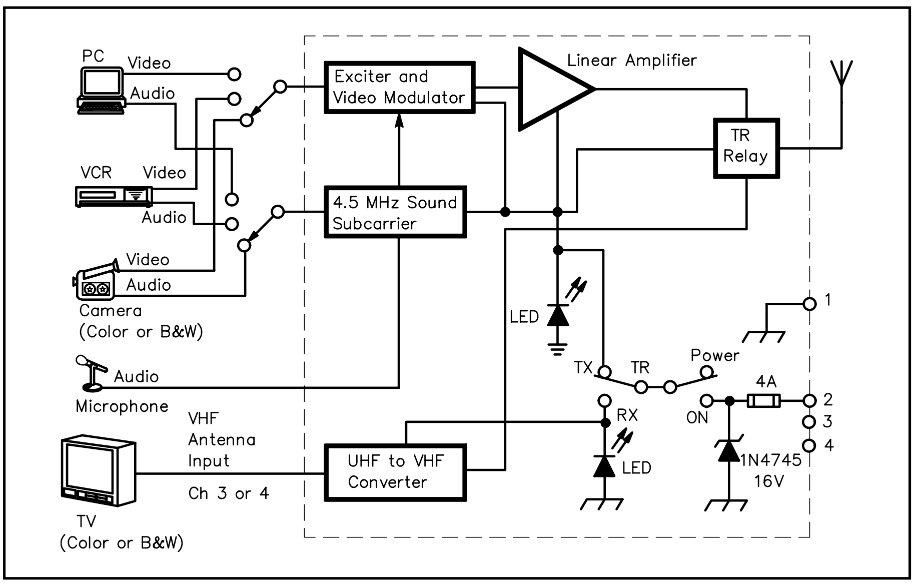



**Fig 12.64—Block diagram of a complete ATV station using the 20-W transceiver.**

The modules must be mounted in an aluminum enclosure for RF shielding and heat sinking. A 2.5×7×7-inch or larger aluminum chassis and bottom cover will make a nice transceiver. The Hammond 1590F diecast aluminum box makes a more rugged and RF tight enclosure. Lay all the modules in the selected chassis to position for best fit before drilling the mounting holes. Board wiring and mounting layouts come with each module.

Mount the PA5-70 amplifier and DMTR TR relay on the back panel, with the Mitsubishi M57716 RF power module as low as possible for best air flow. Unscrew the power module and its board from the heatsink and poke through the four mounting holes and a piece of paper with a pencil. Use this as a template to center punch the drill locations on the chassis from the outside. Make sure the heatsink will mount at least $^{1}/_{8}$ inch above the bottom edge of the chassis. Drill the $^{3}/_{16}$-inch diameter holes and carefully debur each side. The M57716 must be on a perfectly flat surface or the ceramic substrate could crack when its mounting bolts are tightened. Use a thin layer of heatsink compound under both the M57716 and the heatsink. Mount the M57716 and its board inside the chassis, and the heatsink outside by running the four screws from the M57716 side through the chassis into the heatsink.

The DMTR TR relay board mounts directly on a flange N UG58 chassis connector. Use RG-174 (small 50-Ω coax) for the RF leads to the amplifier and downconverter modules. To minimize RF coupling inside the chassis, carefully dress the coax braid back over its outer insulation (no more than $^{1}/_{4}$ inch) and solder the shield directly to the board ground planes. When soldering, make sure there are no bends or stress on the coax. Do not twist the braid into a “pig tail” at UHF.

A four-pin mic jack is used for the +13.8 V dc power connector. It is wired through a 4 A fast-acting fuse to the SPST POWER switch. The two unused pins can be used to control or power external devices such as a camera. A 1N4745A 16-V, 1-W Zener diode is connected from the transceiver side of the fuse to ground to help protect the circuits in case of accidental or reverse voltage. The downconverter, exciter and subcarrier generator can be mounted inside the chassis with #4-40×$^1/_2$ screws with double nuts for spacers (see module board mounting detail). Again, keep the exposed length of the interconnecting RG-174 center conductor less than $^1/_4$ inch. Solder the coax carefully and check with an ohmmeter for shorts. Use #18 wire for the amplifier power leads and #22 solid wire for all of the other wiring. Dress all dc leads away from the RF coaxes and the power module. The video and audio leads, and the panel-pot connections, can be #22 twisted pair (up to 6 inches long). Use RG-174 for longer runs.

You may want to remove and change some of the board mounted trimpots to panel mounted potentiometers to make adjustments easier. (For example, the video gain on the exciter, the mic and line gain on the sound subcarrier board, and the down-converter frequency tuning may be changed.) Remove the trimpots and run three wires from the mounting holes to their respective carbon (no wire wounds as they are inductive at video frequencies) panel potentiometers. 100-Ω carbon panel controls for the video gain are difficult to find, but they are available from PC Electronics.

For RF purposes, bypass each video input connector (100-pF ceramic disc capacitor) and each audio connector (220-pF disc) directly at the connector with short leads.

Most camcorders use phono jacks for the composite-video and line-audio connections. A low-impedance mic with push-to-talk can be used in parallel with the camera or VCR audio, which is mixed in the sound subcarrier board and the transmit receive toggle switch. An F connector on the back panel supplies downconverter output to the TV set antenna input. Use 75-Ω coax for the line to the TV. (300-Ω twin lead picks up too much interference from strong adjacent-channel broadcast TV stations.) Do not put any other boards inside the chassis that might be RF susceptible.

### *Transceiver Checkout*

Use an ohmmeter to verify that there are no short circuits in the coax or +13.8 V dc leads. (The antenna input will show a short because of the stripline tuned circuit.) Connect a good resonant 70-cm antenna, do not run a piece of wire or other band antenna just to try it out. With the transceiver off, connect the downconverter output coax to the TV set antenna jack. Switch the TV set on and select a channel that is not used in your area, usually 3 or 4. Adjust the fine tuning for minimum adjacent-channel interference. Then turn on the transceiver and adjust the downconverter tuning for a known nearby ATV station that you have contacted on 2 m. Peak the input trimmer cap on the TR relay board for minimum snow.

Next, with no video connected, switch the transmitter on for no more than 10 seconds at a time while verifying that you have less than 1 W of reflected power (as shown by an RF power meter in the antenna line). Continued transmission into an SWR of more than 2:1 can damage the SAU4 power module. If the SWR is low, peak the trimmer cap on the DMTR board for maximum output, then proceed to set the blanking pedestal pot on the TXA5-70 exciter.

ATV is a complex waveform that requires that the video to sync ratio remains constant throughout all of the linear amplifiers and with camera contrast changes (see **Fig 12.65**). The modulator contains a blanking clamp circuit that also acts as a sync stretcher to compensate for amplifier gain compression. To set this level, the pedestal control is set to maximum power output and then backed off to 60% of that value. The sync tip, which is the peak power, is constant at the maximum power read and the blanking level is the 60% point. This procedure must be repeated anytime a different power amplifier is added or applied voltage is changed by more than 0.5 V. Any other RF power measurements with an averaging power meter under video modulation are meaningless.

