Chapter 9

Modes and Modulation Sources

The various modes we use partly reflect the different types of information we might wish to transmit. These could include text, graphics, speech (but not music, which is prohibited in Amateur Radio communications) and control messages—that might be used to steer a model plane or turn on a remote piece of equipment. Various levels of quality or fidelity might be desired for each of the above information types depending on the use for which the information was intended.

The communications *Channel* is as important in determining what mode to use as is the message content. With radio communication, we are limited to what exists in nature. The spectrum is a limited resource, putting a restriction on the bandwidth we can occupy. Attenuation is also present in any radio link and this dictates factors such as power and receiver sensitivity. Variations in the propagation path produce fading, delay, frequency shift (Doppler) and distortion. Finally, noise of various types may be present. While all of the above problems affect all our available frequency assignments to some extent, there are wide variations. Some of the bands used by amateurs are wide and well behaved, such as short VHF links. Others may be narrow, unstable and hostile to our signals, such as a long HF path through the auroral zones. Such conditions dictate which mode will be most successful.

ISSUES COMMON TO ALL TRANSMISSION MODES

Bandwidth is the amount of space that a signal occupies. There are narrowband modes, such as CW and PSK31, and wideband modes such as TV and spread spectrum. Not all modes are legal on all amateur bands. Wideband modes can only be used where the total width of the amateur assignment is sufficient to contain the wide signal. In addition, gentlemen's agreements and legal restrictions keep some wideband modes out of certain bands or subbands so that one person's signal does not preclude operation by a large number of others using the narrower modes. All users of the radio spectrum must comply with FCC *occupied bandwidth* rules. The occupied bandwidth is determined not only by the mode being used, but by proper operation of that mode. Many of the legal modes can become too wide when improperly adjusted. Perhaps the greatest source of conflict between ham operators is the "splatter" caused by over-modulated or otherwise improperly operated equipment, regardless of the mode being used. An amateur signal must be no wider than is necessary for good communication, and as clean as the "state of the art" will allow. Section 97.3 (a)(8) of the FCC rules defines occupied bandwidth as the point where spurious energy drops to 26 dB below the total power.

Sensitivity refers to the relative ability of a mode to decode weak signals. Some modes are favored by DXers in that they have a greater ability to "get through" when the signals are very weak. For local communications, sensitivity may not be the major concern. *Fidelity* is not often a major issue for amateurs, although many hams rightly take pride in the clear sound of their transmissions. *Intelligibility* is related to fidelity in a complex way and, sometimes, voice signals are modified in such a way as to make them more understandable, perhaps under difficult conditions, even though not as natural as they might otherwise be. *Quality* is the corresponding term for images, and *accuracy* describes the degree to which a text mode reproduces the original message. *Robustness* or *Reliability* refers to the ability of a mode to maintain continuous communication under changing conditions. For example, a very robust signal is desired when controlling a model airplane. DXers are not overly concerned with reliability in that continuous contact is not needed. However, they do want a signal that gets through when needed to work a rare station.

Efficiency is the ability of a mode to get the signal through with minimum energy expended. Within the legal power limit, energy cost is not a major concern for most home stations. Thus, efficiency is a concern mainly to those on battery power—using handheld or portable stations. Emergency operators also need to consider using efficient modes. For radio services which use high power, such as shortwave broadcasters, efficiency is very important. QRP is a popular activity, where operators take pride in making contact with a very small amount of transmitter power (maximum miles per milliwatt!) *Stability* is the ability to maintain the frequency of the transmission very precisely. Some modes require precise frequency control. Most modern equipment is very stable, but some vintage or homemade gear may be limited in frequency stability. Higher frequency work can put tight limits on frequency stability. Channel stability refers to both frequency, amplitude and phase variations of the transmission medium itself. The inherent instability of a radio channel may permit some modes but preclude others. *Noise immunity* is the ability of a radio system to reject noise of various types that could otherwise destroy the meaning or impair the quality of the message. This is all important in HF mobile operations, and for those living in densely-populated areas. Man-made electrical noise is an increasingly serious threat to ham operations and requires both legal and technical solutions.

Voice Modes

AMPLITUDE MODULATION (AM)

AM is as old as radio. The first wireless signals transmitted by Marconi across his garden consisted of trains of pulses that were excited by a spark which recurred at a more or less audio rate. This excited radio waves in a band of frequencies determined by the broad resonance between a coil and the capacity of the antenna. The rough buzz of early spark transmitters could be received with diodes made of various substances that provided envelope detection, which recovers audio that follows the averaged peak amplitude of the wave.

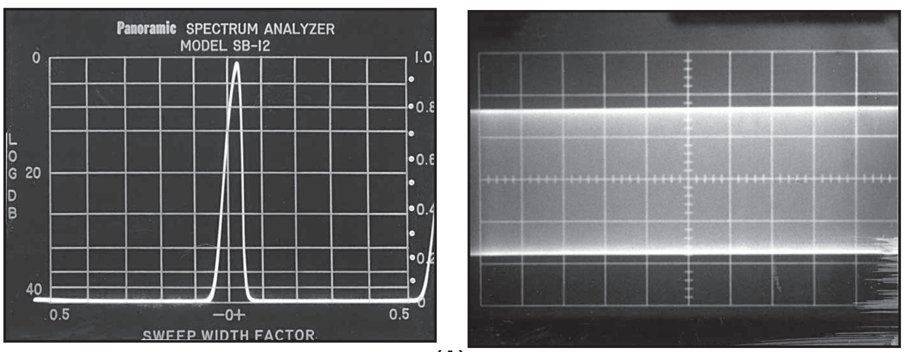
Spark transmitters have passed into history, but amplitude modulation of radio waves has not. Though likely to be replaced by digital speech modes in the not too distant future, classical full carrier AM is, at present, still used by the world's most powerful radio stations which broadcast on long, medium and shortwave frequencies to huge audiences. This material was written by John O. Stanley, K4ERO.

The first AM broadcast of speech and music occurred nearly a century ago, when on Christmas eve of 1906, Fessenden, using a modulated high frequency alternator, surprised ship operators with a program of music, Bible readings and poetry. The development of a continuous wave transmitter, one that produced a constant sine wave output, rather than the rough spark signal, made AM practical. Thus, CW, as this pure wave was called, not only greatly enhanced Morse communications, but allowed voice transmissions as well. By changing the strength or amplitude of this smooth continuous wave, a voice could be superimposed on or "carried" by the radio frequency.

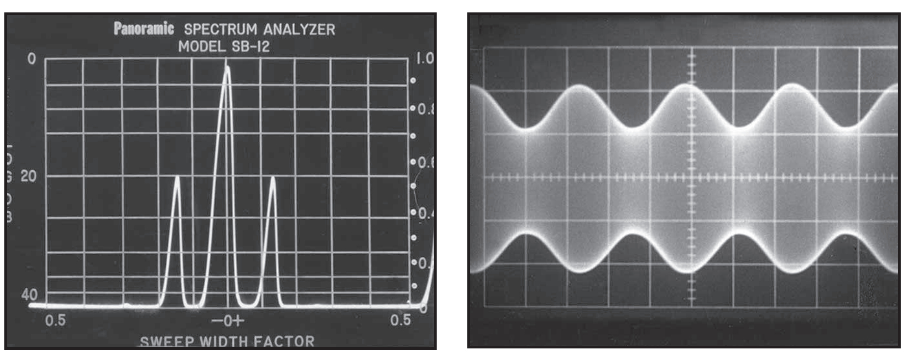
The decade of the 1920s saw not only the rapid development of the broadcast industry, but also enabled many hams to try the new voice mode. Indeed, in those early years, there was sometimes little difference between a ham who used voice and a broadcaster. The legal situation was a mess, and QRM was king! By 1929 it was legally and technically possible to use AM voice in limited portions of our amateur spectrum and, on some bands, only the most qualified licensees had the privilege. Users of AM had to learn that an RF wave could only have a certain amount of audio imposed upon it before overmodulation occurred. Trying to go above 100% modulation produced severe distortion and QRM. AM remained the dominant voice mode for ham operations well into the second half of the 20th century, when it was gradually eclipsed by SSB (SSB is actually a form of AM) and FM. We can still hear AM on the ham bands today, mostly coming from stations using vintage gear. AM'ers usually choose operating times when the bands are less crowded, and often take pride in a clean and clear signal.

The great advantage of AM, and one reason for its long history, is the ease with which a full carrier AM signal can be received. This was all important in broadcasting where, for every transmitter, there were thousands or millions of receivers. With modern integrated circuits, complex detectors now cost very little. Therefore, the biggest reason for keeping AM broadcasting, at present, is to avoid obsoleting the billions of existing receivers. These will gradually have to be replaced when digital broadcasts begin in the AM and shortwave bands.

There are many ways to produce an AM signal, but all of them involve multiplying the amplitude of the information to be transmitted by the amplitude of the radio wave which will carry it. When multiplication of two signals takes place, as opposed to their simple addition, mixing is involved. The result is multiple signals, including the sum and difference of the AF and RF frequencies. These two "products" will appear as sidebands alongside what was the original RF frequency. Mixing, modulation, detection, demodulation, and heterodyning all refer to this multiplication process and can all be analyzed by the same mathematical treatment. See the **Mixers, Modulators and Demodulators**



(A)



(B)

Fig 9.1 — 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.

chapter of this *Handbook* for a more detailed discussion of this process.

If an RF signal is modulated by a single audio tone, and observed on an oscilloscope, it will appear as shown on the right in **Fig 9.1B**. Observing the same signal on a spectrum analyzer will show that the composite signal observed on the scope is composed of three discrete parts as shown on the left in Fig 9.1B. The center peak, which is identical with the original unmodulated wave shown in Fig 9.1A, is usually called the carrier, although this terminology is deceiving and imprecise. It is the composite RF signal, as seen on the oscilloscope, which actually carries the audio in the form of variations in its amplitude, so we might well have referred to the center frequency as a "reference" or some other such term.

As a reference signal, the carrier contains important, though not indispensable, information. For a signal with both sidebands present, it provides a very important frequency and phase reference which allows simple and undistorted detection, using nothing more than a diode. The carrier also provides an amplitude reference, which is used by AM receivers to set the gain of the receiver, using AGC or automatic gain control. The carrier also contains most of the power of the transmitted signal, while most of the important information is in the sidebands. See the **Mixers, Modulators and Demodulators** chapter in this *Handbook* which gives details of power distribution in an AM signal.

Telephone engineers developed a system of using only one of the two sidebands which, being mirror images of each other, contain the same information. SSB systems attracted the attention of hams soon after WWII and gradually became the voice mode of choice for the HF bands. SSB is considered a form of AM, in that it is identical to an AM signal with one sideband, and all or part of the carrier removed. The complexity of generating a SSB signal, plus the difficulty of tuning the generally unstable receivers common in the 1950s, slowed the changeover to the new mode, but its adoption was inevitable. SSB became popular because of its greater power efficiency which allowed each watt of RF to go further. The fact that it occupied less bandwidth was a plus also and very welcome on the most crowded bands. See the sidebar **SSB on 20 and 75 Meters** in this chapter.

While systems used for telephone relays used pilot carriers so that the signal could be reproduced without distortion, hams chose to eliminate the carrier entirely. This required generating a reference frequency at the receiver which, if accurate to within 20 Hz, allowed intelligible speech to be recovered. Since amateur regulations have long prohibited transmission of music, the distortion produced by loss of the exact phase and frequency reference was not serious. The loss of the amplitude reference was overcome with the development of the "hang" AGC, which works on the average value of the received sideband which is constantly changing. While not as fast or accurate as the carrier-based AGC available in AM, this has proven satisfactory, if proper attention is given to its design (See the **Receivers and Transmitters** chapter of this *Handbook*.)

Thus, SSB, while giving up some fidelity and increasing complexity, has proven superior to full-carrier AM for speech communication because of its power and bandwidth efficiency. And under certain circumstances, such as selective fading, it can actually have less distortion than AM. On HF, it is possible for the carrier to fade in an AM signal leaving less than is needed for envelope detection. AM broadcasts often have this problem at night. It can be overcome with "exalted carrier detection." Synchronous detection is a refinement of this method. (See the **Mixers, Modulators and Demodulators** chapter of this *Handbook*.) SSB, in effect, uses exalted carrier detection all the time.

An SSB signal is best visualized as an audio or *baseband* signal that has simply been shifted upwards into the radio frequency spectrum as shown in **Fig 9.2**. The relative frequencies, phases and amplitudes of all the components will be the same as the original frequency components except for having had a fixed reference frequency added to them. Surprisingly, this process, called heterodyning, is not done by directly adding the signals together, but by multiplying them and subsequently filtering or phasing out the carrier and one of the sidebands. The **Mixers, Modulators and Demodulators** chapter of this *Handbook* explains this interesting process in detail.

The relative frequencies within the band of information being transmitted may appear inverted; that is, lower frequencies in the original audio signal are higher in the RF signal. When this happens, we call the signal lower sideband or LSB. LSB is produced when the final frequency is the result of subtraction rather than addition. If a tone of 1 kHz is heterodyned to 14201 kHz by mixing with a 14200 kHz carrier, the result will be upper sideband, since 14200 + 1 gives us that result. When the same tone appears at 3979 kHz by mixing it with a 3980 kHz carrier, we know that an LSB signal was produced since 3980 – 1 gives us the 3979 result. Whenever the audio tone needs a minus sign to find the result, we are on LSB. In most mixing schemes there will be three frequencies involved (carrier, VFO, and band select crystal) but the principle still holds.

The frequency of an SSB transmission is designated as that of the carrier, which is the frequency (or the sum of several frequencies) used to shift the baseband information into the RF spectrum. In a good SSB signal, little or no energy actually appears on the frequency we say we are using. It is strictly a reference. For this reason, some radio services have chosen to designate SSB channels by the center of the occupied bandwidth rather than the carrier frequency. Ham practice is to designate the carrier frequency and whether the upper or lower sideband is in use. An interesting exception is the new five-channel, 60-m amateur band (a secondary allocation) where the FCC specified a 2.8-kHz maximum BW centered on five frequency segments: 5332, 5348, 5368, 5373 and 5405 kHz. Only USB is permitted. Most

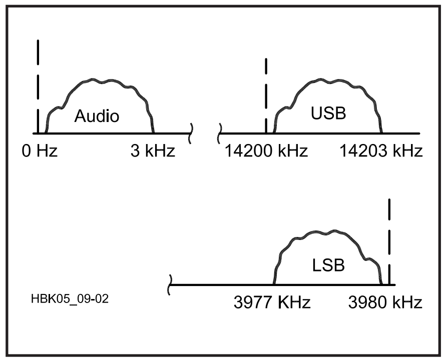


Fig 9.2 — How an occupied radio frequency spectrum shifts with application of an audio (baseband) signal. The dotted line represents the RF carrier point, or in the case of the 3-kHz audio signal, the reference frequency, 0 kHz.

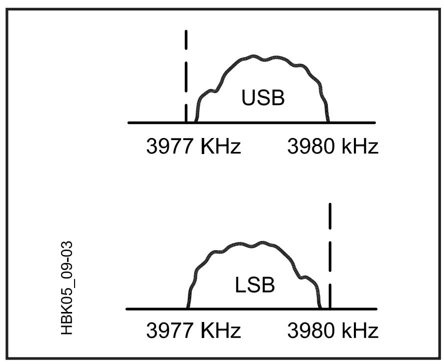


Fig 9.3 — A method of changing sidebands with virtually no change in the frequency spectrum occupied. Note that the carrier point position *has* changed concurrent with the change from LSB to USB.

hams will find it more natural to remember USB at corresponding carrier frequencies of 5330.5, 5346.5, 5366.5, 5371.5 and 5403.5 kHz. Since the USB or the LSB is considered "normal" for each of our bands, it is assumed that the sideband in use is understood. We need to remember when switching sidebands that we will be occupying a different portion of the spectrum than before the switch, and we may inadvertently cause QRM, unless we check for a clear frequency. If one wishes to change from LSB to USB without changing the spectrum occupied, we must retune our dial down about 3 kHz as a careful study of **Fig 9.3** should make clear. This principle applies to digital as well as voice modes, but usually not to CW, where modern rigs make the above adjustment for us. This means that the frequency readout with a CW signal will be the actual frequency occupied, but with analog voice and digital modes this will probably not be the case.

Another need for understanding where sideband signals actually fall is in operating close to the edge of a band or subband. For example, on 20 meters where USB is used, one must not operate above approximately 14.347 MHz, since the transmission will be outside the band if one operates much higher. Operation on exactly 4.0 MHz could be legal on LSB if the signal is very clean, but is not recommended. Most modern rigs which prevent out of band transmissions *do not* preclude the above cases of illegal operation.

Today there are many new modes for text, speech and image transmission, and more will be developed. Often these are transmitted using SSB. Knowing exactly where the signal will appear on the band depends on understanding how LSB and USB signals are produced. These modes use either a separate circuit or more recently a computer sound card to produce audio frequency tones which represent the information in coded form. This is then fed into the audio input of an SSB transmitter. They are then heterodyned to the desired amateur band for transmission. In a transceiver, the incoming signals are similarly heterodyned back to the audio range for processing in the computer sound card or other circuitry. Some computer–based digital modes allow reading the actual signal frequency off the screen, provided the transceiver dial is properly set.

Voice signals and some text and image modes require linear amplification. This means that the amplifiers in the transmitter must faithfully represent the amplitude as well as the frequency of the baseband signal. If they fail to do so, intermodulation distortion (IMD) products appear and the signal becomes much wider than it should be, producing interference (QRM) on nearby frequencies. CW and FM do not require a linear amplifier, but one can be used for them also, at a small price in efficiency. Some VHF "brick" amplifiers have a choice of either the more efficient class C amplification or the more linear class B amplification. The linear or SSB mode must be chosen if SSB voice and some digital modes are being used. Whenever linear amplification is needed, flattopping must be prevented. This results from overdriving the amplifier so that it goes above the design power limit and becomes non-linear at that point.

SSB transmitters and most linear amplifiers use automatic level control (ALC) to prevent overdrive and flattopping. However, there are limits to ALC and flattopping can still occur if the amplifier is grossly over driven. The surest way to create ill will on any band is to cause spatter by over driving your amplifier, regardless of the mode. Amplifiers suitable for both linear and non-linear signals are discussed in the **RF Power Amplifiers** chapter of this *Handbook*. The effects of non-linear amplification are also further treated in the **Mixers, Modulators and Demodulators** chapter herein.

HOW AN SSB SIGNAL IS PRODUCED

When the proper receiver bandwidth is used, an 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 is half that of a comparable AM (DSB) emission. Unlike DSB, the phase of the local carrier generated in the receiver is unimportant. **Table 9.1** 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 bandpass filter, one of the 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 supression. 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 9.4 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.

SSB Generation: The Phasing Method

Fig 9.5 shows another method to obtain an SSB signal. The audio and carrier signals are each split into equal components with a 90 degree phase difference (called *quadrature*) and applied to balanced modulators. When the DSB outputs of the modulators are combined, one sideband is

Table 9.2
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

Table 9.1
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

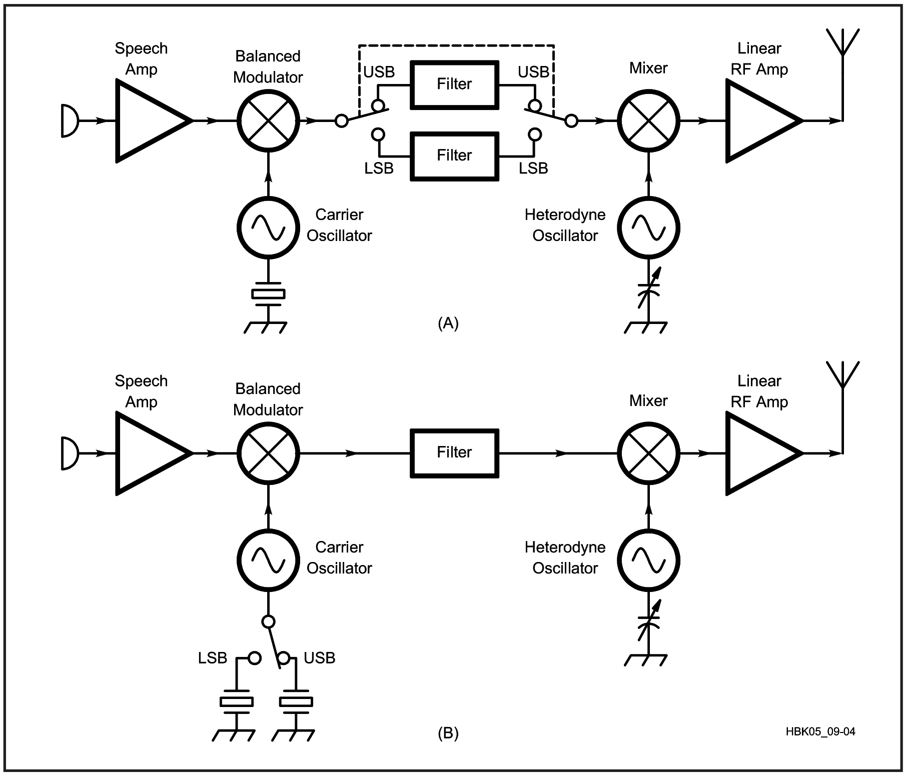


Fig 9.4 — Block diagrams of filter-method SSB generators. They differ in the manner that the upper and lower sideband are selected.

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 amplitude balance of the two channels must be very accurate if the unwanted sideband is to be adequately attenuated. **Table 9.2** 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 achieve unwanted sideband suppression of greater than 40 dB . 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 of 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.

SSB on 20 and 75 meters — the 9 to 5 connection

SSB experiments began on 75 meters because it was the lowest frequency phone band in widespread use. Due to perpetual crowding and its DX potential, 20 meters also seemed to call for use of SSB. Some early rigs included only these two bands. The popular homebrew W2EWL rig was built on the chassis of a war surplus ARC-5 transmitter using its 5 MHz VFO, and generated the sideband signal on 9 MHz using the phasing method. Nine plus five is 14 MHz, and nine minus five is 4 MHz, yielding 75 or 20 meter coverage by choosing which of the two mix products we would filter out and amplify. Thus, two bands were covered with the same VFO/IF combination. Other rigs used a tunable IF from 5.0 to 5.5 MHz. This was subtracted from a 9 MHz crystal to obtain 4.0 to 3.5 MHz, and added to 9 MHz to cover 14.0 to 14.5 MHz. This process reversed the sidebands, and eventually led to the convention of using LSB on the lower bands and USB on the higher bands. This also explains why on some vintage rigs the 75 meter band dial reads backwards!—*K4ERO*

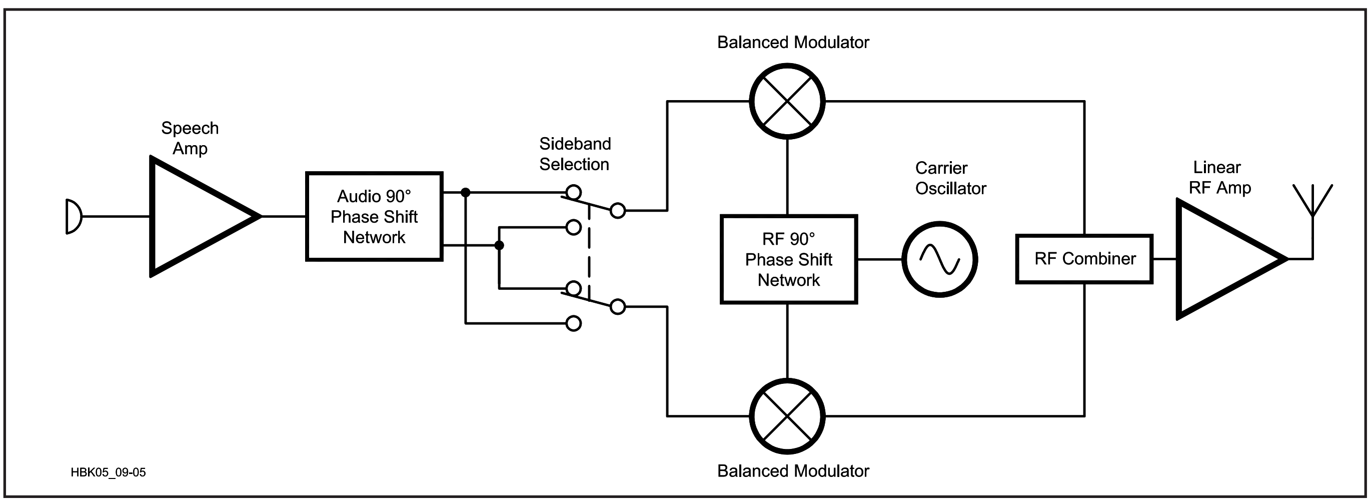


Fig 9.5 — Block diagram of a phasing SSB generator.

Audio Phasing Networks

Since the phasing method requires that all baseband signals be presented to the balanced modulators in both a normal (in phase) and quadrature (90° phase shifted) signal, we must provide, in the case of an audio signal, a network that can produce a constant 90° phase shift over a wide frequency range. Fortunately, the absolute phase shift is not as important as the relative phase between the two channels. Various circuits have been devised that will provide this relative shift. Robert Dome, W2WAM, pioneered a simple network using precision components that achieved this and his network was used in early SSB work. The polyphase network, which appeared in this *Handbook* for several editions, required more—but less precise—components. Methods using active filter techniques are also available.

With DSP, producing a 90° phase shift over a wide frequency range is easily accomplished using the Hilbert transformer. This will likely give new life to the phasing method of SSB generation since many new radios already have DSP capability present for other reasons. See the **Receivers and Transmitters** chapter of this *Handbook* for an example of an SSB receiver using DSP with the phasing method. See also the **Digital Signal Processing** chapter herein.

Producing 90° phase-shifted signals at RF frequencies has also used several approaches. For VHF and up, a quarter-wave section of coax is possible. Generating an RF signal at four times the desired frequency and dividing down with flip-flops generates quadrature signals accurate over a wide range of frequencies. Phase lock loops provide yet another approach.

The phasing method is useful not only for generating an SSB signal, but for any mixing or frequency conversion task. In phase and Quadrature (I&Q) modulators, demodulators and mixers are in common use in modern communication technology. These allow elimination of image frequencies without filters, or greatly relax the specification of filters that are used. Digital modulation can be generated in an I&Q format that can be directly heterodyned into the RF spectrum using I&Q modulators. The **Digital Signal Processing** chapter of this *Handbook* discusses many of these concepts.

FREQUENCY MODULATION (FM)

Unlike AM, which changes the amplitude of a radio wave in accordance with the strength of the modulation signal, FM changes the frequency of the wave so that the instantaneous value of frequency represents a voltage level in the modulating signal as is shown in **Fig 9.6**. This means that the demodulator must extract the information by generating an output whose amplitude is determined by the frequency of the received wave. Thus, FM transmission involves amplitude to frequency conversion and vice-versa. Producing these conversions was not as easy as it was in the case of AM and thus, FM was not employed as early as was AM. As you can see in **Fig 9.7**, the circuits required for FM were especially difficult in the case of the receiver. See also the AM- and Angle-Demodulation subsections of the **Mixers, Modulators and Demodulators** chapter in this *Handbook*. In addition, mathematical analysis seemed to show that FM would require a very large bandwidth (theoretically infinite), and this discouraged early experimenters.

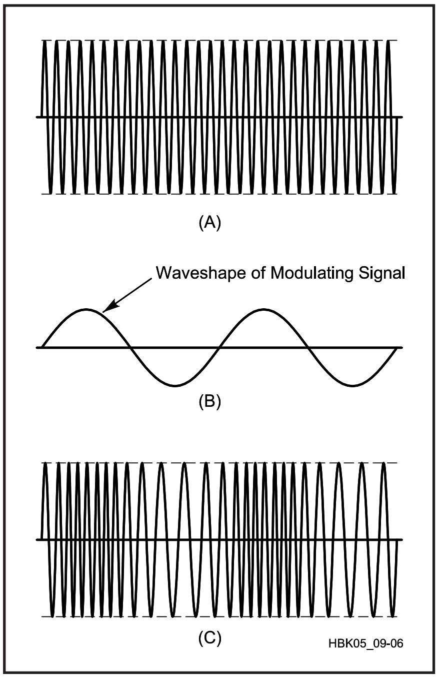


Fig 9.6 — Graphical representation of frequency modulation. In the unmodulated carrier (A) each RF cycle occupies the same amount of time. When the modulating signal (B) is applied, the radio frequency is increased and decreased according to the amplitude and polarity of the modulating signal (C).

Edwin Armstrong was a ham before the days of call signs. While a young man, he invented the regenerative, super-regenerative and superheterodyne receivers. He went on to challenge the prevailing wisdom and developed a practical FM system. His "Yankee Network" provided high fidelity broadcasts throughout the northeastern United States in the late 1930s, using frequencies below our 6-meter band. After WWII, FM was moved to 88-108 MHz and became FM broadcasting as we now know it. Dependable day and night reception was a result of the frequency chosen, not the mode, but wide band FM, which had dictated the use of a VHF frequency where bandwidth was available, provided the wide audio response, high signal to noise ratio, and freedom from static which AM could never have provided, even at VHF. The advantages of FM were proven even when bandwidths were less than infinity. The math had not been wrong, but had just been taken a bit too literally.

Hams experimented with narrowband FM (NBFM) on the HF bands during the 1950s, but nothing much came of it. The explosion in the use of FM in the amateur bands came after surplus commercial FM equipment, using frequencies near 150 MHz, became available in the 1960s and 1970s. Two meters was the first to use this equipment and is still the "workhorse" of the VHF FM bands. Hams, like the com-

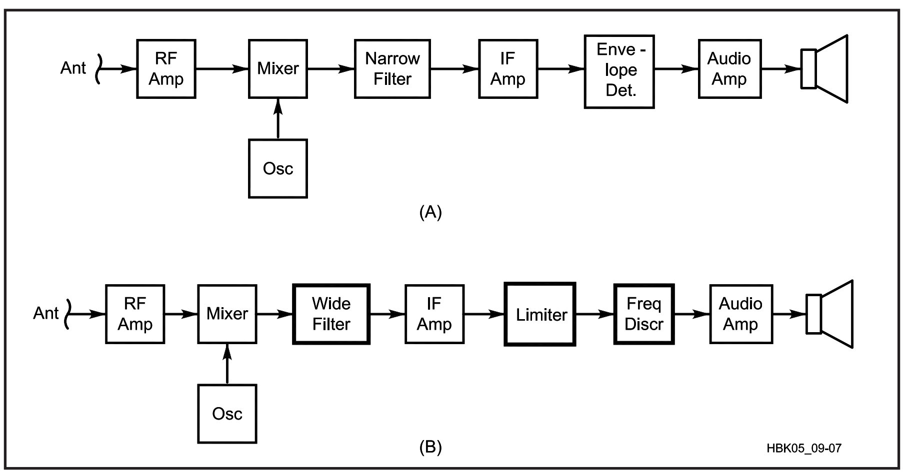


Fig 9.7 — At A, block diagram of an AM receiver. At B, an FM receiver. Dark borders outline the sections that are different in the FM set.

mercial and public service users before them, discovered that FM has certain advantages—less noise, ease of operation, no fussy tuning and suitability for use through repeaters.

A mathematical analysis of FM is complex, and well beyond the scope of this chapter. Readers who are interested in more details can consult the **Mixers, Modulators and Demodulators** chapter of this *Handbook*. Unlike AM, where the occupied bandwidth is simple to calculate (twice the highest modulating frequency), FM bandwidth depends on both the modulating frequency and the Deviation, which is equal to the peak frequency excursion above and below the central "carrier" frequency. As the math predicts, there are sidebands which extend to infinity but, fortunately, these drop off rather quickly and, as Armstrong surmised, ignoring those which contain only a tiny portion of the total energy does not impair the quality of the received signal.

As a rule of thumb, adequate bandwidth for an FM voice system using narrowband modulation (5 kHz or so) is Bn=2(M+D) where Bn is the necessary bandwidth in Hz, M is the maximum modulation frequency in Hertz, and D is the peak deviation in Hertz. For narrowband FM with voice, the bandwidth equals 2 × (3000+ 5000) = 16 kHz. This defines the filter through which the signal can be received without noticeable distortion.

Examples of FM spectra using various modulation indices are found in the **Mixers, Modulators and Demodulators** chapter of this *Handbook*. Note that as more and more sidebands appear, the amplitude of each is reduced. This is because all of the sidebands, plus the carrier, must add together (vectorially) to produce a total wave of constant amplitude, which is characteristic of the FM signal. This constant amplitude signal has the advantage of being easy to amplify without the need of a linear amplifier. Many VHF and UHF brick-type amplifiers have separate settings for FM and SSB. The FM setting is more efficient since, by giving up the requirement for linearity, we can bias the transistors for greater efficiency. Thus, an FM amplifier is easier to build than one suitable for AM or SSB.

However, this constant amplitude characteristic of FM comes at a price. The full power is being transmitted, even between words or when one is holding down the push to talk, but not actually speaking. Thus, for normal speech, the power advantage which FM gains by amplifier efficiency is lost compared to SSB, where power is only transmitted when the voice requires it. One should not, however, conclude that the unmodulated FM signal serves no purpose. Its presence "quiets" the channel, opens the squelch of the receiver(s), and turns on any repeater(s) that might be in the circuit. There may also be various control tones (squelch, etc.) present, even though these may be inaudible due to being in a frequency range that the human ear does not easily perceive.

Using FM and PM with Digital Modes

Frequency Shift Keying (FSK) is a means of producing frequency modulation that has discrete states; that is, the instantaneous frequency takes on definite values representing digital information. FSK is a form of FM and some of the same principles apply. FSK is covered further in the section on RTTY and other digital modes.

Phase modulation (PM) is very similar to FM in that it is not possible to change the frequency of a signal without impacting its phase and vice versa. Instantaneous frequency can be considered to be the rate of change of phase of a signal. Some FM modulators have used this relationship to produce FM by phase modulation along with audio frequency shaping to convert the PM signal into the equivalent of an FM signal. This issue is discussed further in the **Mixers, Modulators and Demodulators** chapter of this *Handbook*. Phase shift keying (PSK) is a form of phase modulation suitable for digital transmissions. It is discussed further in the following pages of this chapter. Both FSK and PSK produce sidebands in accordance with the same principles discussed above. However, in order to control bandwidth, digital signals using PSK may depart from the requirement that an FM signal have a constant amplitude. Such signals are really a combination of FM and AM, and linear amplification must be used.

BESSEL FUNCTIONS

Bessel functions are employed—using the carrier null method—to set deviation. Some version of the chart shown in **Fig 9.8** has appeared in the *ARRL Handbook* for 50 years. This chart is unlike previous ones in that the values are plotted in dB, which is more familiar to anyone who uses an S meter or a spectrum analyzer to observe the various FM sidebands. This version also plots all values as positive because receivers, including spectrum

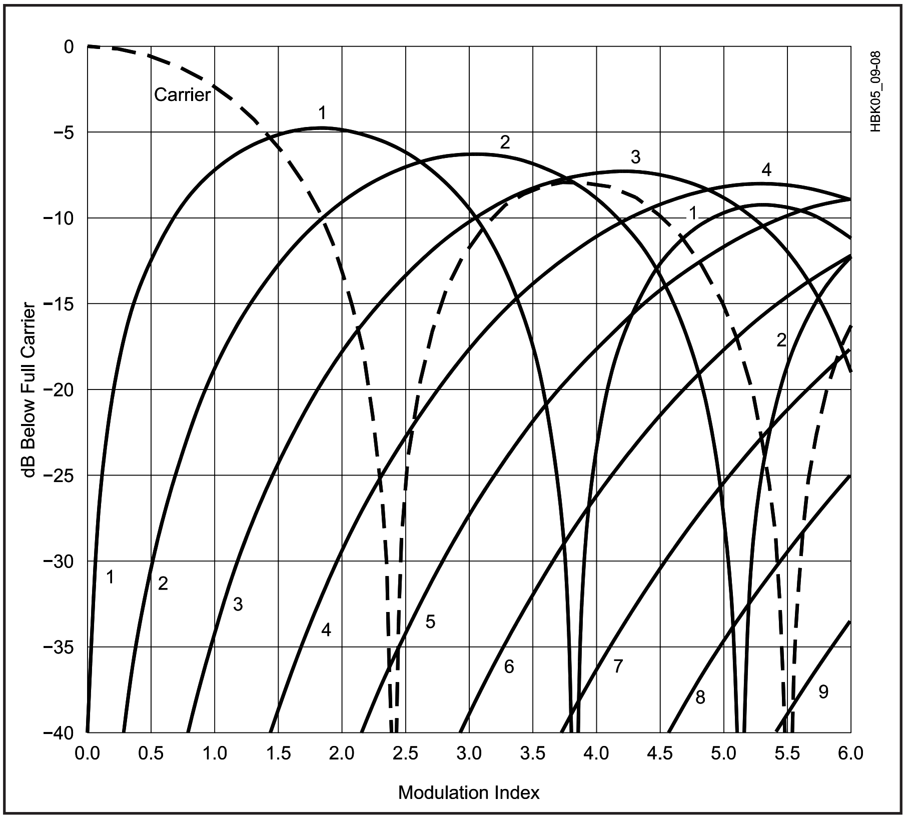


Fig 9.8 — Amplitude of the FM carrier and sidebands with modulation index. This is a graphical representation of mathematical functions developed by F. W. Bessel. Note that the carrier completely disappears at modulation indexes of 2.405 and 5.52.