The camera video can now be connected, and the video gain set for best picture as described by the receiving station (or by observing a video monitor connected to the output jack on the DMTR board). Be

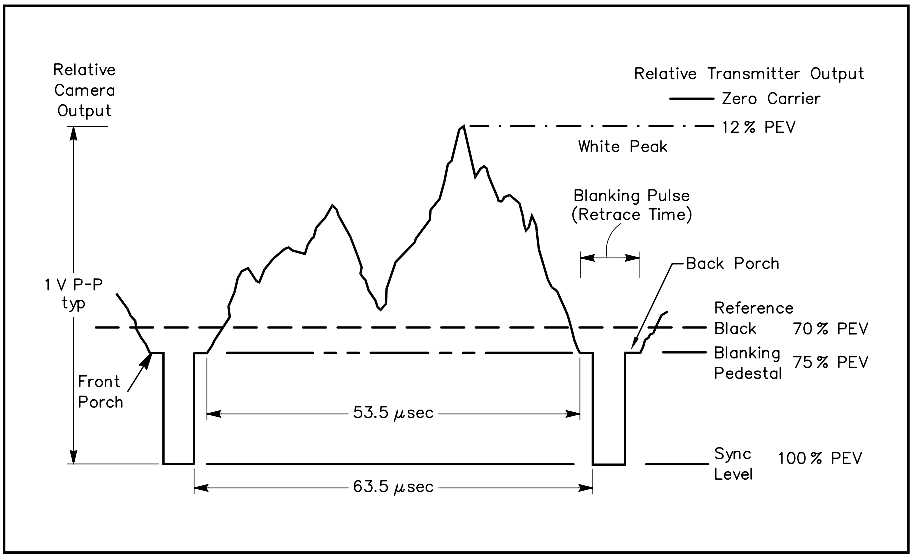



**Fig 12.65—An ATV waveform. Camera and corresponding transmitter RF output power levels during one horizontal line scan for black-and-white TV. (A color camera would generate a "burst"of 8 cycles at 3.58 MHz on the back porch of the blanking pedestal.) Note that "black" corresponds to a higher transmitter output power than does "white." For the purposes of blanking pedestal setup with a RF power meter rather than an oscilloscope, the 75% PEV corresponds to slightly less than 60% power.**

careful not to overmodu-late. Overmodulation is indicated by white smearing in the picture and sync buzz in the audio.

Connect a low-impedance (150 Ω to 600 Ω) dynamic mic (Radio Shack has some tape recorder replacements with a push-to-talk switch) into the mic jack and adjust the audio gain to a comfortable level as described by the receiving station. Electret mics are not good for this application because they are more susceptible to RF pickup (symptom: sync buzz in the audio). RF pickup may also be a problem with inadequately shielded mic cords. For example, it may be necessary to replace a cord having a spiral wrapped shield with one that has a braided shield, in order to improve shielding at UHF. The FMA5-F board has a soft limiter that comes in at the standard 25-kHz deviation.

The line-audio input has an independent volume control for the camcorder amplified mic or VCR audio, which is mixed with the low impedance mic input. This feature is great for voice-over commenting during video tapes.

## Driving Amplifiers with ATV

Wide-band AM video requires some special design considerations for linear amplifiers (as compared to those for FM and SSB amplifiers). Many high-power amateur amplifiers would oscillate (and possibly self destruct) from high gain at low frequencies if they were not protected by feedback networks and power RF chokes. These same stability techniques can affect some of the 5-MHz video bandwidth. Sync, color and sound can be very distorted unless the amplifier has been carefully designed for both stability and AM video modulation.

Mirage, Teletec and Down East Microwave either make special ATV amplifiers or offer standard models that were designed for all modes, including ATV. Basically the collector and base bias supplies have a range of capacitors to keep the voltage constant under modulation while at the same time using the minimum-value low-resistance series inductors or chokes to prevent self oscillation.

Almost all amateur linear power amplifiers have gain compression from half to their full rated peak envelope power. To compensate for this, the ATV exciter/modulator has a sync stretcher to maintain the proper transmitted video to sync ratio (see **Fig 12.66**). With both video and sound subcarrier disconnected, the pedestal control is set

**Fig 12.66—An oscilloscope used to observe a video waveform. The lower trace is the video signal as it comes out of the sync stretcher. The upper trace is the signal from the Mirage D1010-N amplifier.**

for maximum power output. Peak sync should first be set to 90% of the rated peak envelope power. (This is necessary to give some head room for the 4.5 MHz sound that is mixed and adds with the video waveform.) The TXA5-70 exciter/modulator has a RF power control to set this. Once this is done, the blanking pedestal control can be set to 60% of the peak sync value. For example, a 100-W amplifier would first be set for 90 W with the RF power control and then 54 W with the pedestal control. Then the sound subcarrier can be turned back on and the video plugged in and adjusted for best picture. If you could read it on a peak-reading power meter made for video, the power is between 90 and 100 W PEP. On a dc oscilloscope connected to a RF diode detector in the antenna line, it can be seen that the sync and blanking pedestal power levels remain constant at their set levels regardless of video gain setting or average picture contrast. On an averaging meter like a Bird 43, however, it is normal to read something less than the pedestal set up power.

## ATV Repeaters

Basically there are two kinds of ATV repeaters: in band and cross band. 70-cm in-band repeaters are more difficult to build and use, yet they are more popular because equipment is more available and less expensive. Indeed, cable-ready TV sets tune the 70-cm band with no modifications.

Why are 70-cm repeaters more difficult to build? The wide bandwidth of ATV makes for special filter requirements. Response across the 6-MHz passband must be as flat as possible with minimum insertion loss, but also must sharply roll off to reject other users as little as 12 MHz away. Special multipole interdigital or combline VSB filters are used to meet the requirement. An ATV duplexer can be used to feed one broadband omnidirectional antenna, but an additional VSB filter is needed in the transmitter line for sufficient attenuation of noise and IMD products.

A cross-band repeater, because of the great frequency separation between the input and output, requires less sophisticated filtering to isolate the transmitter and receiver. In addition, a cross-band repeater makes it easier for users to see their own video (no duplexer is needed, only sufficient antenna spacing). Repeater linking is easier too, if the repeater outputs alternate between the 23- and 33-cm bands.