Effect of the HF Path on Pulses

Digital signals, including Morse signals copied by ear, are a type of pulse transmission. That means that something about them changes abruptly, and the different states in which they exist carry the digital information. Understanding what happens to pulses of all types when they travel via the ionosphere is important for knowing how to design a workable digital system for use in HF. VHF and UHF signals generally use a more benign path but some of the same principles important in HF will also apply to them. However, it is not without reason that HF will probably be the last slice of the spectrum where analog methods will be fully replaced by digital.

Since Morse CW using OOK (on-off keying) is the oldest as well as the simplest digital mode, it is useful to analyze the propagation effects on a simple Morse character, namely the letter E. This consists of a single pulse, which can be distinguished from the T by its length provided the keying speed is known beforehand. Using the ARRL-recommended values for shaping so as to prevent excessive key clicks, a single Morse "E" will appear as shown on the left in **Fig A** as it leaves the transmitter. A ham living nearby would observe the pulse essentially unchanged and could comment usefully on the shape of the keying. However, a receiver far enough removed so that sky wave was the dominant mode would see a much different picture. If both ground wave and sky wave were present, as could occur on 80 meters at a distance of 20 miles, the received dit might appear as shown on the right in Fig A. The pulse is distorted by several things. One is multipath, meaning that the signal arrives by more than one route. One route might be the ground wave, another the single hop sky wave, and others, very likely considerably weaker, multiple-hop sky waves. In addition, if the ionosphere is moving up or down, Doppler shifts will change the received frequency slightly. If operation is near the MUF, some energy may be greatly delayed by reflections from varying heights, further smearing the pulse. Noise of various types will be added to the pulse. Finally, fading effects may be noticed even on such a short time frame as a single Morse letter.

The successful decoding of a Morse signal consists of nothing more than deciding if a pulse was sent or not, and its length. Trained humans have been doing this rather well for over a hundred years and may not appreciate the difficulty of teaching a machine to do the same. For machine decoding, a decision threshold must be established which effectively says, "if energy rises above this value, the signal is present; if below it, it is not present". This works well with the clean original signal, but with the one that has been degraded by the transmission path, it runs into problems as shown on the right in Fig A. If the threshold is set too high, multiple pulses may be detected, or nothing at all. If too low, noise is detected as signal. Even when set just right, the length of the detected pulse may be changed. Thus, the letter E might be decoded as an I, a T, or a space. Furthermore, noise pulses or QRM on nearby frequencies, may also be decoded as various letters. There is no error-correction scheme with machine-copied CW so many errors may appear in the final copy.

Humans have been given a brain and sense organs that, when properly trained, can cope with all these problems to a high degree. Our brain's ability to filter out extraneous noise and signals is remarkable. The dynamic range and sensitivity of both vision and hearing are close to the theoretical limits. AGC operates to provide a continually variable "decision threshold" and many a ham has passed his Morse exam by using near real time error correction, also known as guessing. If all else fails during on-air contacts, repeats are requested. They're not allowed in FCC exams however!!

A satisfactory non-human Morse decoder would have to replicate all of the above human functions. As it turns out, Morse, using OOK, is one of the hardest digital systems to copy error free, although by using fixed speed, and with reasonably good propagation, it is possible. Those who have worked with software or hardware schemes for decoding Morse will point out that narrow filtering will clean up the above pulse considerably. This is really a form of averaging or integration, which smooths out the abrupt changes in amplitude. A phase-locked loop detector provides averaging and threshold detection in one circuit. Even with these improvements, OOK signals are hard to decode reliably.

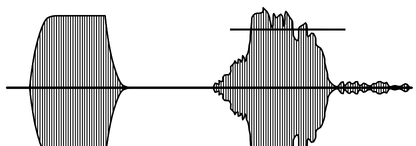


Fig A

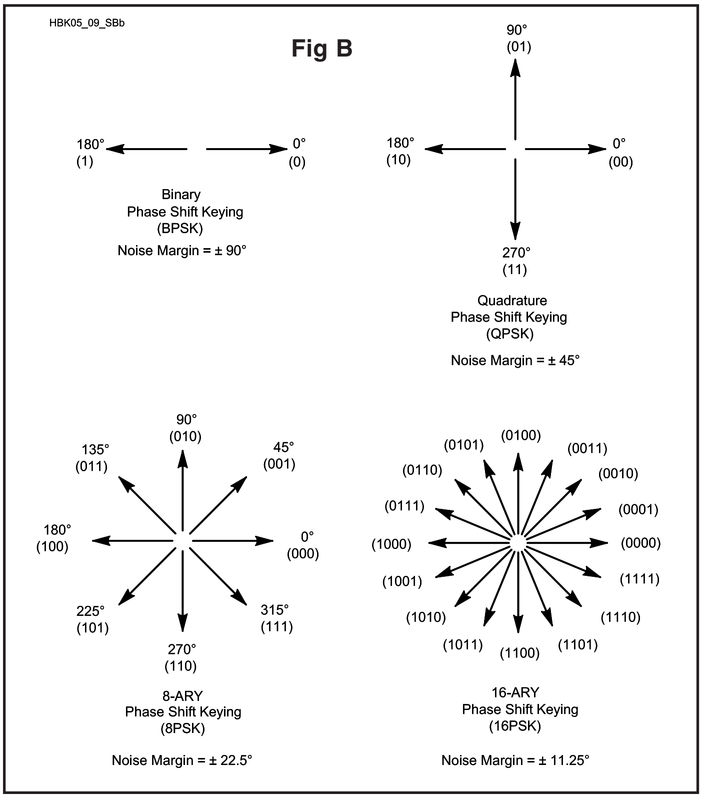
Mechanical recording of Morse signals, as opposed to mechanical decoding, is very old. In fact, it was commonly used in the earliest wireless telegraph system. A paper tape was drawn through an inker which made a mark proportional to the signal received. A human would then read the code from the long and short marks. This was the first "fuzzy mode"; a term now used to refer to facsimile systems that transmit text which is then read by a human operator. Such systems draw on the human capacity for visual pattern recognition which, in many applications, far surpasses the most powerful computers.

The second oldest method of transmitting pulses was with frequency shift keying (FSK) where, instead of turning a signal on and off, it was made to jump to a different frequency to correspond to the "key up" and "key down" conditions. Morse was sent by FSK in the early days as a way of keeping the transmitter (alternator, spark gap, etc.) loaded even during the key-up periods, an aid to stability. The key-down signal was listened to by ear and the key-up signal simply ignored. Mechanical recorders might use both signals, however.

RTTY does not require the use of FSK and some have used OOK for RTTY. However, because of long use, most of us associate FSK transmissions with RTTY; the mechanical or computer decoding of text (and crude images) using the Baudot code. Digital text modes have used AM (of which OOK is an example, but not the only one) FSK, or PSK (phase shift keying). As is discussed in this chapter's treatment of FM, phase keying and frequency keying are somewhat similar, since the frequency of a wave cannot be changed without changing the phase as well. As with FM analog transmissions, FSK and PSK tend to discriminate against amplitude noise, which is the most common type on HF. Thus, amplitude changes due to fading and/or superimposed static can be reduced. However, as discussed in this *Handbook's* **AC/RF Sources** chapter, AM noise can change into phase noise, so this is not a complete solution. Modes based on phase modulation, such as the PSK modes, are not noise free. One need only observe the little phase "compass" included in some PSK31 software to be aware that even when transmitting the idle tone, the received signal's phase will be seen to be jumping around.

Doppler shifts introduce noise into phase and frequency shift systems. If we are counting on seeing a certain frequency shift as a signal, we don't want the path adding any shifts that will show up as noise. Thus, Doppler shift places a lower limit on the amount of frequency or phase shift needed to distinguish between the various data bits. On HF, ionospheric Doppler, which can be up to 5 Hz, places a lower limit on the carrier spacing of multi carrier modes. Lengthening of pulses of up to several milliseconds is common on HF paths. About the only solution for channel-produced lengthening of pulses is simply to use a slower rate of transmission. Thus, the pulse has time to settle before being interpreted.

The fastest CW operators slow down when rare multipath conditions smear the characters together. A machine can operate at speeds where even normal multipath can smear the characters, not just extreme cases. This factor puts an upper limit on the symbol rate. If a bit rate is to be increased beyond this point, each pulse must carry more than one bit. This calls for the use of complex modulation schemes involving multiple states per pulse. These can be represented by a number of different frequencies in an FSK mode, many phases in a PSK mode, many levels in an AM mode or a combination of several or even all of the above. An example of multi-level phase modulation is illustrated in **Fig B**. A complex system would be QAM, which requires special equipment to observe.



With complex modulation, noise–induced changes in both amplitude and phase will cause some pulses to fall into oblivion where one simply cannot say what was sent. Thus, a higher signal to noise ratio is needed for complex modulation such as 64QAM, whereas BPSK may work down into the noise.

For very high bit rates on HF, many carriers may be used, each one carrying multiple amplitude and phase levels. As shown in **Fig 9.25**, Clover is a good example of a multi-carrier system with complex modulation. Though the path limits the symbol rate, there is no theoretical limit to the bit rate on radio, either HF or higher, provided spectrum is available to carry the ever wider signals. Of course, there are legal and practical limits. Thus, while it is feasible to transmit digital speech in a 3-kHz BW, full-motion, high-resolution digital TV signals would be very difficult to transmit on HF, even if an entire amateur band were used. Digital slow scan TV is, of course, another matter requiring a much lower bit rate than fast scan TV.

Schemes that use more than one frequency, such as FSK and multicarrier systems, must cope with *selective fading*, where not all frequencies fade at the same time. Users of RTTY FSK systems have long observed that either the mark or space frequency may momentarily fade away leaving only the other. A good decoder will work with only one of the two tones present. Multicarrier systems using *OFDM* may need some kind of periodic calibration pulses to detect the fading of each individual carrier frequency. It is not enough to keep the composite signal constant, which could be done with AGC. One approach is to look for changes between one pulse and the next, rather than the absolute value of either phase or amplitude.

All present and future digital modes must address the above problems, and the development of new text modes has seen a steady progression from the original and hallowed CW and RTTY modes. As digital voice, image, text and control modes develop, they will all cope with the above channel limitations in various ways and with varying degrees of success. Some will be very resistant to QRM; others will be efficient in use of spectrum or in *throughput*—the amount of data that can be sent in a given time. Modes that are optimum for a QRM-free VHF channel may not work at all on HF. Optimum HF modes will be too slow for some VHF applications. Error correction schemes will always be useful for the more difficult channels, since on HF there is no such thing as a 100% reliable channel. On all frequencies, hams have a habit of "pushing the limits" so that marginal paths will always be used. The design of digital communications systems will always be an exercise in trade-offs. For that reason, and the varied nature of both our paths and our purposes in communicating, there will never be one universal digital mode. We must be prepared to select the mode most suited to our present goal, while expecting that new and exciting digital modes will appear all the time.

analyzers, do not distinguish between positive and negative phase values. Thus, this plot will give values directly in dB below the unmodulated carrier of each component of a frequency modulated wave, based on the modulation index.

Since the carrier and each sideband of a frequency–modulated signal change amplitude according to fixed rules as the deviation and modulating frequency change, we can use those rules to set deviation, provided we have a way to observe the FM spectrum. Based on a set of mathematical functions named after F.W. Bessel, who developed them, we know that a modulation index of 2.405 will produce what is called the "first carrier null". Thus, if we wish to set our deviation to 5 kHz, we can use an audio tone of 5000/2.405 or 2079 Hz. While observing the spectrum, we can then increase the deviation from zero until the carrier is in a null. This guarantees that the deviation is now 5000 Hz. If we use a frequency counter to set the audio tone accuracy, the exactness of the deviation setting should be very high. Similarly, for setting the deviation to 3 kHz, we could use the audio frequency of 1247 kHz and adjust for the first carrier null. If a spectrum analyzer is not available, an all–modes receiver using a narrow CW filter could be used to detect the carrier null, using the S meter and carefully tuning on the carrier. Additional carrier nulls occur with modulation indices of 5.52 and 8.654. These would be useful for wideband FM or when using low audio frequencies for setting deviation with narrowband FM.

Other methods of setting deviation include observing the bandwidth on a spectrum analyzer using a very low frequency audio tone and using a deviation meter—an FM receiver whose audio output is metered on a scale calibrated directly in kHz of deviation. An FM service monitor may include both a deviation meter and a spectrum display. The carrier null method is the most accurate of the three methods and can be used to calibrate a deviation meter.

You can also use this plot to predict the bandwidth of any given audio frequency and deviation combination. Consider this example. We wish to keep our bandwidth narrow enough to pass through a 15–kHz receive filter and we are transmitting a tone of 3 kHz. Since the third set of sidebands will be 18 kHz apart, we would do well to keep them, and all higher sidebands, below –40 dB. A quick look at the chart shows that this means the modulation index must be no more than about 0.7, meaning the deviation should be $0.7 \times 3 = 2.1$ kHz. If we are willing to allow the third set of sidebands to be only 34 dB down, we can use a modulation index of 1, meaning the deviation will be 3 kHz—close to the recommended value for 9600 bits/s digital signals on FM. The above calculations strictly apply only when the highest audio frequency (3 kHz) is present. If there is little chance of 100% modulation at the highest audio frequency as, for example, with a normal voice signal, higher deviation could be used. When new digital modems and modes are used on FM, the above procedures should be part of the design.

DIGITAL VOICE MODES

There is risk in saying anything about an area in which new developments are occurring monthly, if not weekly. However, interest in amateur digital voice has developed to the point where it cannot be ignored. In 2000, the ARRL Board of Directors approved the creation of a Digital Voice Working Group to investigate and promote digital voice in the Amateur Service.

As was the case with SSB, the adoption of digital voice will take some time. And as with SSB, we have examples to follow in the commercial field. Cell phones have already largely made the transition to digital, thus proving that at least at UHF frequencies, it is practical. Satellite and terrestrial digital radio systems are becoming common in many countries for entertainment and information services. These also use VHF and higher frequencies.

On HF, where problems of digital transmission are much more difficult, Digital Radio Mondiale (DRM) has led the way. DRM is a non-profit consortium of broadcasters, manufacturers, educational and governmental organizations devoted to developing a single standard for digital broadcasting in long, medium and short-wave bands. In April, 2001, the ITU approved their system, also called DRM, but going by the technical name ITU-R BS.1514. This document defines the standards to be used worldwide for HF digital broadcasting. As of 2003, hundreds of *software* radios were being used to hear regular DRM broadcasts on HF from several countries. A modified HF receiver works through a computer sound card using software available from DRM. Details can be found at **www.drmrx.org**. Digital AM broadcasts also began in the United States during 2003. See *QST*, March 2003, p 28.

The DRM standard has several modes, some aimed at high fidelity music and others suitable for voice only. The broadcaster can select the most appropriate mode, and the receiver will switch automatically to that mode. The various DRM modes occupy 4.5, 5, 9, 10 or 20 kHz according to the spectrum available and the program quality desired. See **Fig 9.9**. The Jan-Feb 2003 issue of *QEX* (p. 49) describes variations of the DRM system that are contenders for an Amateur Radio standard for digital voice. Tests by hams on HF paths have shown promise. These standards use many carriers spaced about 50 Hz apart, filling up a 3-kHz spectrum. This is called Coded Orthogonal Frequency-Division Multiplex (COFDM), and many digital systems use it. In order to achieve a bit rate sufficient for acceptable quality voice reproduction, each carrier will use 16QAM (Quadrature Amplitude Modulation with 16 discrete stages in each pulse) or some other dense modulation scheme. For use in the presence of multipath, the symbol rate

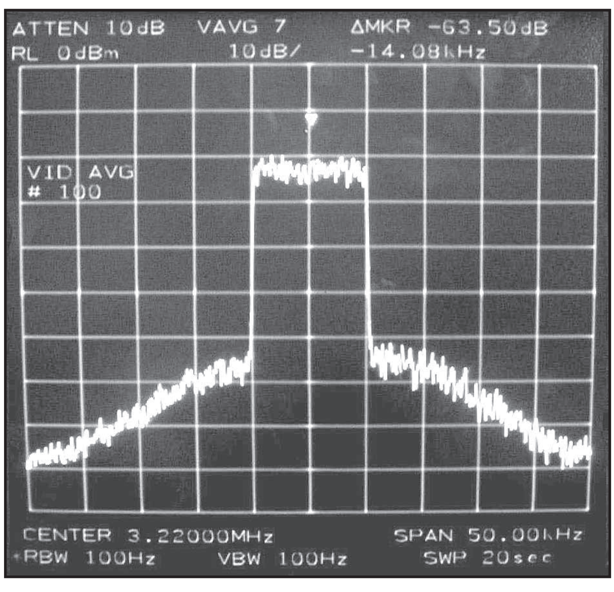


Fig 9.9 — A DRM HF digital broadcast signal. Per-division resolution is 5 kHz horizontal and 10 dB vertical.

must be limited to a few hundred baud. Thus, the high bit rate needed for voice requires both multiple carriers and complex modulation. The most obvious way to produce such a signal is to use a computer sound card as is presently done with digital text modes. The availability of a suitable computer program will do much to increase activity using digital voice modes. There will be trade-offs between complexity, weak signal sensitivity, reliability under difficult conditions, and speech quality.

Trade-offs are not new to ham operations. We have learned to use less than a 2 kHz audio bandwidth on SSB when the band is very crowded. We may also learn to copy and even enjoy digital voice modes, whose quality would be unacceptable in broadcast service—if it gets through when a higher quality mode would not. As an extreme example of that, we could use a voice recognition system to generate text, transmit that text with something like PSK31 and then, use a voice synthesizer to read off the message. A few additional bits might be dedicated to indicating the approximate pitch or other characteristics of the original voice. This would take some getting used to, in that everyone would tend to sound alike, and the personality of the speaker would not come through. This is not unlike what happened when bugs and hand keys were replaced by automatic keyers and keyboards, which some found upsetting, as they could no longer recognize their friends by their "fist" or "swing". And, just as an electronic keyer could be adopted independently of what the other operator did, this approach would not require that both sides of the QSO be digitizing the voice signal. One station could be using a keyboard and screen, the other a microphone and speaker.

Somewhere between the 4.5-kHz, high-quality DRM voice mode and the above depersonalized but spectrum efficient approach, there is a compromise solution that will eventually become the HF standard for amateur digital voice. There is much room for innovation and experiment. A great deal of work will go into developing the digital voice modes that we will be using 10 years from now. Those interested in being a part of this exciting technology should begin by mastering the material in the **Electrical Signals and Components** and **DSP** chapters of this *Handbook*, and keeping up with *QST* material on digital speech.

If it is so difficult, why bother? For ham operation, digital voice modes have the potential to use less bandwidth and provide better quality, greater reliability or greater sensitivity than SSB, their chief competition on HF. The digital approach could allow selection as to which of the above advantages would be emphasized in a given QSO, depending on the path and the desires of the operators. The equipment could adapt automatically to changing conditions, providing intelligible but perhaps a bit "robotic" sounding speech under marginal conditions, but reverting to very high-quality "arm chair copy" when conditions permitted. The present system of signal reporting will need changing but, in any case, that will be handled by the software. The digital radio of the future may have both an S meter and a Q meter on its video display.

Another advantage of digital transmission is that some ancillary data can be imbedded in the data stream without difficulty. Thus, the receiver will be able to display both calls, the other station's brag screen, your signal report etc, in text format, even while you are listening to the voice. The distinction between text and voice modes may become a bit blurred. This could be a huge advantage in working hams in foreign countries whose English skills may be limited. Obviously, log sheets could be generated without operator intervention. Super weak signal modes such as moon bounce and VLF operation may benefit from a voice mode that does not operate in real time. Slow data rates could provide a voice signal that would be announced when ready. The receiver could grind away for some minutes or even hours before you heard the answer.

For VHF, a whole different set of rules apply. Where strong, stable signals and bandwidth are available, all of the other advantages could be had simultaneously. It is not without reason that commercial VHF communications have gone, or are going, digital. Standards will have to be developed and only experience will show what mode or modes are best for a given use. Digital voice on VHF will almost certainly be different from what is used on HF, although for weak signal work and certainly for skip, the HF mode might be best even for VHF.

Digital voice on the VHF bands has made more progress than on HF as might be expected. Some manufacturers are beginning to include digital voice options in their equipment, and this trend will continue. To keep pace with rapidly changing digital voice technology and use, check both *QST* and *QEX* regularly, and also check the following web sites: *ARRL*Web at **www.arrl.org/tis/info/digivoice.html** and **www.arrl.org/announce**; **www.doug-smith.net**; **www.temple.edu/k3tu/digital_voice.htm** and **www.DRM.org**.

Text Modes

MORSE TELEGRAPHY (CW)

Text messages sent by On-Off-Keying (OOK) is the original mode for both amateur and commercial radio. It is alive and well today and is not expected to fade away. For many amateurs, it is the principal, or even the only, mode used and many take great and justifiable pride in their use of it.

CW continues in use, however, not just for reasons of nostalgia. When coupled with an experienced operator, it can rival most any mode for "getting through" under marginal conditions and is absolutely unrivaled in terms of the simplicity of the equipment needed. Methods of generating the code characters, and even of decoding them, have used the latest technology, but the straight key and "copy by ear" are still in use. Of all modes, CW is the most versatile in terms of signaling speed. It is used at speeds—measured in words per minute (WPM)—of less than one, and up to several hundred. Direct human copy works well between about 5 and 60 WPM, and for very slow or very fast speeds, the signals may be recorded and the speed adjusted to allow human decoding. Very slow speeds and extremely narrow filters make possible communication using signals well below the noise while very fast speeds are useful for meteor scatter communication where bandwidth is large, but the reflection path lasts a second or less.

The bandwidth occupied by a CW signal depends on the keying rate (See the **Mixers, Modulators and Demodulators** chapter of this *Handbook*), with higher speeds requiring a wider filter to pass the sidebands. In addition, occupied bandwidth depends on the rise and fall time and the shape of the keyed RF envelope. That shape should be somewhat rounded (no abrupt transitions) in order to prevent "key clicks"—harmonics that can extend over several kHz and cause unnecessary interference. The ideal RF envelope of a code element would rise and fall in the shape of

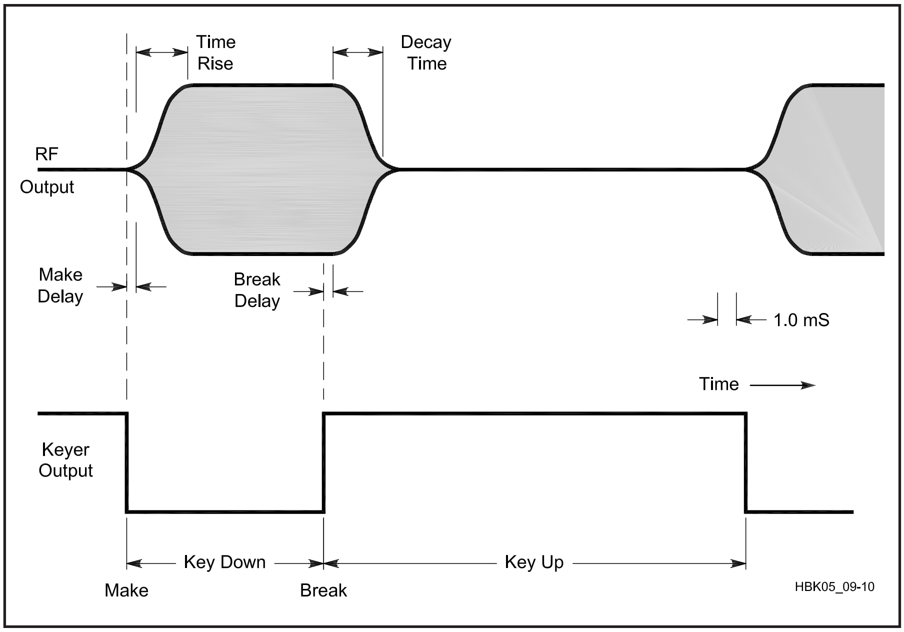


Fig 9.10 — Optimum CW keying waveforms. The on-off transitions of the RF envelope should be smooth, ideally following a sine-wave curve. See text.

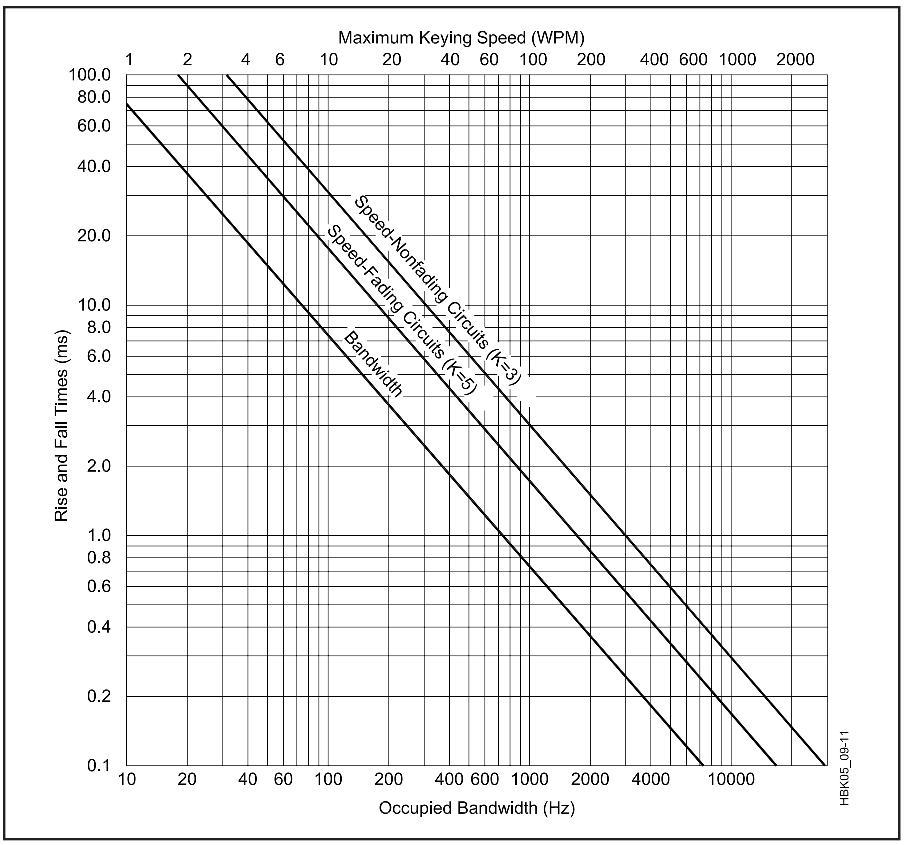


Fig 9.11 — Keying speed vs rise and fall times vs bandwidth for fading and nonfading communications circuits. For example, to optimize transmitter timing for 25 WPM on a nonfading circuit, draw a vertical line from the WPM axis to the K = 3 line. From there draw a horizontal line to the rise/fall time 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.

a sine wave. See **Figs 9.10** and **9.11**. ARRL has long recommended a 5-ms rise time for CW, up to 60 WPM, which keeps the signal within a 150-Hz bandwidth. Use of a narrower filter than this on the receive end is uncommon for ear-copied CW; therefore, narrower bandwidth is unnecessary and would make the signal sound "mushy." Very fast pulses, such as would be used for High-Speed CW (HSCW) meteor scatter work, are computer generated, and can occupy a normal SSB filter bandwidth. On the VHF bands where this mode is used, bandwidth is not at a premium.

Morse code is one of the most efficient modes in terms of information sent per baud. The commonly accepted ratio for Bauds to WPM is WPM = 1.2 × B. Thus, a keying speed of 25 dots per second or 50 Bauds is equal to 60 WPM. The efficiency of Morse text messages is based on the use of the shortest code combinations to represent the most commonly used letters and symbols. Efficiency is further achieved by extensive use of abbreviations and Q signals. By making use of these multiple levels of universally recognized coding schemes, CW can get essential information across quickly. CW abbreviations are universal so that simple contacts can be made without the need of a shared language. In skilled hands, CW can achieve a QSO or traffic rate approaching that of phone operation while using $^{1}/_{10}$ the bandwidth.

The practice needed to achieve highly efficient CW operation is both a blessing and a curse. Throughout ham history, "learning the code" has been feared by many and has kept some out of the hobby. Real proficiency requires significant dedication and interest as well as time to achieve but, once gained, provides lifelong satisfaction. As skill with Morse is downgraded or even eliminated as a legal requirement for a ham license, CW will become more and more the domain of amateurs in the original sense of the word; that is, those who do something simply because they love it.

BAUDOT (ITA2) RADIOTELETYPE

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 on the CD-ROM. In Great Britain, the almost-identical code is called *Murray* code. There are many variations in five-bit coded character sets, principally to ac-

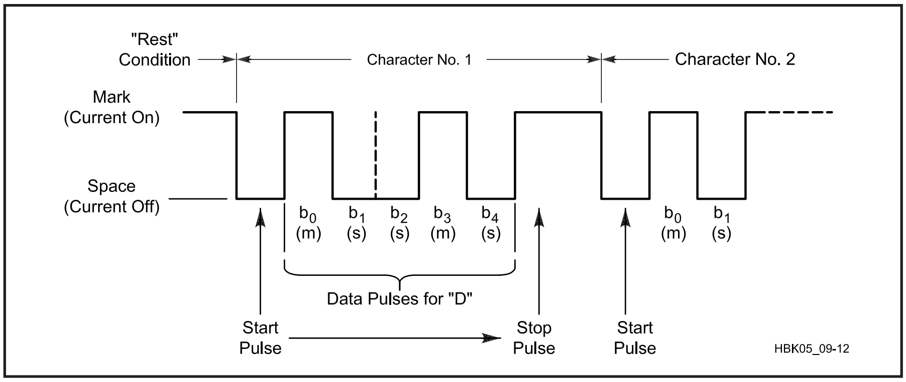


Fig 9.12 — A typical Baudot timing sequence for the letter "D."

commodate 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 9.12**. 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 9.3** 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.

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.

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) and computer sound cards, 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)

Table 9.3
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

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 and computer sound cards running specialized software.

MCPs accept AF signals from a radio and translate them into common ASCII text or graphics file formats (see **Fig 9.13**). 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! Computer sound cards can also do the work of MCPs. There is even multimode software available. Sound cards have become so popular as digital modulators and demodulators, they are rapidly replacing MCPs.

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, addingdiversity may be equivalent to raising transmitter power sevenfold (8 dB).

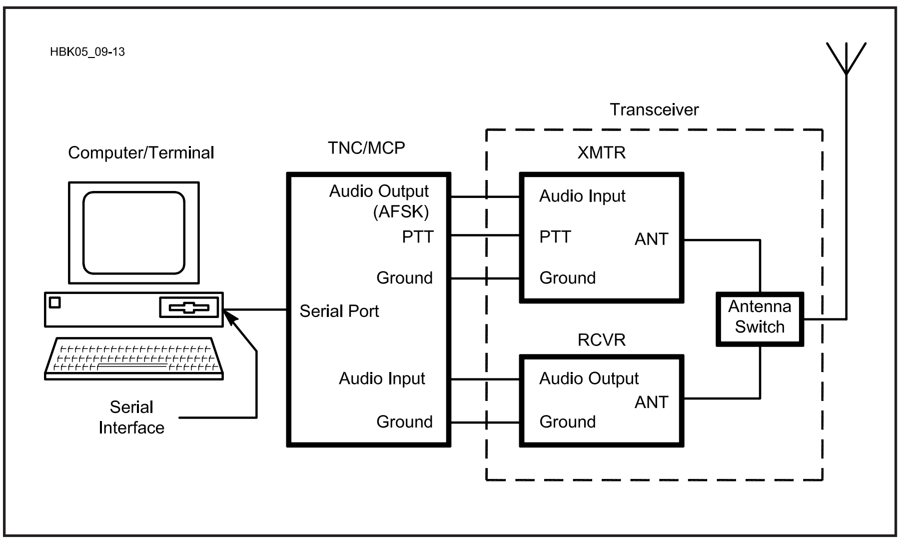


Fig 9.13 — A typical MCP station. MCPs can do all available data modes as well as SSTV and fax.

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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.

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—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.

COFDM—Coded Orthogonal Frequency Division Multiplex, OFDM plus coding to provide error correction and noise immunity.

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

Constellation—A set of points in the complex plane which represent the various combinations of phase and amplitude in a QAM or other complex modulation scheme.

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).

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

Data set—Modem.

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

DRM—Digital Radio Mondiale. A consortium of broadcasters, manufacturers, research and governmental organizations who are developing a system for digital broadcasting in the AM bands between 100 kHz and 30 MHz. The term is also used to refer to the broadcasts themselves.

EIA—Electronic Industries Alliance.

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.

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.

FSK—Frequency-shift keying.

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.

IBOC—In Band On Channel. A method of using the same channel on the AM or FM broadcast bands to transmit simultaneous digital and analog modulation.

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 theinput of a demodulator.

LSB—Least-significant bit.

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).

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.

OFDM—Orthogonal Frequency Division Multiplex. A method of using spaced subcarriers which are phased in such a way as to reduce the interference between them.

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.

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.

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.

QAM—Quadrature Amplitude Modulation. A method of simultaneous phase and amplitude modulation. The number which preceeds it, e.g., 64QAM, indicates the number of discrete stages in each pulse.

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.

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

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.

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.

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

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" on the CD-ROM.

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 9.14A**.

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.

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 9.4**. 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 9.5**). 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.

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

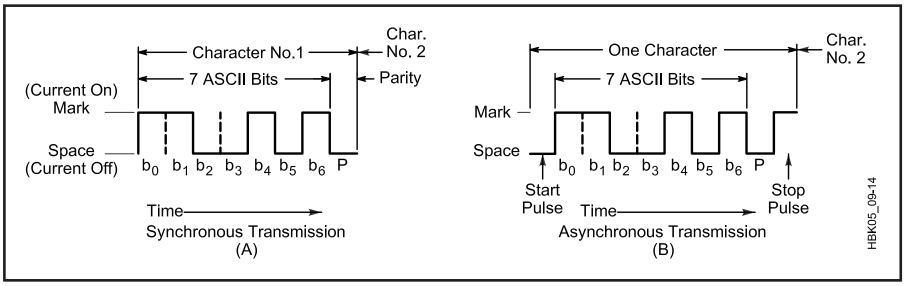

Fig 9.14 — Typical serial synchronous and asynchronous timing for the ASCII character S.

Table 9.4
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 Supplementary: 1800, 3000, 4200, 5400, 6000, 6600 7800, 8400, 9000, 10200, 10800	±0.01%
V.21	110, 150, 300 (where possible)	≤200 bit/s ≤300 bit/s
V.23	600 1200 75 (backward channel)	≤600 bit/s ≤1200 bit/s ≤75 bits
V.34	28800, 26400, 24000, 21600, 19200, 16800 or 14400	
V.35	Preferred: 48000	
V.36	Recommended for international use: 48000 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 Parallel: 75, 150, 300, 600, 900, 1200	
X3.36	Above 9600 bit/s, signaling rates shall be in integral multiples of 8000 bit/s. Selected standard rates: 16000, 56000, 1344000 and 1544000 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	

Table 9.5
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

CPS = characters per second

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

$$\text{WPS} = \text{words per minute} \frac{\text{CPS}}{6} \times 60$$

$$= \text{number of 5-letter-plus-space groups per minute}$$

more than two (mark and space) states. In *dibit* (pronounced die-bit) modulation, two ASCII bits are sampled 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 multi-path 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 multi-path 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.

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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. (See the ITA2 and AMTOR codes table on the CD-ROM.)

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 9.6**. 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.

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.

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 9.15**.

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.

AMTOR Bibliography

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DATACOM, British Amateur Radio Teleprinter Group (BARTG).

Table 9.6
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.

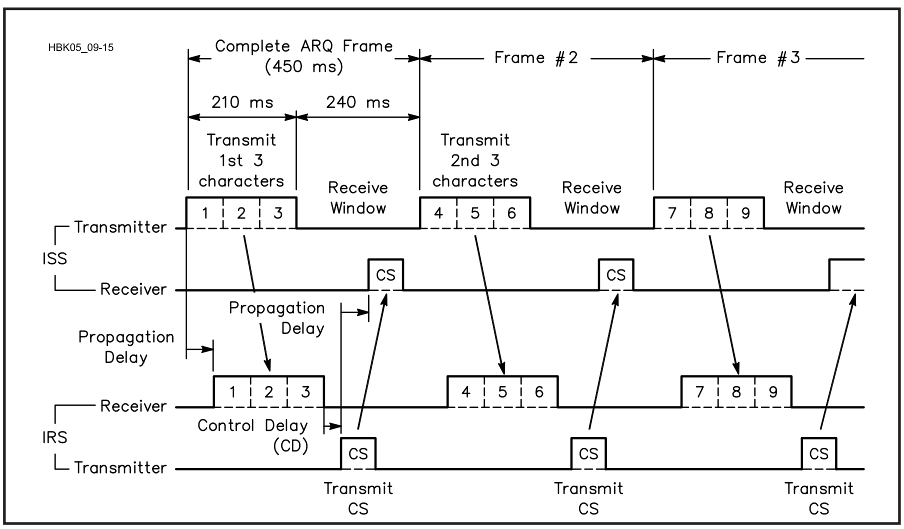

Fig 9.15 — Typical AMTOR timing. Dark arrows indicate the signal path from the ISS to the IRS and vice versa. Note the propagation delays; they determine the minimum and maximum communcations distances.