**Fig 12.67** shows a block diagram for a simple 70-cm in-band repeater. No duplexer is shown because

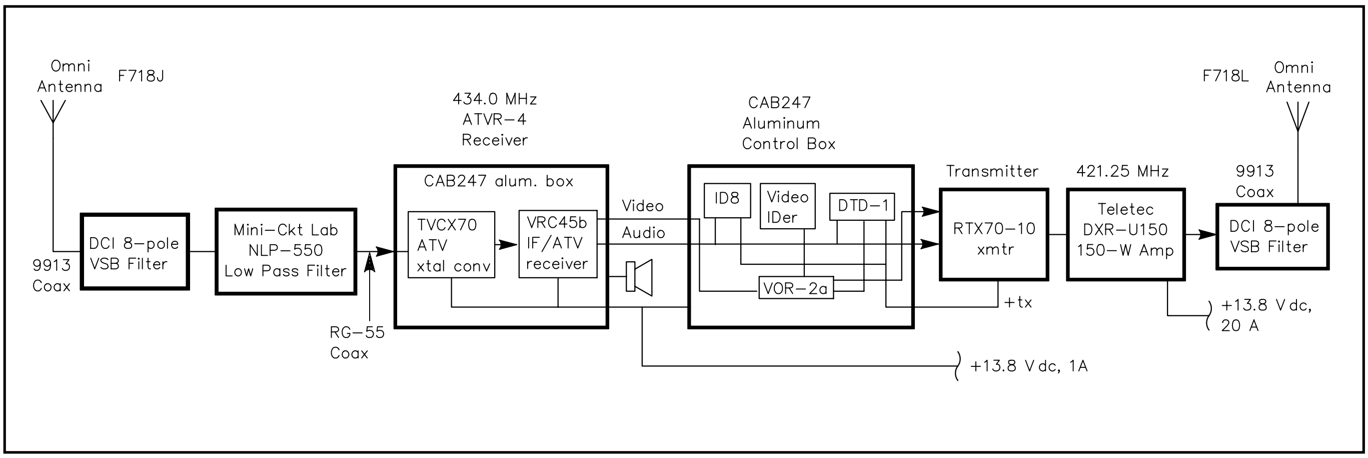



**Fig 12.67—A block diagram of a 70-cm in-band ATV repeater. The antennas are Diamond omnidirectional verticals, which require 20 ft (minimum) of vertical separation to prevent receiver desensitization. The VSB filters are made by DCI; they have the proper band-pass characteristics and only 1 dB insertion loss. A low pass filter on the reciever is also necessary because cavity type filters repeat a pass-band at odd harmonics and the third-harmonic energy from the transmitter may not be attenuated enough. The receiver, 10-W transmitter and VOR are made by PC Electronics. The Communications Specialists DTD-1 DTMF decoder and ID8 Morse identifier (optional if a video ID is used) are used to remotely turn the repeater transmitter on or off and to create a CW ID, respectively. Alternatively, an Intuitive Circuits ATV4-4 ATV repeater controller board can do all the control box functions as well as remotely select from up to four video sources.**

the antennas and VSB filters provide adequate isolation. The repeater transmitter power supply should be separate from the receiver and exciter supply. ATV is amplitude modulated, therefore the current varies greatly from maximum at the sync tip to minimum during white portions of the picture. Power supplies are not generally made to hold tight regulation with such great current changes at rates up to several megahertz. Even the power supply leads become significant inductors at video frequencies; they will develop a voltage across them that can be transferred to other modules on the same power-supply line.

To prevent unwanted key up from other signal sources, ATV repeaters use a video operated relay (VOR). The VOR senses the horizontal sync at 15,734 Hz in much the same manner that FM repeaters use CTCSS tones. Just as in voice repeaters, an ID timer monitors VOR activity and starts the repeater video ID generator every nine minutes or a few seconds after a user stops transmitting.

### Frequency Modulated ATV (FMATV)

While AM is the most popular mode because of greater equipment availability, lower cost, less occupied bandwidth and use of a standard TV set, FMATV is gaining interest among experimenters and also repeater owners for links. FM on the 1200-MHz band is the standard in Europe because there is little room for video in their allocated portion of the 70-cm band. FMATV occupies 17 to 21 MHz depending on deviation and sound subcarrier frequency. The US 70-cm band is wide enough but has great interference potential in all but the less populated areas. Most available FMATV equipment is made for the 1.2, 2.4 and 10.25-GHz bands. **Fig 12.68** is a block diagram of an FMATV receiver.

The US standard for FMATV is 4 MHz deviation with the 5.8-MHz sound subcarrier set to 10 dB below the video level. 1252 or 1255 MHz are suggested frequencies in order to stay away from FM voice repeaters and other users higher in the band while keeping sidebands above the 1240-MHz band edge. Using the US standard with Carsons rule for FM occupied bandwidth, it comes out to just under 20 MHz. So 1250 MHz would be the lowest possible frequency. Almost all modern FMATV equipment is synthesized, but if yours is not, use a frequency counter to monitor the frequency for warm up drift. Check with local frequency coordinators before transmitting because the band plan permits other modes in that segment.

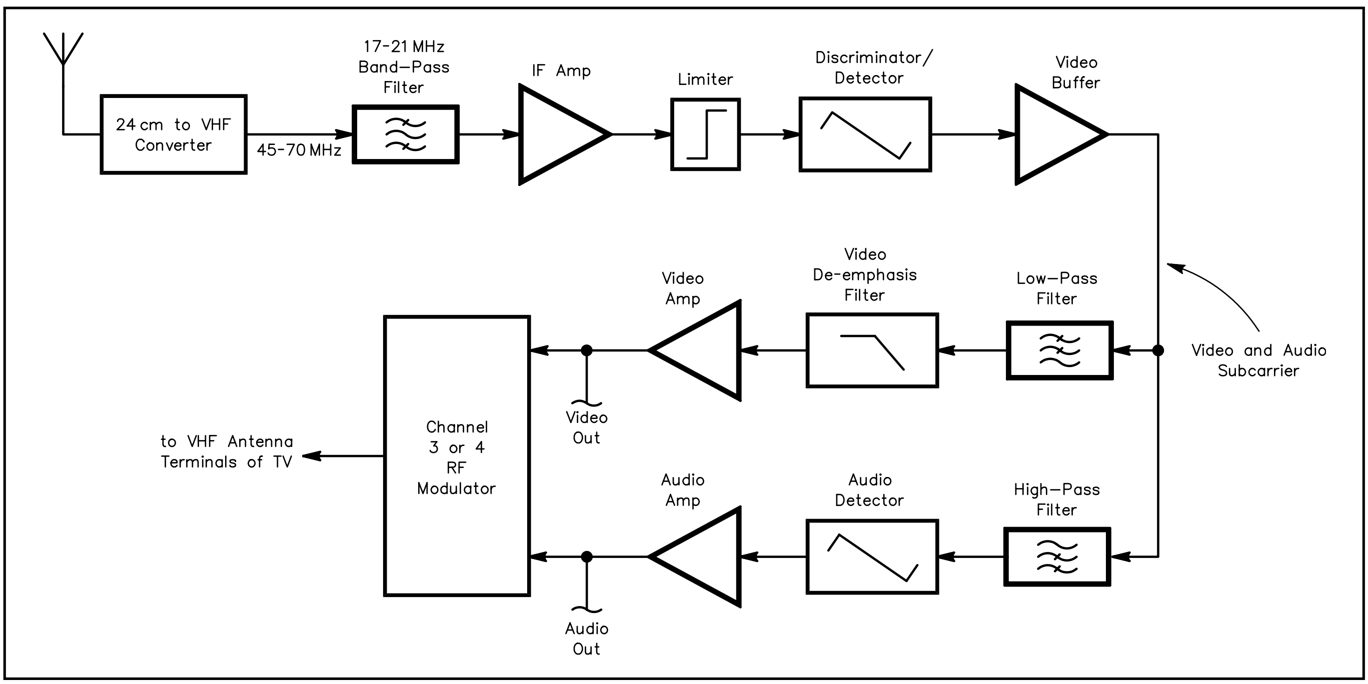



**Fig 12.68—Block diagram of an FMATV receiver.**

Experimentally, using the US standard, FMATV gives increasingly better picture-to-noise ratios than AMATV at receiver input signals greater than 5 µV. Because of the wider noise bandwidth and FM threshold effect, AM video can be seen in the noise well before FM. For DX work, it has been shown that AM signals are recognizable signals in the snow at four times (12 dB) greater distance than FM signals with all other factors equal. Above the FM threshold, however, FM rapidly overtakes AM; snow-free pictures occur above 50 µV, or 4 times farther away than with AM signals. The crossover point is near the signal level where sound and color begin to appear for both systems. **Fig 12.69** compares AM and FMATV across a wide range of signal strengths.