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.
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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.

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 9.16**).

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. Primary node-to-node links are also active on the Internet.

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.

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 9.17**). 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 9.18**).

If you're saying to yourself, "These TNCs sound a lot like telephone modems,"

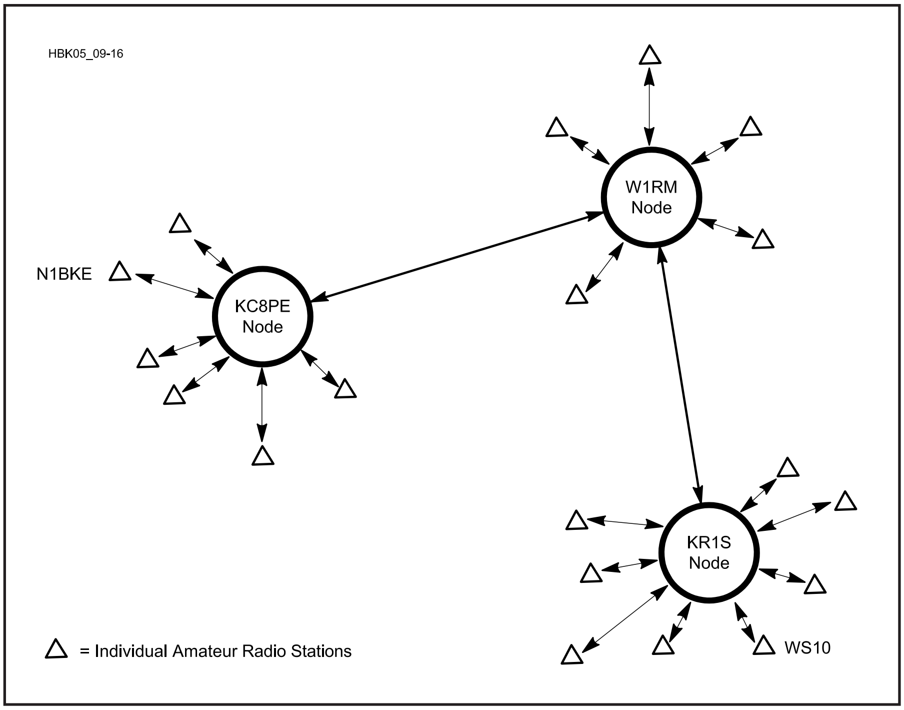

Fig 9.16 — *DX PacketClusters* are networks comprised of individual nodes and stations with an interest in DXing and contesting. In this example, N1BKE is connected to the KC8PE node. If he finds a DX station on the air, he'll post a notice—otherwise known as a *spot*—which the KC8PE node distributes 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.

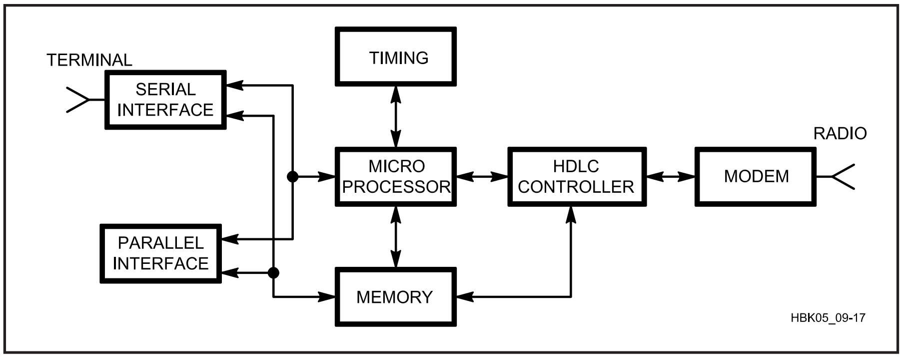

Fig 9.17 — The functional block diagram of a typical TNC.

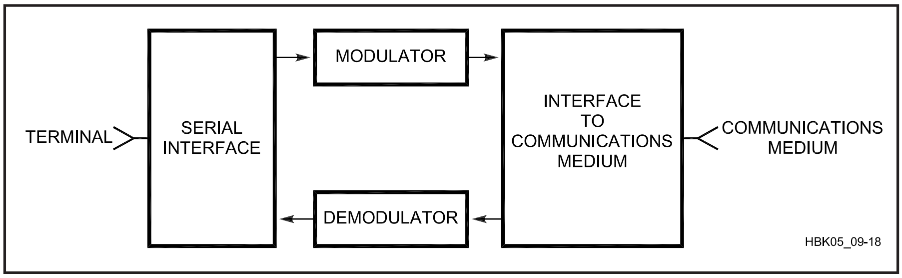

Fig 9.18 — A block diagram of a typical modem.

Fig 9.19 — Four 1200 bit/s packet TNCs: (clockwise, from bottom left) the MFJ-1270C, AEA PK-88, Kantronics KPC-3 and the DRSI DPK-2.

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 9.19**). 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 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.

TNC Emulation

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.

It is also possible to emulate a TNC using software and a sound card. The AGW packet engine for *Windows* is one example. You can download this application at **www.elcom.gr/sv2agw/inst.htm**.

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.

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.

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

In the early '90s, 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 9.20**).

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.

ROSE networks are appearing in many areas of the country. They are especially popular in the southeast and midAtlantic states.

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.

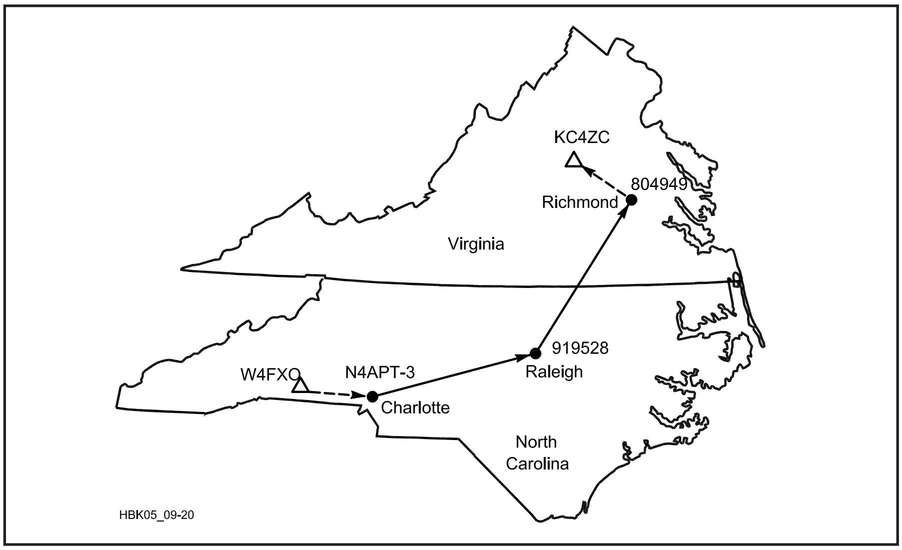

Fig 9.20 — In this hypothetical example, W4FXO, near Charlotte, North Carolina, uses the ROSE network to establish a connection to KC4ZC northwest of Richmond, Virginia. All that W4FXO has to do is issue a connect request that includes his local ROSE switch (N4APT-3) and the ROSE address of the switch 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.

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 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. *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.

APRS

Bob Bruninga, WB4APR, developed Automatic Position Reporting System (APRS) as a result of trying to use packet radio for real-time communications for public service events. Packet radio is not well suited for those real-time events, where information has a very short life time. To solve this problem, APRS avoids the complexity and limitations of trying to maintain a connected network. It uses UI (unconnected) frames to permit any number of stations to participate and exchange data, just like voice users would on a single voice net. Stations that have information to contribute simply transmit it, and all stations monitor and collect all data on frequency. Secondly, APRS recognizes that one of the greatest real-time needs at any special event or emergency is the knowledge of where all stations and other key assets are located. APRS accomplishes the real-time display of operational traffic via a split screen and map displays.

Since the object of APRS is the rapid dissemination of real-time information using packet UI frames, a fundamental precept is that old information is less important than new information. All beacons, position reports, messages and display graphics are redundantly transmitted, but at longer and longer repetition rates. Each new beacon is transmitted immediately, then again 20 seconds later. After every transmission, the period is doubled. After ten minutes only six packets have been transmitted. After an hour this results in only three more beacons; and only three more for the rest of the day! Using this redundant UI broadcast protocol, APRS is actually much more efficient than if a fully connected link had to be maintained between all stations!

The standard configuration for packet radio hardware (radio-to-TNC-to-computer) also applies to APRS until you add a GPS (Global Positioning System) receiver to the mix. You don't need a GPS receiver for a stationary APRS installation (nor do you need a computer for a mobile or tracker APRS installation). In these cases, an extra port or special cable is not necessary. It is necessary, however, when you desire both a computer and a GPS receiver in the same installation.

One way of accomplishing this is by using a TNC or computer that has an extra serial port for a GPS receiver connection. Alternatively, you can use a hardware single port switch (HSP) cable to connect a TNC and GPS receiver to the same serial port of your computer. The HSP cable is available from a number of sources including TNC manufacturers Kantronics, MFJ and PacComm.

Whichever GPS connection you use, make sure that you configure the APRS software so it is aware that a GPS receiver is part of the hardware configuration and how the GPS receiver connection is accomplished.

APRS also supports an optional weather station interface. The wind speed, direction, temperature and rainfall are inserted into the station's periodic position report. The station shows up on all APRS maps as a large blue dot, with a white line showing the wind speed and direction. Several automatic APRS weather reporting stations, supported with additional manual reporting stations, can form a real-time reporting network in support of SKYWARN activities

For additional information see the book, *APRS: Moving Hams on Radio and the Internet* by Stan Horzepa, WA1LOU, published by ARRL.

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.

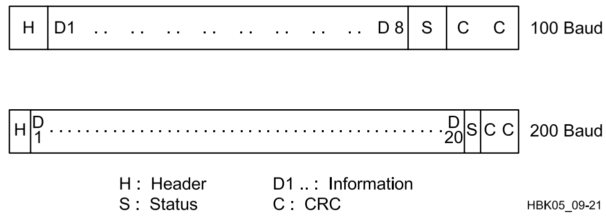

Fig 9.21 — PACTOR data packet format.

Transmission Formats

Information Blocks

All packets have the basic structure shown in **Fig 9.21**, and their timing is as shown in **Table 9.7**:

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 9.8**

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

Acknowledgment Signals

The PACTOR acknowledgment signals are similar to those used in AMTOR, except for CS4 (see **Table 9.9**). 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 an 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 9.10**

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 can acknowledge with CS4. TX

Table 9.7
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 9.8
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.

Table 9.9
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 9.10
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

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 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.

G-TOR

This brief description has been adapted from "A Hybrid ARQ Protocol for Narrow Bandwidth HF Data Communication" by Glenn Prescott, WBØSKX, Phil Anderson, WØXI, Mike Huslig, KBØNYK, 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 band-width 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 9.22**) 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.

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

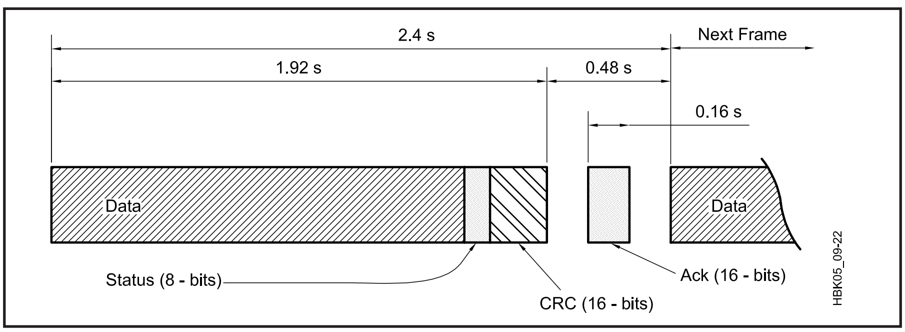

Fig 9.22 — G-TOR ARQ system timing and frame structure before interleaving. The data portion may be 69 (300 baud), 45 (200 baud) or 21 (100 baud) bytes depending on the channel quality.

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 9.23**. 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.

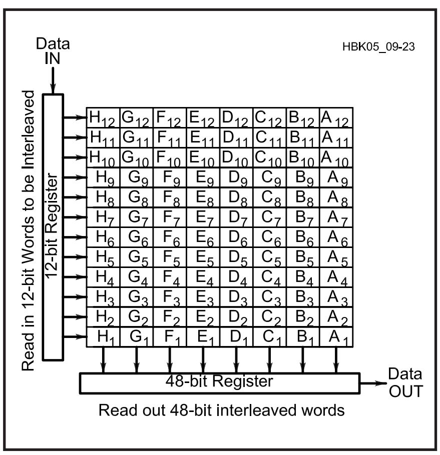

Fig 9.23 — Interleaving the bits to be transmitted.

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

Table 9.11

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

of data modes that may be used (see **Table 9.11**). 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 9.24**, **9.25** and **9.26**. 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 180º 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 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

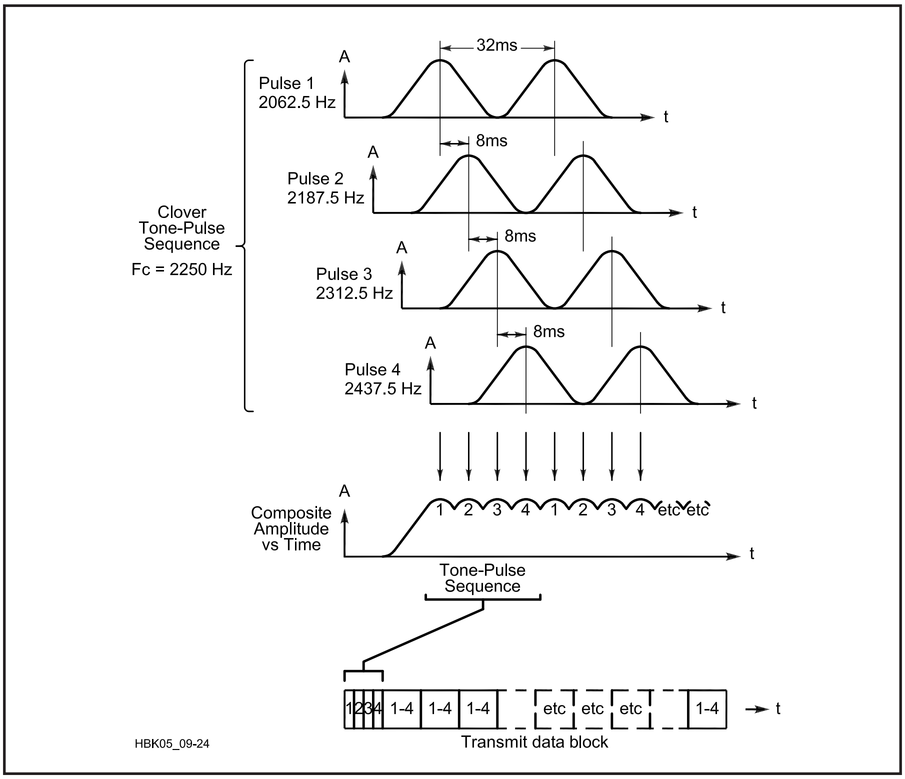

Fig 9.24 — Ampitude vs time plots for CLOVER-II's four-tone waveform.

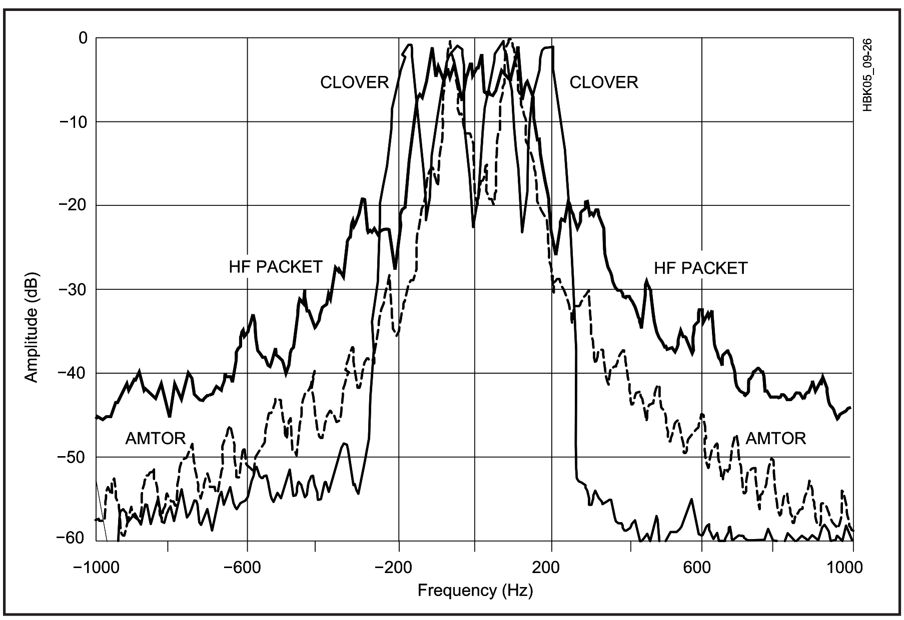

Fig 9.26 — Spectra plots of AMTOR, HF packet-radio and CLOVER-II signals.

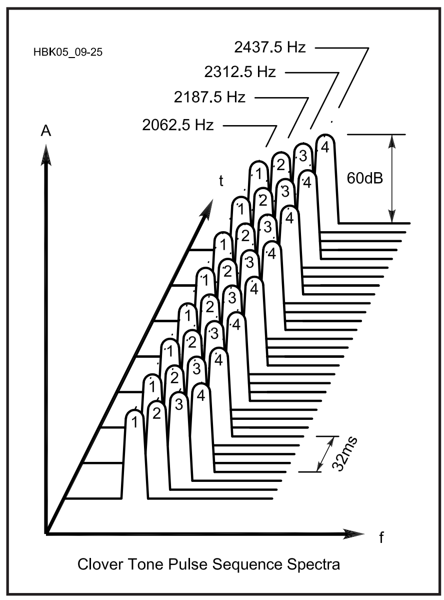

Fig 9.25 — A frequency-domain plot of a CLOVER-II waveform.

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 9.12 and **Table 9.13** detail the relationships between block size, coder efficiency, data bytes per block and correctable byte errors per block.

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 9.26 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 9.27 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).

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.

Table 9.12
Data Bytes Transmitted Per Block

Block Size	*Reed-Solomon Encoder Efficiency* 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 9.13
Correctable Byte Errors Per Block

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

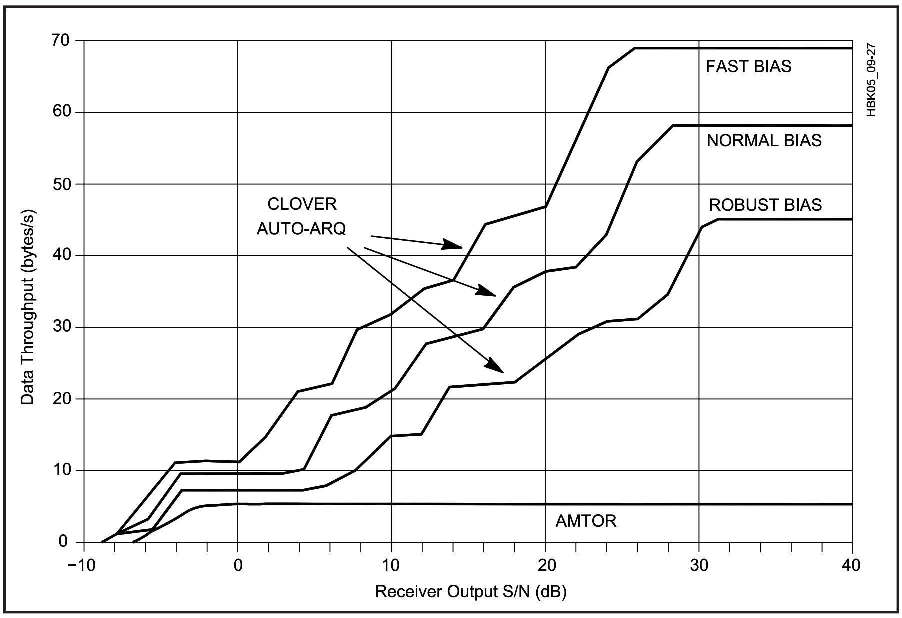

Fig 9.27 — ARQ-mode data throughput vs receiver S/N ratio for AMTOR and three different CLOVER-II configurations.

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 9.28**). Much like the Morse code, the more commonly used letters have shorter codes.

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

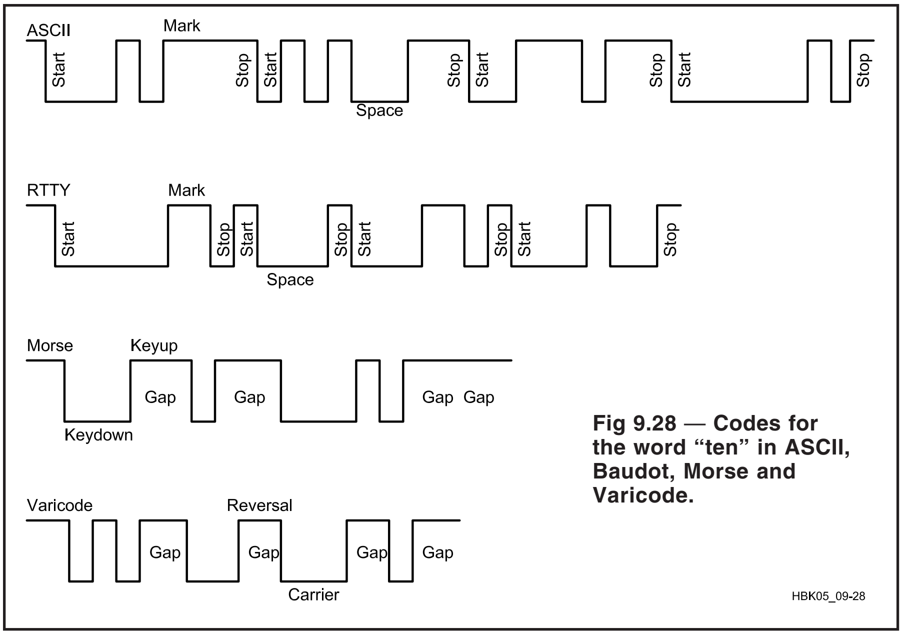

Fig 9.28 — Codes for the word "ten" in ASCII, Baudot, Morse and Varicode.

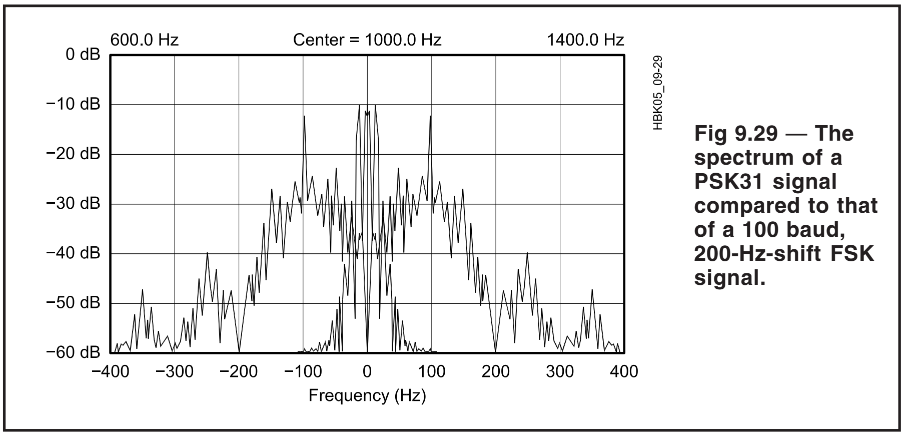

Fig 9.29 — The spectrum of a PSK31 signal compared to that of a 100 baud, 200-Hz-shift FSK signal.

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 9.29**.

Error Correction

Martinez 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 correspond 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 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, Steven Karty, N5SK, and Ralph Taggart, WB8DQT, 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] and the *ARRL Image Communications Handbook*.[3]

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

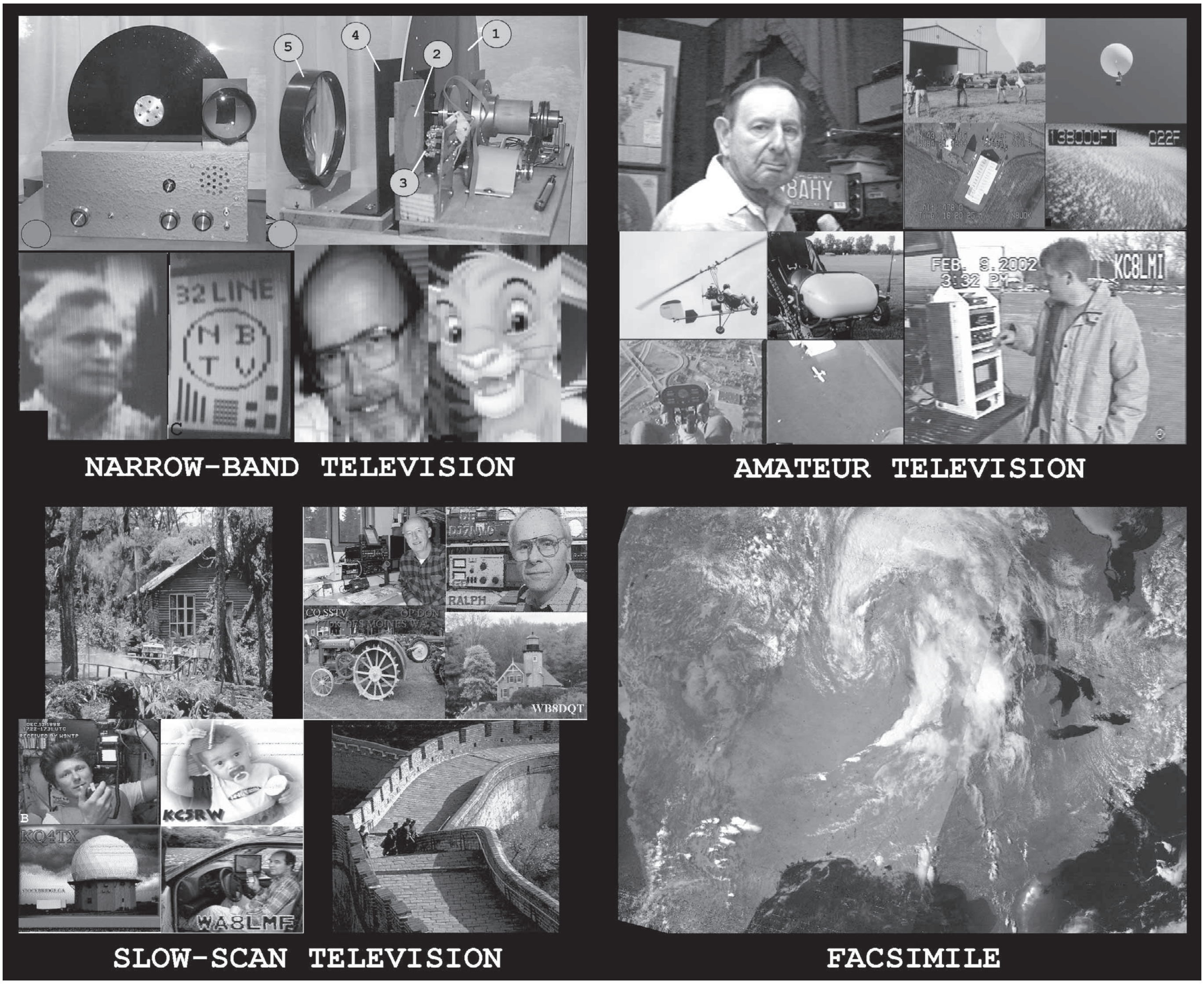

Fig 9.30 — Amateur Image Communications encompass a wide range of activities, a few of which are illustrated here. Narrowband Television (NBTV) experimenters explore the history and technology of the earliest days of television by restoring or recreating mechanical TV gear while exploring the possibilities of narrowband, full motion TV, primarily using computer technology. Amateur Television (ATV) operators use standard broadcast television, typically in color, to communicate on UHF and microwave frequencies. The scope of their operating activities ranges from point-to-point communication (simplex or via local ATV repeaters), roving or portable operation for a variety of reasons, including emergency and public service communications, and the application of ATV to remote sensing via aircraft, high-altitude balloons, and remote-control vehicles of all sorts. Slow-scan Television (SSTV) involves the transmission of medium and high-resolution images, usually in full-color, using standard Amateur voice equipment (typically SSB or FM). Most modern SSTV activity is computer-based, offering international DX on HF frequencies and local, regional, or space communications (satellite, MIR, and now the International Space Station) on VHF and UHF. Facsimile (Fax) encompasses the transmission and reception of very high-resolution still images (typically using computers) over a period of several to many minutes. One of the most popular areas of Amateur experimentation and operation has involved the reception of imagery from polar-orbit and geostationary weather satellite. While this *Handbook* will provide a brief introduction to some of these activities, all of them and more are covered in much greater detail in the *ARRL Image Communications Handbook*.

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 share ware 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 (files) link in the site index. You can use any commercial search site to look for "fax" AND "software." Examples of 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. 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*. 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.

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. 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. 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 the HF radio link (based on the "phone number"), and handshakes with the controller at the other end to start the receiving fax machine. 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.

Notes

[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).

[3]Taggart, R.E., *ARRL Image Communications Handbook*, 1st Ed. (Newington: ARRL, 2002).

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 available was very expensive and home-brewing was much too complicated

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.

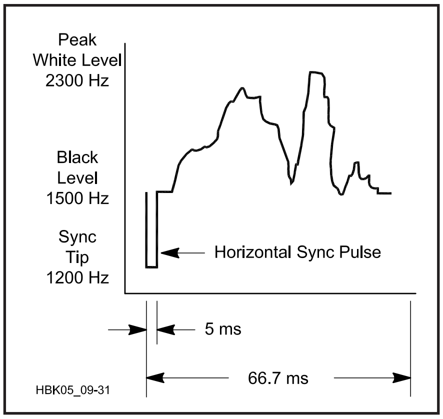

Fig 9.31 — Early SSTV operators developed a basic 8-second black and white transmission format. The sync pulses are often called "blacker than black." A complete picture would have 120 lines (8 seconds at 15 ms per line). Horizontal sync pulses occur at the beginning of every line; a 30 ms vertical sync pulse precedes each frame.

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 homebrew 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 9.31**. 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 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

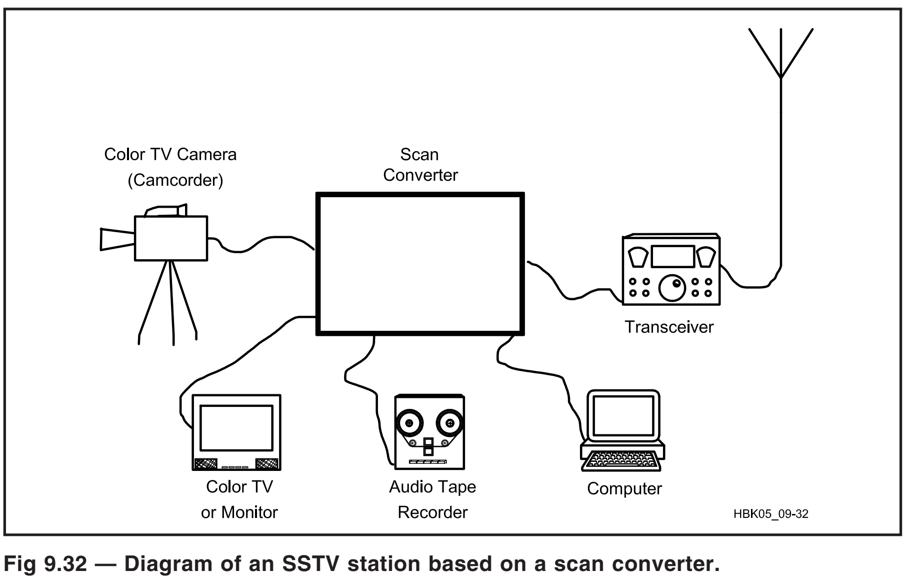

Fig 9.32 — Diagram of an SSTV station based on a scan converter.

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 9.14** 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 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 9.32 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 9.33**. 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 in-

Table 9.14
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.

ternal 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**.

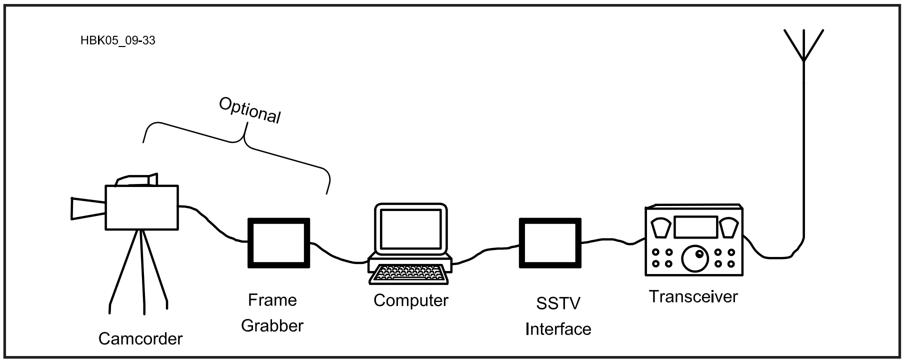

Fig 9.33 — A modern, PC-based SSTV station.

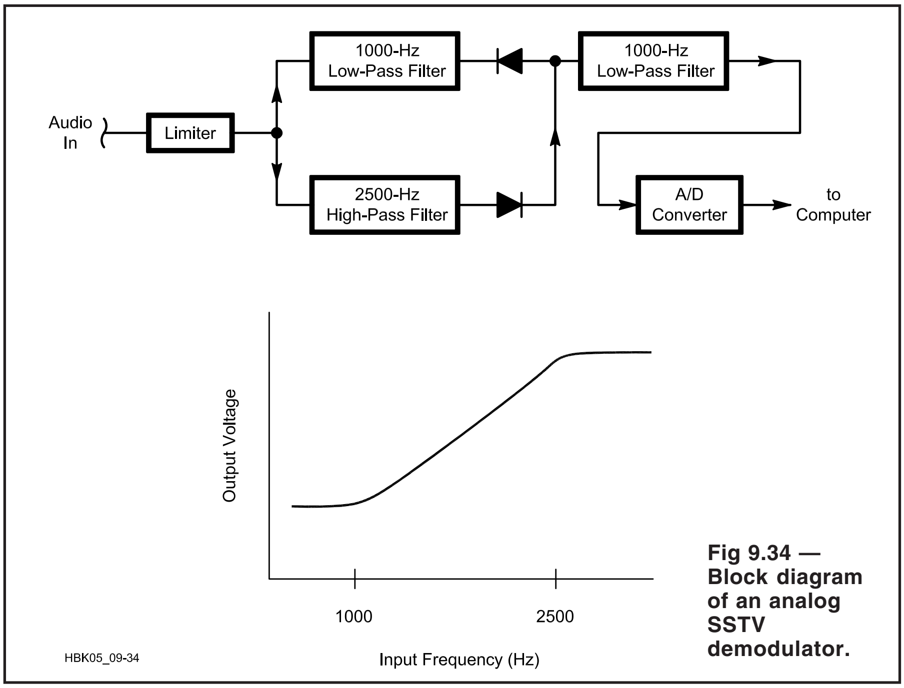

Fig 9.34 — Block diagram of an analog SSTV demodulator.

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

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.

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*

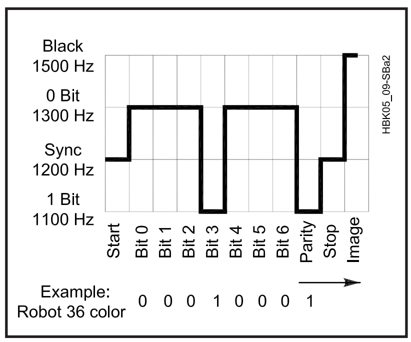

Fig A — Composition of the vertical interval signaling (VIS) code.