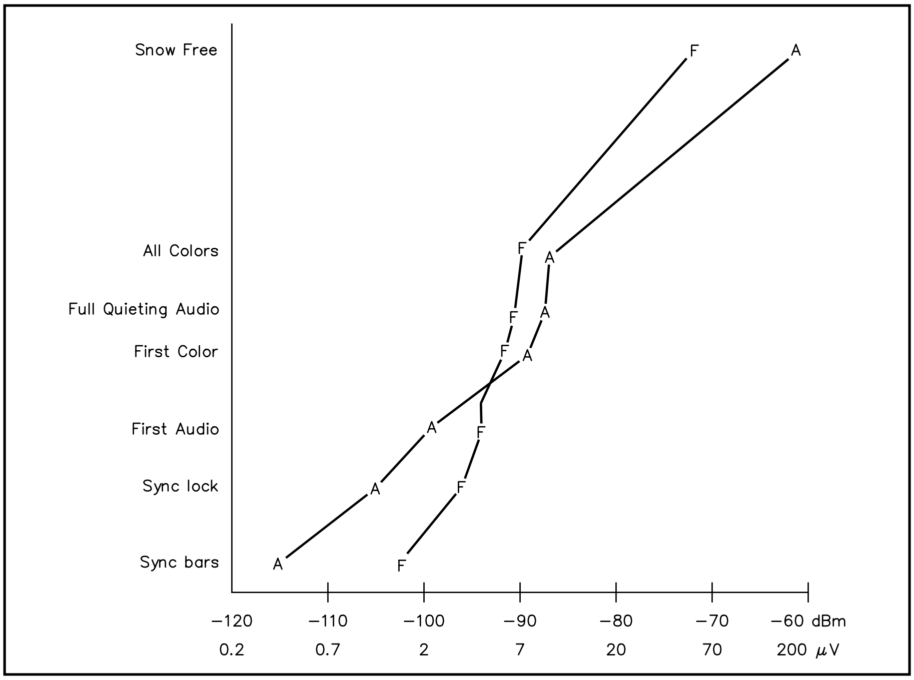



**Fig 12.69—Two approaches to ATV receiving. This chart compares AM (A) and FM (F) ATV as seen on a TV receiver and monitor. Signal levels are into the same downconverter with sufficient gain to be at the noise floor. The FM receiver bandwidth is 17 MHz, using the US standard.**

There are a variety of methods to receive FMATV. Older satellite receivers have a 70 or 45-MHz input and require a down converter with 40 to 50 dB gain ahead of them. Also satellite receivers are made for wider deviation and need some video gain to give the standard 1 volt peak to peak video output when receiving a signal with the standard 4-MHz deviation. Current satellite receivers directly tune anywhere from 900 to 2150 MHz and they only need a preamp added at the antenna for use on the 33 and 23-cm ham bands. The additional video gain can often be had by adjusting an internal pot or changing the gain with a resistor.

Some of the inexpensive Part 15 license-free wireless video receivers in the 33-cm band are 4 MHz deviation FM video, and most of the 2.4 GHz ones are FM, which can be used directly. However, they may or may not have the standard de-emphasis video network which may have to be added. On 2.4 GHz, some of the Part 15 frequencies are outside the band and care should be taken to use only those inside the 2390 to 2450 MHz ham band if modified. Wavecom Jr has been the most popular 2.4 GHz license-free video transmitter and receiver (available from ATV Research) and have been modified for higher power and other features as well as having all 4 of the channels in the ham band using interface boards from PC Electronics.

Gunnplexers on 10.4 GHz make inexpensive point to point ATV links for public-service applications or between repeaters. A 10-mW Gunnplexer with 17-dB horn can cover over 2 miles line-of-sight when received on a G8OZP low noise 3-cm LNB and satellite receiver. An application note for construction of the 3-cm transmitter comes with the GVM-1 Gunnplexer video modulator board from PC Electronics.

For short distance ATV from R/C vehicles, low-power FM ATV modules with 50 to 100-mW output in the 33, 23 or 13-cm bands are often used. These offer less desense possibility to the R/C receiver. An example can be seen on the model Humvee in the photo.

For greater distance such as with R/C aircraft, use up to a 1-W ATV transmitter board operating in the 70-cm band. Since R/C receivers at 50 or 72 MHz were not designed to be placed right next to a transmitter, it is necessary to shield the R/C receiver and put a simple 3-pole 100-MHz low-pass filter at the antenna input. An application note is available from PC Electronics.

### Further ATV Reading

*Amateur Television Quarterly Magazine*.

*CQ-TV*, British ATV Club, a quarterly publication available through *Amateur Television Quarterly Magazine*.

Ruh, *ATV Secrets for the Aspiring ATVers*, Vol 1, 1991 and Vol 2, 1992. Available through *Amateur Television Quarterly Magazine*.

Taggart, "An Introduction to Amateur Television," April, May and June 1993 *QST*.

### ATV Equipment Sources

Contact information for these sources appears in the Address List in the **References** chapter.

Advanced Receiver Research
ATV Research
Digital Communications, Inc (DCI)
Down East Microwave
Elktronics
Intuitive Circuits
Mini-Circuits Labs
PC Electronics
Phillips-Tech Electronics
Spectrum International
TX/RX Systems
Teletec
Wyman Research Inc

# Radio Control


N8QPJ mounted an ATV setup aboard this model Humvee.

Amateur Radio gave birth to the radio control (R/C) hobby as we know it today. Part 97 of the FCC regulations (§97.215) specifically permits "remote control of model craft" as a licensed amateur station activity. Station identification is not required for R/C, and the transmitter power is limited to 1 W. Before 1950, development of telecommand radio systems small enough to be used for remote radio control of model aircraft, cars and boats, was primarily an Amateur Radio activity. In the early 1950s, the FCC licensed R/C transmitter operation on nonham frequencies, without an operator license examination. The invention of the transistor and the subsequent increase in R/C development activity lead to the sophisticated electronic control systems in use today. This section was contributed by H. Warren Plohr, W8IAH.

The simplest electronic control systems are currently used in low-cost toy R/C models. These toys often use simple on/off switching control that can be transmitted by on/off RF carrier or tone modulation. More expensive toys and R/C hobby models use more sophisticated control techniques. Several simultaneous proportional and switching controls are available, using either analog or digital coding on a single RF carrier.

R/C hobby sales records show that control of model cars is the most popular segment of the hobby. Battery powered cars like that shown in **Fig 12.70** are the most popular. Other popular types include models powered by small internal combustion "gas" engines.

R/C model aircraft are next in the line of popularity and include a wide range of styles and sizes. Fixed-wing models like those shown in **Fig 12.71** are the most popular. They can be unpowered (gliders) or powered by either electric or "gas" motors. The basic challenge for a new model pilot is to operate the model in flight without crashing. Once this is achieved, the challenge extends to operating detailed scaled models in realistic flight, performing precision aerobatics, racing other models or engaging in model-to-model combat. The challenge for the R/C glider pilot is to keep the model aloft in rising air currents. The most popular rotary-wing aircraft models are helicopters. The sophistication of model helicopters and their control systems can only be appreciated when one sees a skilled pilot perform a


Fig 12.70—Photo of three R/C model electric cars.

Fig 12.71—Photo of two R/C aircraft models.

schedule of precision flight maneuvers. The most exotic maneuver is sustained inverted flight, a maneuver not attainable by a full-scale helicopter.

R/C boats are another facet of the hobby. R/C water craft models can imitate full-scale ships and boats. From electric motor powered scale warships that engage in scale battles, to "gas" powered racing hydroplanes, model racing yachts and even submarines.