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 9.34** 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 9.35**). When the signal amplitude crosses zero, a counter is reset. It then proceeds to count 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 9.36** 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 non-Robot 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 9.37**) system for IBM

```
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 9.36 — An outline of typical software written to display an SSTV frame from received digital picture information.

Fig 9.37 — An example SSTV image.

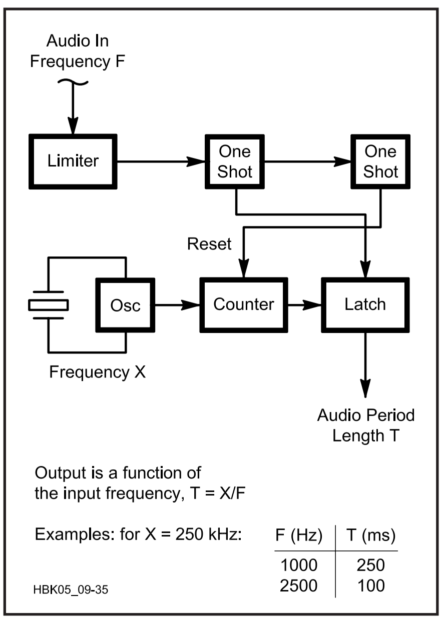

Fig 9.35 — Block diagram of a digital SSTV demodulator.

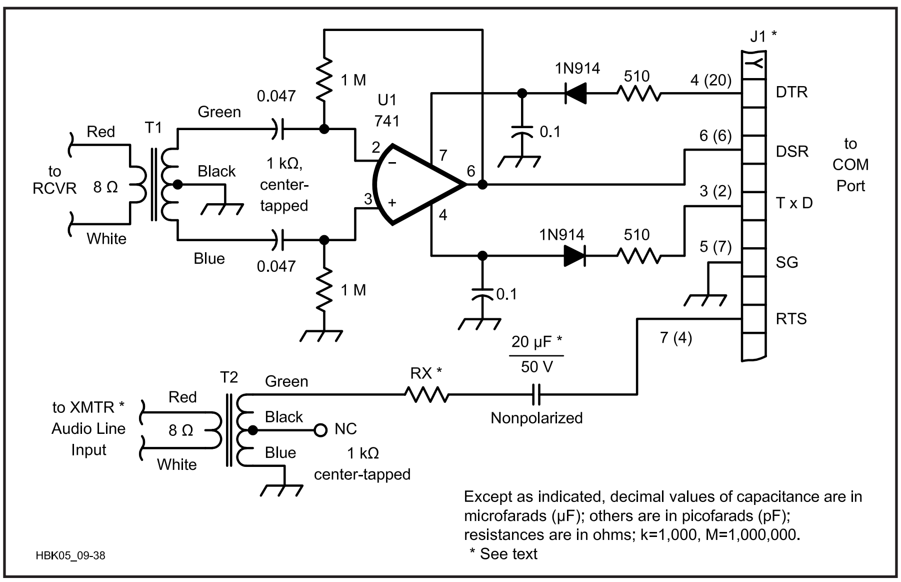

Fig 9.38 — Schematic of the simple SSTV receive and transmit circuit. This circuit appears on page 34 of July 1998 *QST*. T1 and T2 are RadioShack 273-1380 audio-output transformers; the 20-µF, 50-V capacitor is a parallel combination of two RadioShack 272-999 10-µF, 50-V nonpolarized capacitors; equivalent 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.

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* (arrl.org).

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 9.38 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 if 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 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, but perfboard construction employing short leads works fine.

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.

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"

For More Information

Amateur Television Quarterly
Web: **www.hampubs.com**
Phone: 815-398-2683
Postal: Harlan Technologies
5931 Alma Drive
Rockford, IL 61108-2409
USA

American Radio Relay League, Inc. (ARRL)
Web: **www.arrl.org**
Phone: 860-594-0200
Postal: 225 Main St
Newington, CT 06111-1494
USA

Benelus NBTV Website (ON1AIJ)
Web: **users.pandora.be/ON1AIJ/English.htm**

Dave Jones CQ SSTV Web site
Web: **www.tima.com/~djones/**

Experimental Television Society (ETS)
Web: **pyanczer.home.mindspring.com/Tour/**
Phone: 314-822-1748
Postal: c/o Peter Yanczer
835 Bricken Place
Warson Woods, MO 63122-1613
USA

***Image Communications Handbook*—Web site**
Web: **taggart.glg.msu.edu/ICH/ICH.htm/**

International Visual Communications Association
Web: **www.mindspring.com/~sstv/**

Narrow-Band Television Association (NBTVA)
Web: **www.nbtv.wyenet.co.uk/**

Remote Imaging Group (RIG)
Web: **www.rig.org.uk/**

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 and Software Systems Consulting 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.

Perhaps the single most significant breakthrough in computer-based SSTV is the wide range of new Windows-based programs using the ever-present PC soundcard as the transmit/receive interface. Information on all current computer SSTV software is available at **www.tima.com/~djones**. The subject of computer SSTV software and interfacing is discussed at length in the *Image Communications Handbook* published by ARRL.

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.

Notes

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

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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.

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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 9.39** and **9.40**). What-

Fig 9.39 — Students enjoy using ATV to communicate between science and computer classes.

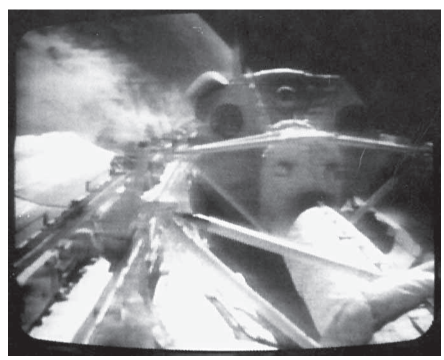


Fig 9.40 — The ATV view shows the aft end of the Space Shuttle cargo bay during mission STS-9.

Table 9.15
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

ever 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.

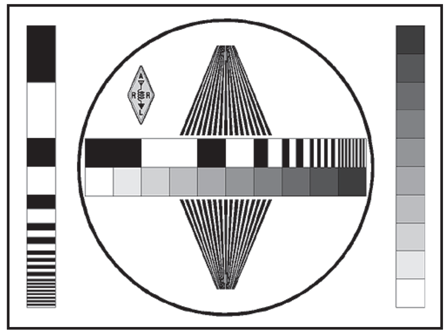
Emergency service coordinators have found that live video from a site gives a better understanding of a situation than is possible from voice descriptions alone. Weather-radar video, WEFAX, or other computer generated video has also been carried by ATV transmitters for RACES groups during significant 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?

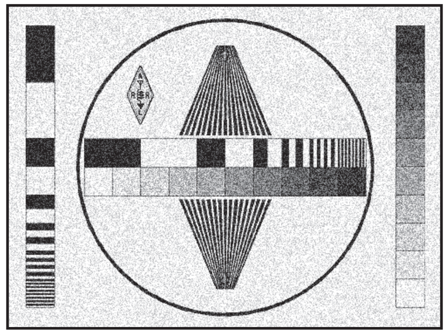
The theoretical snow-free line-of-sight distance for 10 W, given 15.8-dBd antennas and 2-dB feed-line loss at both ends, is 91 miles. (See **Table 9.15**.) However, except for temperature-inversion skip conditions, reflections, or through high hilltop repeaters, direct line-of-sight ATV contacts seldom 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 9.41**) requires at least 200 µV (–61 dBm) of signal at the input of the ATV receiver, depending 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.

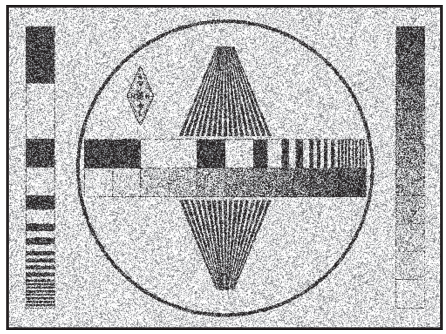
Two-meter FM is used to coordinate



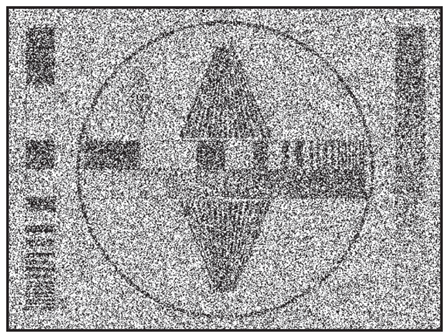
P5—Excellent



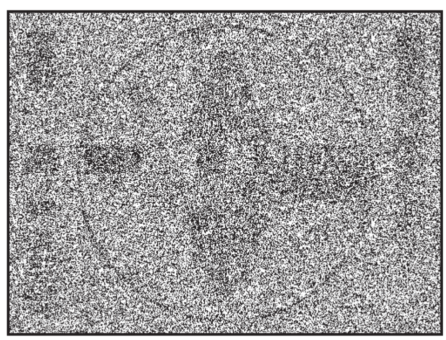
P4—Good



P3—Fair



P2—Poor



P1—Barely perceptive

Fig 9.41—An ATV quality reporting system.

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.

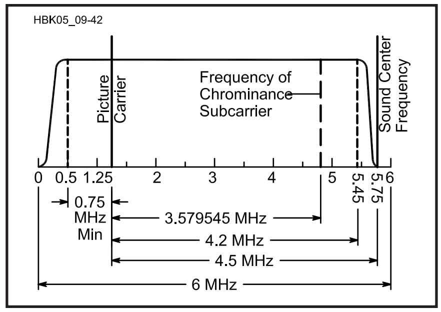


Fig 9.42 — A 6-MHz video channel with the video carrier 1.25 MHz up from the lower edge. The color subcarrier is at 3.58 MHz and the sound subcarrier at 4.5 MHz above the video carrier.

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 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 9.42**.) 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 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 9.43**, 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 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 9.44**). 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.

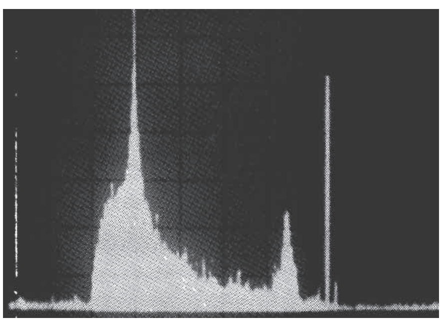


Fig 9.43 — A spectrum-analyzer photo of a color ATV signal. Each vertical division represents 10 dB; horizontal divisions are 1 MHz. Spectrum power density varies with picture content, but typically 90% of the sideband power is within the first 1 MHz.

Fig 9.44 — 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 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 9.45**. 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 9.46** shows a block diagram of this transceiver.

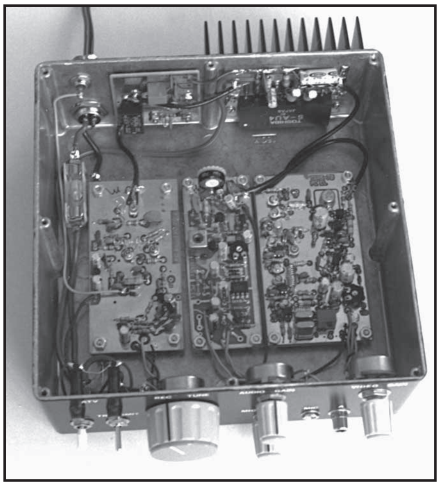
The complete 20-W ATV transceiver consists of the:

- *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.

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

(A)



(B)

Fig 9.45 — 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.

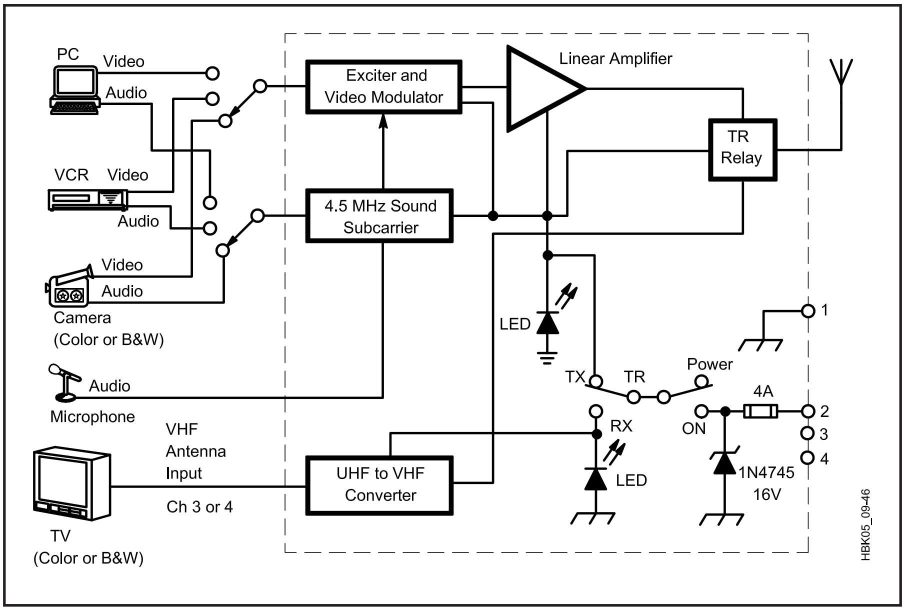


Fig 9.46 — Block diagram of a complete ATV station using the 20-W transceiver.

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 wirewounds 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 downcon-

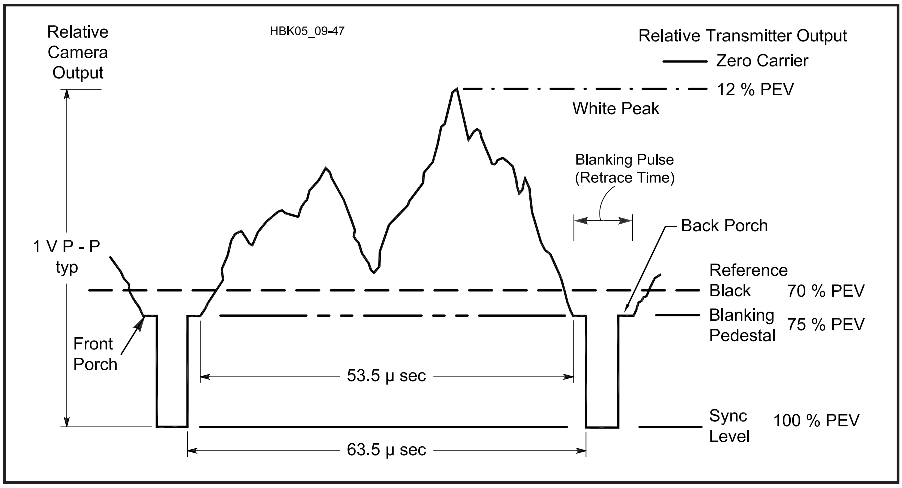


Fig 9.47 — An ATV waveform. Camera and corresponding transmitter RF output power levels during one horizontal line scan for black-and-white TV. (A color camera would generate a "burst"of 8 to 10 cycles at 3.58 MHz on the back porch of the blanking pedestal.) Note that "black" corresponds to a higher 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.

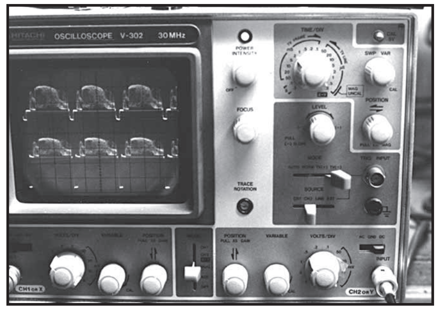


Fig 9.48 — An oscilloscope used to observe a video waveform. The lower trace is the video signal as it comes out of the sync stretcher. The upper trace is the signal from the Mirage D1010-N amplifier.

verter 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 9.47**). 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 careful not to overmodulate. 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 9.48**). With both video and sound subcarrier disconnected, the pedestal control is set 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 9.49 shows a block diagram for a simple 70-cm in-band repeater. No duplexer is shown because 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 9.50** 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.

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 9.51** compares AM and FMATV across a wide

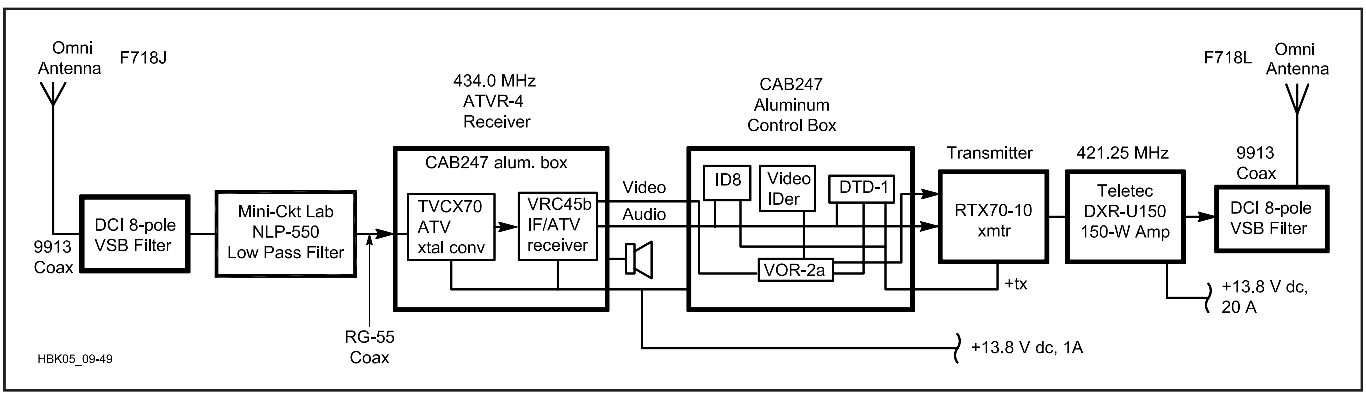


Fig 9.49 — A block diagram of a 70-cm in-band ATV repeater. The antennas are Diamond omnidirectional verticals, which require 20 ft (minimum) of vertical separation to prevent receiver desensitization. The VSB filters are made by DCI; they have the proper band-pass characteristics and only 1 dB insertion loss. A low pass filter on the receiver 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.

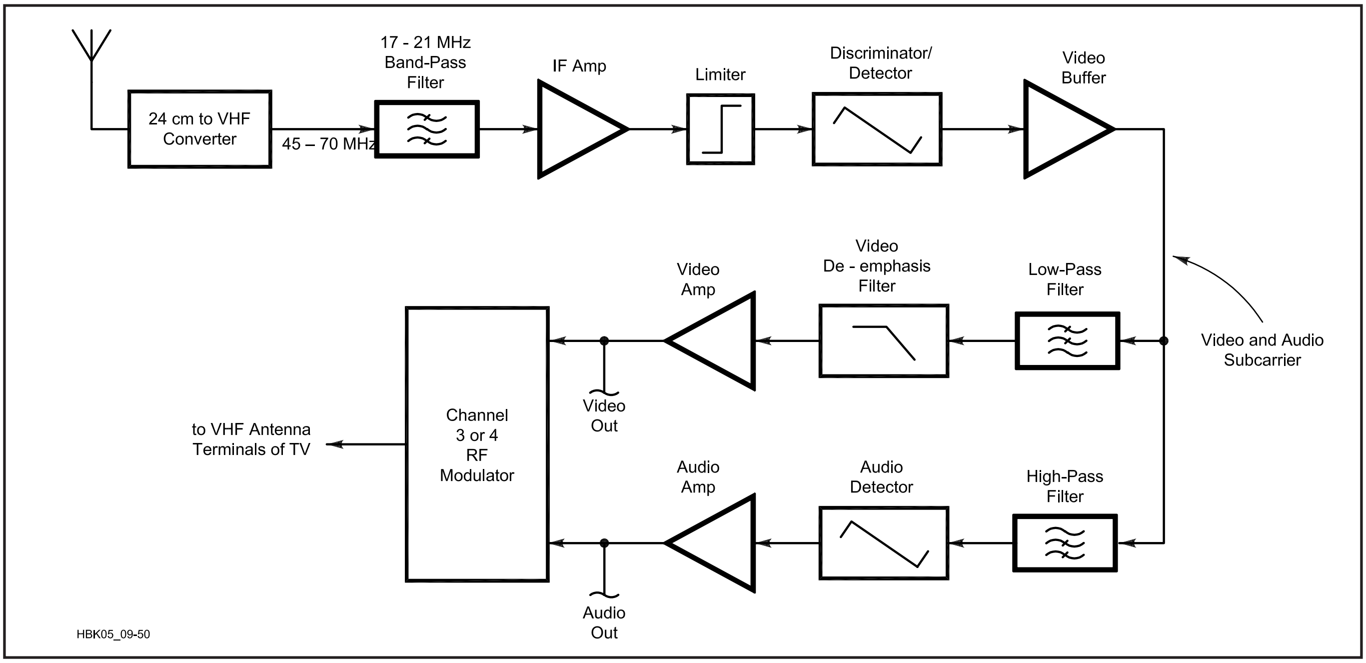


Fig 9.50 — Block diagram of an FMATV receiver.

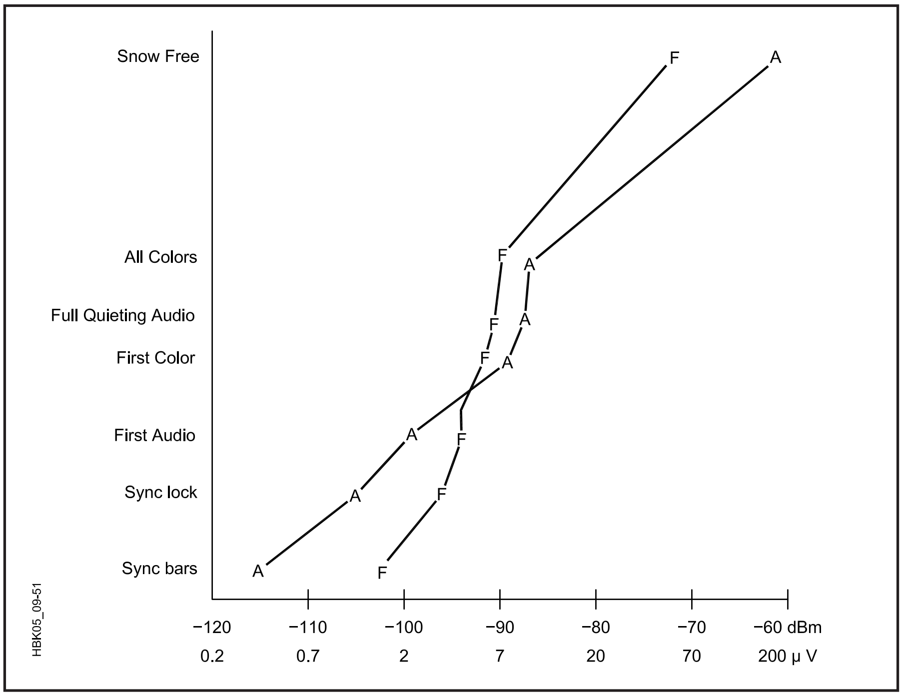