Most R/C operation is no longer on Amateur Radio frequencies. The FCC currently authorizes 91 R/C frequencies between 27 MHz and 76 MHz. Some frequencies are for all odels, some are for aircraft only and others for surface (cars, boats) models only. Some frequencies are used primarily for toys and others for hobbyist models. Amateur Radio R/C operators use the 6-m band almost exclusively. Spot frequencies in the upper part of the band are used in geographical areas where R/C operation is compatible with 6-m repeater operation and TV Channel-2 signals that can interfere with control. Eight spot frequencies, 53.1 to 53.8 MHz, spaced 100 kHz apart, are used. There is also a newer 200 kHz R/C band from 50.8 to 51.0 MHz providing ten channels spaced 20 kHz apart. The close channel spacing in this band requires more selective receivers than do the 53-MHz channels. The AMA *Membership Manual* provides a detailed list of all R/C frequencies in current use.[7] The *ARRL Repeater Directory* lists current Amateur Radio R/C frequencies.

**Fig 12.72** shows a typical commercial R/C system, consisting of a hand-held aircraft transmitter (A), a multiple-control receiver, four control servos and a battery (B). This particular equipment is available for any of the ten R/C frequencies in the 50.80 to 51.00 MHz band. Other commercially available control devices include relays (solid-state and mechanical) and electric motor speed controllers.

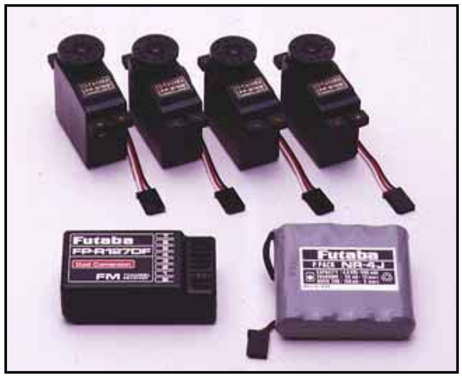
Some transmitters are tailored to specific kinds of models. A helicopter, for example, requires simultaneous control of both collective pitch and engine throttle. A model helicopter pilot commands this response with a linear motion of a single transmitter control stick. The linear control stick signal is conditioned within the transmitter to provide the encoder with a desired combination of nonlinear signals. These signals then command the two servos that control the vertical motion of the helicopter.

Transmitter control-signal conditioning is provided by either analog or digital circuitry. The signal conditioning circuitry is often designed to suit a specific type of model, and it is user adjustable to meet an individual model's control need. (Low-cost transmitters use analog circuitry.) They are available for helicopters, sailplanes and pattern (aerobatic) aircraft.

More expensive transmitters use digital microprocessor circuitry for signal conditioning. **Fig 12.73** shows a transmitter

[7] Academy of Model Aeronautics *Membership Manual* is available from AMA, Muncie, IN 47302.


**(A)**




**(B)**

**Fig 12.72—A, photo of Futaba's Conquest R/C aircraft transmitter. B shows the matching airborne system.**

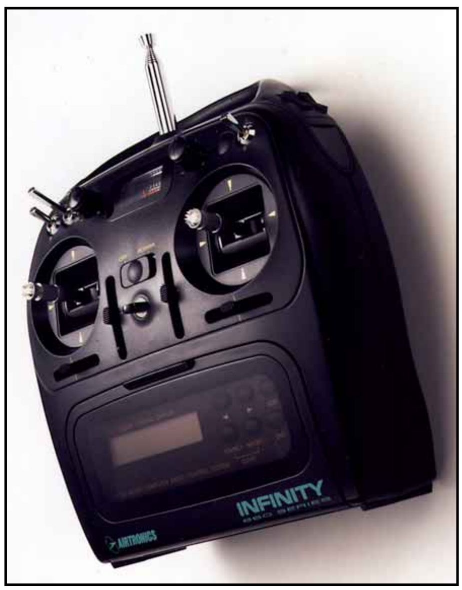



**Fig 12.73—Photo of Airtronics Infinity 660 R/C aircraft transmitter.**

that uses a programmable microprocessor. It is available on any 6-m Amateur Radio R/C frequency with switch-selectable PPM or PCM coding. It can be programmed to suit the needs of a helicopter, sailplane or pattern aircraft. Nonvolatile memory retains up to four user-programmed model configurations.

Many R/C operators use the Amateur Radio channels to avoid crowding on the nonham channels. Others do so because they can operate home-built or modified R/C transmitters without obtaining FCC type acceptance. Still others use commercial R/C hardware for remote control purposes around the shack. Low-cost R/C servos are particularly useful for remote actuation of tuners, switches and other devices. Control can be implemented via RF or hard wire, with or without control multiplexing.

## R/C RF MODULATION

The coded PPM or PCM information for R/C can modulate an RF carrier via either amplitude- or frequency-modulation techniques. Commercial R/C systems use both AM and FM modulation for PPM, but use FM exclusively for PCM.

The AM technique used by R/C is 100% "down modulation." This technique switches the RF carrier off for the duration of the PPM pulse, usually 250 to 350 µs. A typical transmitter design consists of a third-overtone transistor oscillator, a buffer amplifier and a power amplifier of about 1/2 W output. AM is achieved by keying the 9.6-V supply to the buffer and final amplifier.

The FM technique used by R/C is frequency shift keying (FSK). The modulation is applied to the crystal-oscillator stage, shifting the frequency about 2.5 or 3.0 kHz. The direction of frequency shift, up or down with a PPM pulse or PCM code, can be in either direction, as long as the receiver detector is matched to the transmitter. R/C manufacturers do not standardize, so FM receivers from different manufacturers may not be compatible.

## SIGNALING TECHNIQUES

### *Background*

Radio control (R/C) of models has used many different control techniques in the past. Experimental techniques have included both frequency- and time-division multiplexing, using both electronic and mechanical devices. Most current systems use time-division multiplexing of pulse-width information. This signaling technique, used by hobbyist R/C systems, sends pulse-width information to a remotely located pulse-feedback servomechanism. Servos were initially developed for R/C in the 1950s and are still used today in all but low-cost R/C toys.

**Fig 12.74** is a block diagram of a pulse-feedback servo. The leading edge of the input pulse triggers a linear one-shot multivibrator. The width of the one-shot output pulse is compared to the input pulse. Any pulse width difference is an error signal that is amplified to drive the motor. The motor drives a feedback potentiometer that controls the one-shot timing. When this feedback loop reduces the error signal to a few microseconds, the drive motor stops. The servo position is a linear function of the input pulse width. The motor-drive electronics are usually timed for pulse repetition rates of 50 Hz or greater and a pulse width range of 1 to 2 ms. A significantly slower repetition rate reduces the servomechanism slew rate but not the position accuracy.

In addition to motor driven servos, the concept of pulse-width comparison can be used to operate solid-state or mechanical relay switches. The same

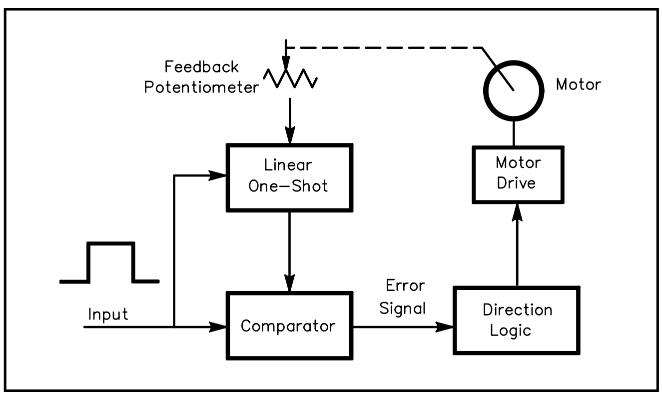



**Fig 12.74—Diagram of a pulse-feedback servo.**

concept is used in solid-state proportional electric motor speed controllers. These speed controllers are used to operate the motors powering model cars, boats and aircraft. Currently available model speed controllers can handle tens of amperes of direct current at voltages up to 40 V dc using MOSFET semiconductor switches.

### *Requirements*

The signaling technique required by R/C is the transmission of 1- to 2-ms-wide pulses with an accuracy of ±1 μs at repetition rates of about 50 Hz. A single positive-going dc pulse of 3 to 5 V amplitude can be hard wire transmitted successfully to operate a single control servomechanism. If such a pulse is sent as modulation of an RF carrier, however, distortion of the pulse width in the modulation/demodulation process is often unacceptable. Consequently, the pulse-width information is usually coded for RF transmission. In addition, most R/C systems require pulse-width information for more than one control. Time-division multiplexing of each control provides this multichannel capability. Two coding techniques are used to transfer the pulse-width information for multiple control channels, pulse-position modulation (PPM) and pulse-code modulation (PCM).

## Pulse-Position Modulation

PPM is analog in nature. The timing between transmitted pulses is an analog of the encoded pulse width. A train of pulses encodes multiple channels of pulse-width information as the relative position or timing between pulses. Therefore the name, pulse-position modulation. The transmitted pulse is about 300 μs in width and uses slow rise and fall times to minimize the transmitter RF bandwidth. The shape of the received waveform is unimportant because the desired information is in the timing between pulses. **Fig 12.75** diagrams a frame of five pulses that transmits four control channels of pulse-width information. The frame of modulation pulses is clocked at 50 Hz for a frame duration of 20 ms. Four multiplexed pulse widths are encoded as the times between five 300-μs pulses. The long period between the first and the last pulse is used by the decoder for control-channel synchronization.

PPM is often incorrectly called digital control because it can use digital logic circuits to encode and decode the control pulses. A block diagram of a typical encoder is shown in **Fig 12.76**. The 50-Hz clock frame generator produces the first 300-μs modulation pulse and simultaneously triggers the first one-shot in a chain of multivibrators. The trailing edge of each one-shot generates a 300-μs modulation pulse while simultaneously triggering the succeeding multivibrator one-shot. In a four-channel system the fifth modulation pulse, which indicates control of the fourth channel, is followed by a modulation pause that is dependent on the frame rate. The train of 300-μs pulses are used to modulate the RF.

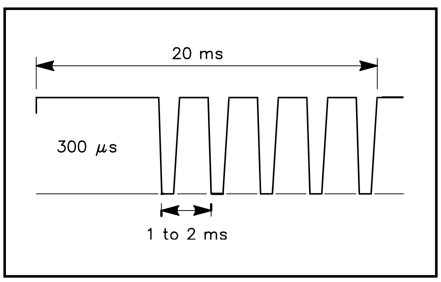



**Fig 12.75—Diagram of a four-channel PPM RF envelope.**

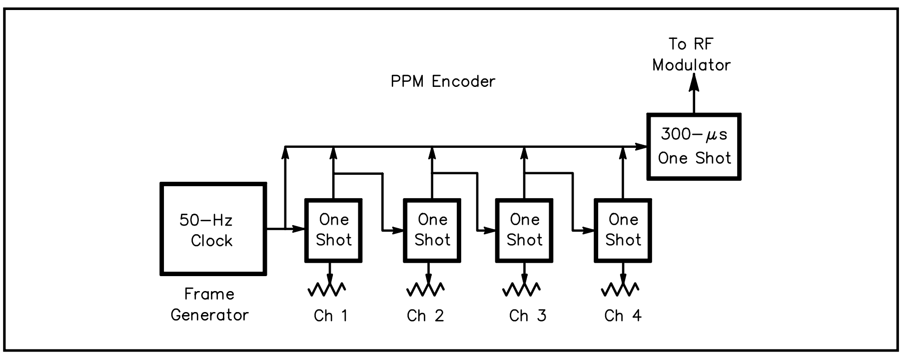



**Fig 12.76—Diagram of a PPM encoder.**

Received pulse decoding can also use digital logic semiconductors. **Fig 12.77** shows a simple four-control-channel decoder circuit using a 74C95 CMOS logic IC. The IC is a 4-bit shift register operated in the right-shift mode. Five data pulses spaced 1 to 2 ms apart, followed by a synchronization pause, contain the encoded pulse-width information in one frame. During the sync pause, the RC circuit discharges and sends a logic-one signal to the 74C95 serial input terminal. Subsequent negative going data pulses remove the logic-one signal from the serial input and sequentially clock the logic one through the four D-flip-flops. The output of each flip-flop is a positive going pulse, with a width corresponding to the time between the clocking pulses. The output of each flip-flop is a demultiplexed signal that is used to control the corresponding servo.

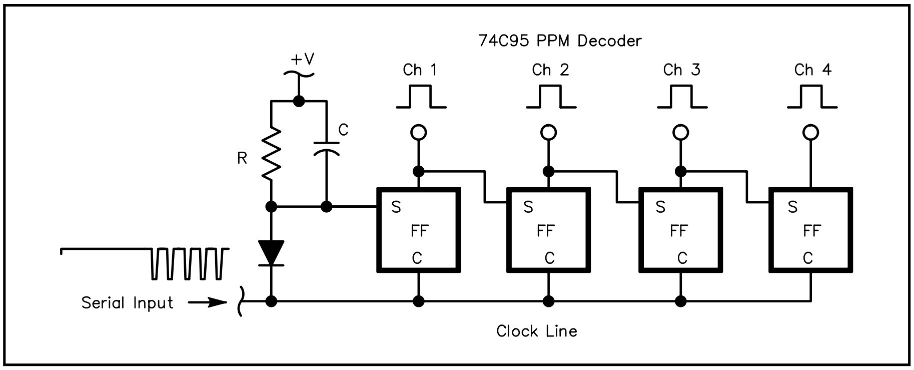



Fig 12.77—Diagram of a 74C95 PPM decoder.

## Pulse Code Modulation

PCM uses true digital code to transfer R/C signals. The pulse width data of each control channel is converted to a binary word. The digital word information of each control channel is coded and multiplexed to permit transmission of multiple channels of control on a single RF carrier. On the receiving end, the process is reversed to yield the servo control signals.

There is no standard for how the digital word is coded for transmission. Therefore PCM R/C transmitters and receivers from different makers are not interchangeable. Some older PCM systems provide only 256 discrete positions for 90° of servo motion, thereby limiting servo resolution. Newer systems use more digital bits for each word and provide smooth servo motion with 512 and 1024 discrete positions. All PCM and PPM systems use the same servo input-signal and supply voltages. Therefore the servos of different manufacture are interchangeable once compatible wiring connectors have been installed.

# Spread Spectrum

This introduction to spread spectrum communications was written by André Kesteloot, N4ICK. *The ARRL Spread Spectrum Sourcebook* contains a more complete treatment of the subject.

## A Little History

Spread spectrum has existed at least since the mid 1930s. Despite the fact that John Costas, W2CRR, published a paper on nonmilitary applications of spread spectrum communications in 1959,[8] spread spectrum was used almost solely for military purposes until the late 1970s. In 1981, the FCC granted the Amateur Radio Research and Development Corporation (AMRAD) a Special Temporary Authorization to conduct Amateur Radio spread spectrum experiments. In June 1986, the FCC authorized all US amateurs to use spread spectrum above 420 MHz.

## Why Spread Spectrum

Faced with increasing noise and interference levels on most RF bands, traditional wisdom still holds that the narrower the RF bandwidth, the better the chances that "the signal will get through." This is not so.

In 1948, Claude Shannon published his famous paper, "A Mathematical Theory of Communication" in the *Bell System Technical Journal*, followed by "Communications in the Presence of Noise" in the *Proceedings of the IRE* for January 1949. A theorem that follows Shannon's, known as the Shannon-Hartley theorem, states that the channel capacity C of a band-limited gaussian channel is

$$C = W \log_2 (1 + S/N) \text{ bits/s} \qquad (6)$$

where
W is the bandwidth,
S is the signal power and
N is the noise within the channel bandwidth.

This theorem states that should the channel be perfectly noiseless, the capacity of the channel is infinite. It should be noted, however, that making the bandwidth W of the channel infinitely large does *not* make the capacity infinite, because the channel noise increases proportionately to the channel bandwidth.

Within reason, however, one can trade power for bandwidth. In addition, the power density at any point of the occupied bandwidth can be very small, to the point that it may be well *below* the noise floor of the receiver. The US Navy Global Positioning System (GPS) is an excellent example of the use of what is called direct-sequence spread spectrum. The average signal at the GPS receiver's antenna terminals is approximately −160 dBW (for the C/A code). Since most sources of interference are relatively narrow-band, spread-spectrum users will also benefit, as narrow-band interfering signals are rejected automatically during the despreading process, as will be explained later in this section.

These benefits are obtained at the cost of fairly intricate circuitry: The transmitter must spread its signal over a wide bandwidth in accordance with a certain prearranged code, while the receiver must somehow synchronize on this code and recombine the received energy to produce a usable signal. To generate the code, use is made of pseudo-noise (PN) generators. The PN generators are selected for their correlation properties. This means that when two similar PN sequences are compared out of phase their correlation is nil (that is, the output is 0), but when they are exactly in phase their correlation produces a huge peak that can be used for synchronization purposes.

This synchronization process has been (and still is) the major

[8] "Poisson, Shannon and the Radio Amateur," *Proceedings of the IRE*, Dec 1959.

complicating factor in any spread spectrum link, for how can one synchronize on a signal that can be well below the receiver's noise floor? Because of the cost associated with the complicated synchronization processes, spread spectrum applications were essentially military-related until the late 1970s. The development of ICs then allowed for the replacement of racks and racks of tube equipment by a few plug-in PC boards, although the complexity level itself did not improve. Amateur Radio operators could not afford such levels of complexity and had to find simpler solutions, at the cost of robustness in the presence of interference.

### Spread-Spectrum Transmissions

A transmission can be called "spread spectrum" if the RF bandwidth used is (1) much larger than that needed for traditional modulation schemes and (2) independent of the modulation content. Although numerous spread spectrum modulation schemes are in existence, only two, frequency-hopping (FH) and direct-sequence spread spectrum (DSSS) are specifically authorized by the FCC for use by the Amateur Radio community.

To understand FH, let us assume a transmitter is able to transmit on any one of 100 discrete frequencies F1 through F100. We now force this equipment to transmit for 1 second on each of the frequencies, but in an apparently random pattern (for example, F1, F62, F33, F47…; see **Fig 12.78**). Should some source interfere with the receiver site on three of those discrete frequencies, the system will still have achieved reliable transmission 97% of the time. Because of the built-in redundancy in human speech, as well as the availability of error-correcting codes in data transmissions, this approach is particularly attractive for systems that must operate in heavy interference.

In a DSSS transmitter, an RF carrier and a pseudo-random pulse train are mixed in a doubly balanced mixer (DBM). In the process, the RF carrier disappears and is replaced by a noise-like wide-band transmission, as shown in **Fig 12.79**. At the receiver, a similar pseudo-random signal is reintroduced and the spread spectrum signal is correlated, or despread, while narrow-band interference is spread simultaneously by the same process.

The technical complexity mentioned above is offset by several important advantages for military and space applications:

- *Interference rejection*. If the interference is not synchronized with the original spread spectrum signal, it will not appear after despreading at the receiver.
- *Security*. The length and sophistication of the pseudo-random codes used can be such as to make unauthorized recovery difficult if not impossible.
- *Power density*. Low power density makes for easy hiding of the RF signal and a resulting lower probability of detection.

As far as the Amateur Radio community is concerned, particular benefit will be derived from the interference rejection just mentioned, as it offers both robustness and reliability of transmissions, as well as low probability of interference to other users. Additionally, spread spectrum has the potential to allow better utilization of the RF spectrum allocated to amateurs. There is a limit as to how many conventional signals can be placed in a given band before serious transmission degradation takes place. Addi-

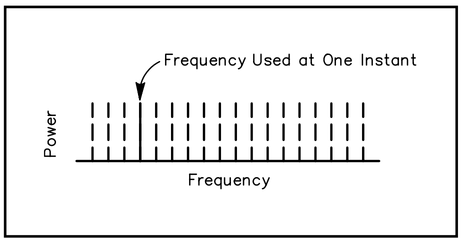



**Fig 12.78—Power vs frequency for frequency-hopping spread spectrum signals. Emissions jump around to discrete frequencies in pseudo-random fashion.**

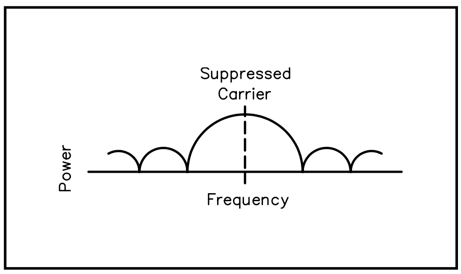



**Fig 12.79—Power vs frequency for a direct-sequence-modulated spread spectrum signal. The envelope assumes the shape of a (sin x/x)$^2$ curve. With proper modulating techniques, the carrier is suppressed.**

tional spread spectrum signals will not cause severe interference, but may instead only raise the background noise level. This becomes particularly important in bands shared with other users and in our VHF and UHF bands increasingly targeted by would-be commercial users. The utilization of a channel by many transmitters is essentially the concept behind CDMA (Code Division Multiple Access), a system in which several DSSS transmissions can share the same RF bandwidth, provided they utilize orthogonal pseudo-random sequences.

## Amateur Spread Spectrum

When radio amateurs (limited in both financial resources and time available for experimentation) decided to try their hand at spread spectrum transmissions, they had to attack the problem by simplifying several assumptions. Security and privacy, the primary goals of the military, were sacrificed in favor of simplicity of design and implementation.

Experimentation sponsored by AMRAD began in 1981 and continues to this day. These experiments have lead to the design and construction of a practical DSSS UHF link. This project was described in May 1989 *QST* and was reprinted in *The ARRL Spread Spectrum Sourcebook*. In it, N4ICK offered a simple solution to the problem of synchronization. (Because of its simplicity, this solution does not offer all the anti-jamming properties of more sophisticated systems, but this should not be of concern to Amateur Radio operators.) The block diagram is shown in **Fig 12.80**. **Fig 12.81** shows the RF signals at the transmitter output, at the receiver antenna terminals and the recovered signal after correlation. James Vincent, G1PVZ, replaced the original FM scheme with a continuously variable delta modulation system, or CVSD. A description of his work can be found in the September and October 1993 issues of the British magazine *Electronics World & Wireless World.*

In addition to *The ARRL Spread Spectrum Sourcebook*, interested readers may want to pay particular attention to Robert Dixon's text, *Spread Spectrum Systems*. Additional information can be found in the publications and magazines listed below.

## SS References


Dixon, *Spread Spectrum Systems*, second edition, 1984, Wiley Interscience, New York.
Dixon, *Spread Spectrum Techniques*, IEEE.
Golomb, *Shift Register Sequences*, 1982, Aegean Park Press, Laguna Hills, California.

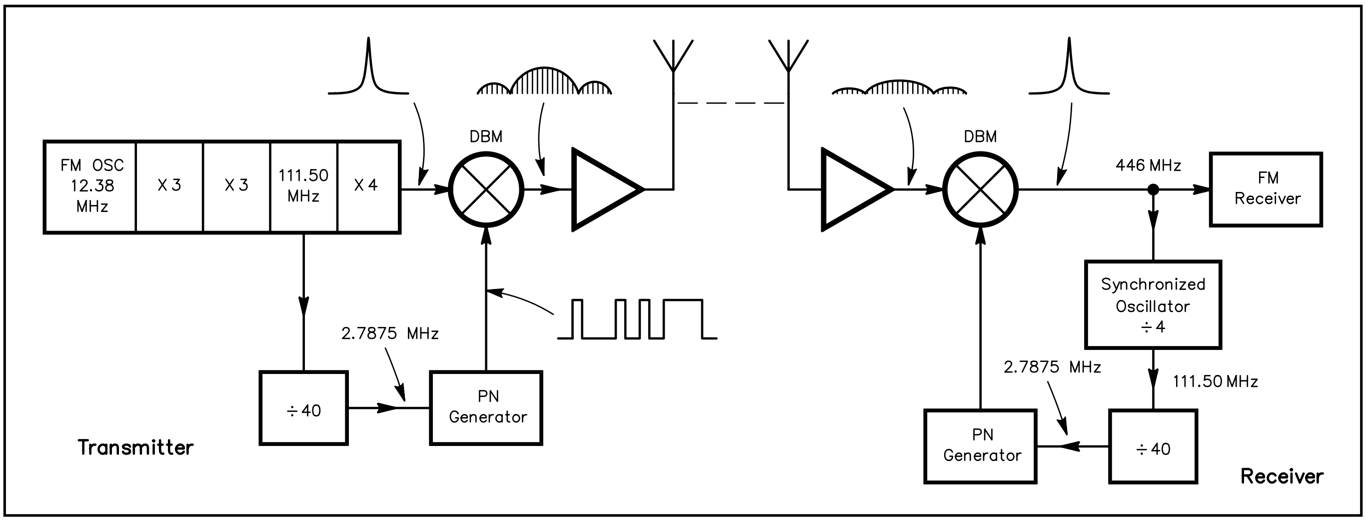




**Fig 12.80—A block diagram of the practical spread spectrum link. The success of this arrangement lies in the use of a synchronized oscillator (right) to recover the transmitter clock signal at the receiving site.**

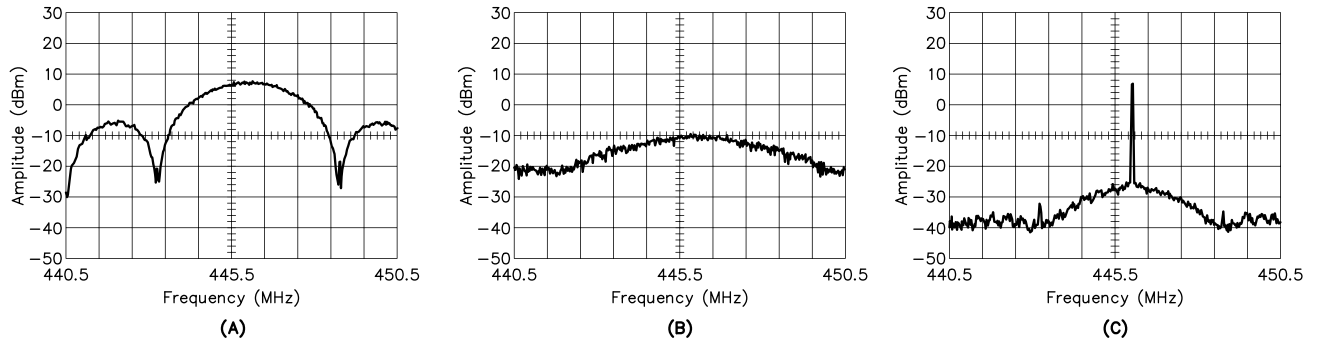



**Fig 12.81—(A) The envelope of the unfiltered biphase-modulated spread spectrum signals as viewed on a spectrum analyzer. In this practical system, band-pass filtering is used to confine the spread spectrum signal to the amateur band. (B) At the receiver end of the line, the filtered spread spectrum signal is apparent only as a 10-dB hump in the noise floor. (C) The despread signal at the output of the receiver DBM. The original carrier—and any modulation components that accompany it—has been recovered. The peak carrier is about 45 dB above the noise floor—more than 30 dB above the hump shown at B. (These spectrograms were made at a sweep rate of 0.1 s/division and an analyzer bandwidth of 30 kHz; the horizontal scale is 1 MHz/division.)**

Hershey, *Proposed Direct Sequence Spread Spectrum Voice Techniques for Amateur Radio Service*, 1982, US Department of Commerce, NTIA Report 82-111.
Holmes, *Coherent Spread Spectrum Systems*, 1982, Wiley Interscience, New York.
Kesteloot, Ed., *The ARRL Spread Spectrum Sourcebook* (Newington, CT: ARRL, 1990). Includes Hershey, *QST* and *QEX* material listed separately here.

The *AMRAD Newsletter* carries a monthly column on spread spectrum and reviews ongoing AMRAD experiments. Contact information appears in the Address List in the **References** chapter.

The following articles have appeared in Amateur Radio publications. All of the articles from *QST* and *QEX* are reproduced in *The ARRL Spread Spectrum Sourcebook*.

Feinstein, "Spread Spectrum—A report from AMRAD," *73*, November 1981.
Feinstein, "Amateur Spread Spectrum Experiments," *CQ*, July 1982.
Kesteloot, "Practical Spread Spectrum: A Simple Clock Synchronization Scheme," *QEX*, October 1986.
Kesteloot, "Experimenting with Direct Sequence Spread Spectrum," *QEX*, December 1986.
Kesteloot, "Extracting Stable Clock Signals from AM Broadcast Carriers for Amateur Spread Spectrum Applications," *QEX*, October 1987.
Kesteloot, "Practical Spread Spectrum Achieving Synchronization with the Slip-Pulse Generator," *QEX*, May 1988.
Kesteloot, "A Practical Direct Sequence Spread-Spectrum UHF Link," *QST*, May 1989.
Kesteloot, "Practical Spread Spectrum Clock Recovery With the Synchronous Oscillator," *QEX*, June 1989.
Rohde, "Digital HF Radio A Sampling of Techniques," *Ham Radio*, April 1985.
Rinaldo, "Spread Spectrum and the Radio Amateur," *QST*, November 1980.
Sabin, "Spread Spectrum Applications in Amateur Radio," *QST*, July 1983.
Williams, "A Digital Frequency Synthesizer," *QST*, April 1984.
Williams, "A Microprocessor Controller for the Digital Frequency Synthesizer," *QST*, February 1985.