Fig 9.51 — Two approaches to ATV receiving. This chart compares AM (A) and FM (F) ATV as seen on a TV receiver and monitor. Signal levels are into the same downconverter with sufficient gain to be at the noise floor. The FM receiver bandwidth is 17 MHz, using the US standard.

range of signal strengths.

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 **Fig 9.52**.

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

Fig 9.52 — N8QPJ mounted an ATV setup aboard this model Humvee.

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*.

Taggart, R., WB8DQT, *Image Communications Handbook*, Published by ARRL, Newington, CT, 2002. ARRL Order No. 8616. See also **www.arrl.org/catalog**.

ATV Equipment Sources

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

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 9.53** 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 9.54** 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 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 cur-

Fig 9.53 — Photo of three R/C model electric cars.

Fig 9.54 — Photo of two R/C aircraft models.

rently authorizes 91 R/C frequencies between 27 MHz and 76 MHz. Some frequencies are for all models, 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. The *ARRL Repeater Directory* lists current Amateur Radio R/C frequencies.

(A)

(B)

Fig 9.55 — A, photo of Futaba's Conquest R/C aircraft transmitter. B shows the matching airborne system.

Fig 9.55 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 9.56** shows a 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.

Fig 9.56 — Photo of Airtronics Infinity 660 R/C aircraft transmitter.

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

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 9.57 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

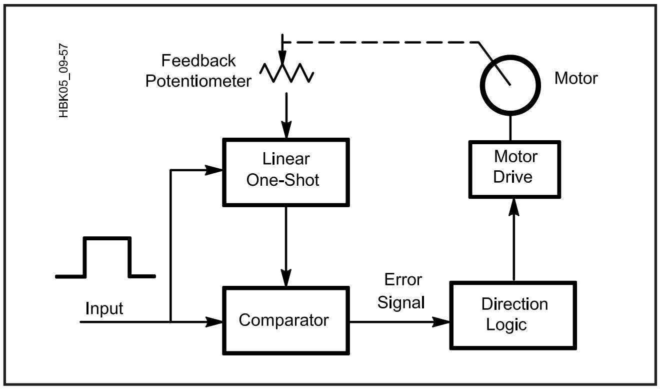


Fig 9.57 — Diagram of a pulse-feedback servo.

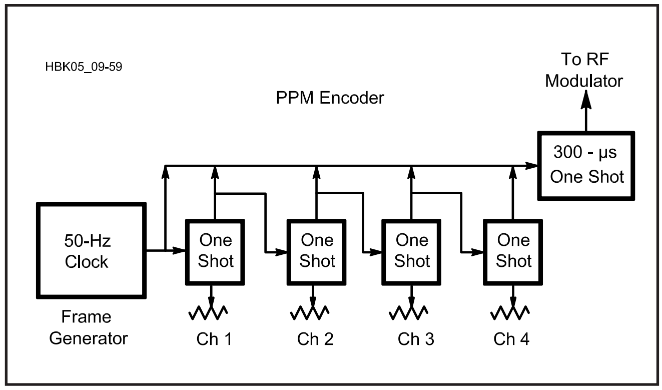


Fig 9.59 — Diagram of a PPM encoder.

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 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).

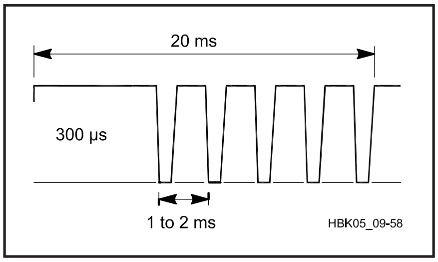


Fig 9.58 — Diagram of a four-channel PPM RF envelope.

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 9.58** 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 9.59**. 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.

Received pulse decoding can also use digital logic semiconductors. **Fig 9.60** shows a simple four-control-channel decoder circuit using a 74C95 CMOS logic

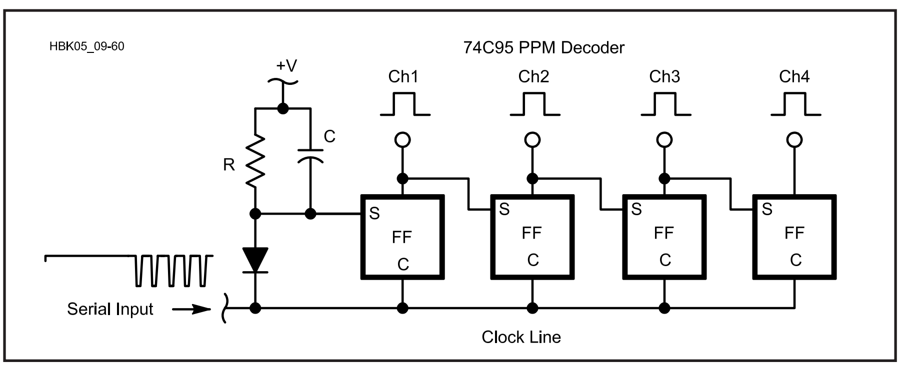


Fig 9.60 — Diagram of a 74C95 PPM decoder.

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.

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

The introduction to spread spectrum communications was written by André Kesteloot, N4ICK. *The ARRL Spread Spectrum Sourcebook* contains a more complete treatment of the subject. The information which follows on contemporary Amateur Radio use of spread spectrum technology for multimedia amateur radio communications was written by John Champa, K8OCL, and Kris Mraz, N5KM.

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,[1] 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

$$C = W \log_2 \left(1 + \frac{S}{N}\right) \text{ bits/s} \quad (6)$$

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 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, amateurs can use any of them as long as there modulation scheme has been published, for example on the ARRL website. By far, frequency-hopping (FH) and direct-sequence spread spectrum (DSSS) are the most popular forms within 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 9.61**). Should some source interfere with the

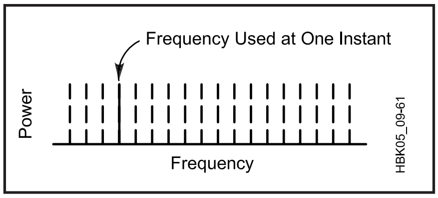


Fig 9.61 — Power vs frequency for frequency-hopping spread spectrum signals. Emissions jump around to discrete frequencies in pseudo-random fashion.

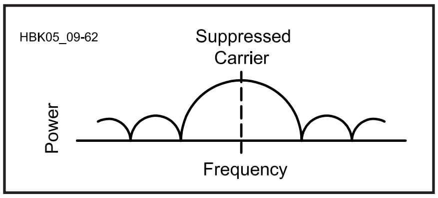


Fig 9.62 — Power vs frequency for a direct-sequence-modulated spread spectrum signal. The envelope assumes the shape of a (sin x/x)[2] curve. With proper modulating techniques, the carrier is suppressed.

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 9.62**. 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. Additional 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 Radio 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. The block diagram is shown in **Fig 9.63**. **Fig 9.64** 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 1989 in a paper titled *License-Free Spread Spectrum Packet Radio,* Al Broscius, N3FCT, suggested the use of Part 15 spread spectrum wireless local area network (WLAN) devices that were becoming available be put to use in amateur radio.

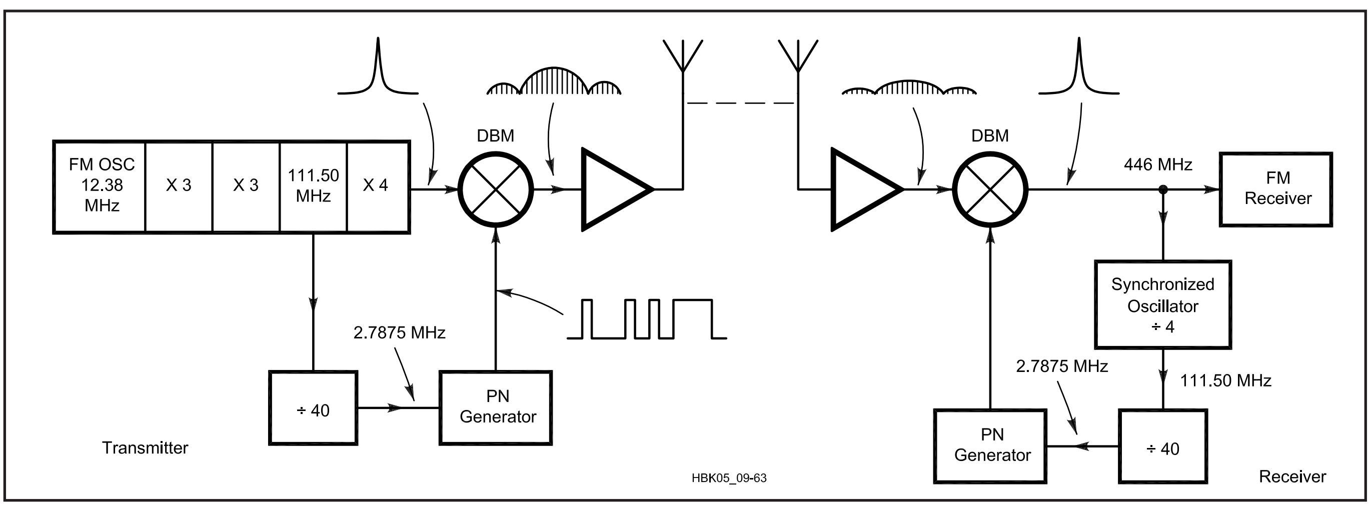


Fig 9.63 — A block diagram of the practical spread spectrum link. The success of this arrangement lies in the use of a synchronized oscillator (right) to recover the transmitter clock signal at the receiving site.

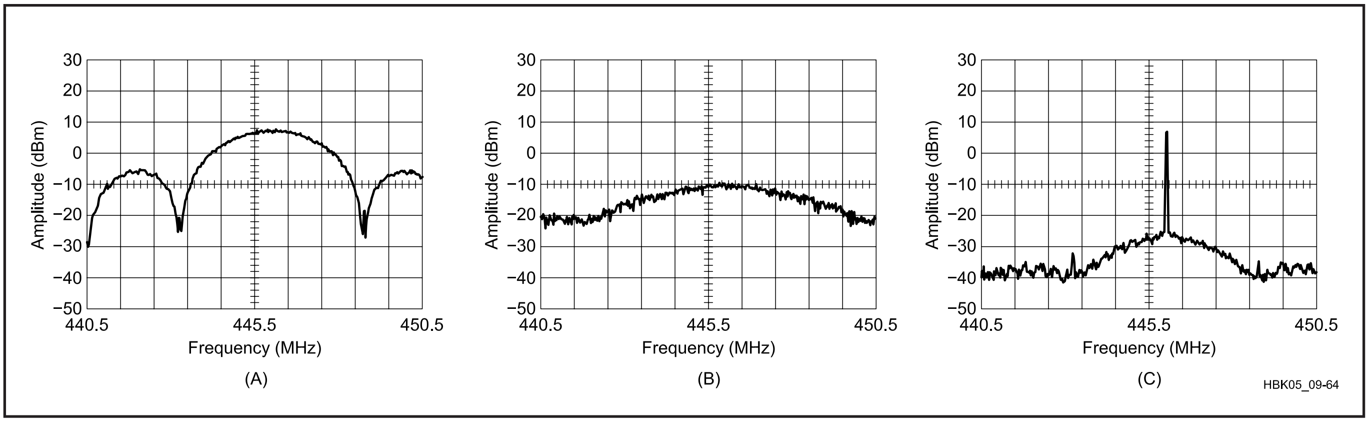


Fig 9.64 — (A) The envelope of the unfiltered biphase-modulated spread spectrum signals as viewed on a spectrum analyzer. In this practical system, band-pass filtering is used to confine the spread spectrum signal to the amateur band. (B) At the receiver end of the line, the filtered spread spectrum 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.)

In 1997 TAPR started the development of a 1 watt, 128 kbps, FHSS radio for the amateur radio 902 MHz band.

In late 1999 the FCC considerably relaxed the amateur radio service rules regarding the use of spread spectrum. These changes allowed amateurs to use any commercial off-the-shelf (COTS) Part 15 spread spectrum device to be reclassified under Part 97 (see Part 97.311 for details).

Emergence of Commercial Part 15 Equipment

Just as military surplus radio equipment fueled Amateur Radio in the 1950's, and commercial FM radios and repeaters snowballed the popularity of VHF/UHF amateur repeaters in the 1960's and 70's, the availability of commercial wireless LAN (WLAN) equipment is driving the direction and popularity of Amateur Radio use of spread spectrum in the 2000's. FCC Part 15 provides the technical rules for commercial spread spectrum equipment. The IEEE has provided the standards under which manufacturers have developed equipment for sale commercially. IEEE 802.11 standardized FHSS and DSSS for the 2.4 GHz ISM band at data rates of 1 and 2 Mbps. Next came the release of 802.11b which provided the additional data rates of 5.5 and 11 Mbps but only for DSSS. FHSS was not carried forward. This was followed by 802.11g which provided standardization using Orthogonal Frequency Division Multiplexing (OFDM) for data rates of 6, 9, 12, 18, 24, 36, 48 and 54 Mbps as well as backward compatibility with 802.11b. As of this writing the most recent release of the standard is 802.11a. This release addresses the use of OFDM in the 5 GHz ISM and UNII bands. It provides the same data rates as 802.11g. The currently unreleased 802.11n standard promises data rates in excess of 108 Mbps.

Frequency Hopping Spread Spectrum

FHSS radios, as specified in 802.11, hop among 75 of 79 possible non-overlapping frequencies in the 2.4 GHz band. Each hop occurs approximately every 400 ms with a hop time of 224 µs. Since these are Part 15 devices the radios are limited to a maximum peak output power of 1 Watt and a maximum bandwidth of 1 MHz (–20 db) at any given hop frequency. The rules allow using a smaller number of hop frequencies at wider bandwidths (and lower power: 125 mW) but most manufacturers have opted not to develop equipment using these options. Consequently, off-the-shelf equipment with this wider bandwidth capability is not readily available to the amateur.

The hopping sequences are well defined by 802.11. There are three sets of 26 such sequences (known as channels) consisting of 75 frequencies each. The ordering of the frequencies is designed as a pseudo-random sequence hopping at least 6 MHz higher or lower that the current carrier frequency such that no two channels are on the same frequency at the same time. Channel assignment can be coordinated among multiple collocated networks so that there is minimal interference among radios operating in the same band.

The FHSS radio can operate at data rates of 1 and 2 Mbps. The binary data stream modulates the carrier frequency using frequency shift keying. At 1 Mbps the carrier frequency is modulated using 2-Level Gaussian Frequency Shift Keying (2GFSK) with a shift of ±100 kHz. The data rate can be doubled to 2 Mbps by using 4GFSK modulation with shifts of ±75 KHz and ±225 KHz.

Direct Sequence Spread Spectrum

DSSS uses digital modulation to accomplish signal spreading. That is, a well-known pseudo-random digital pattern of ones and zeros is used to modulate the data at a very high rate. In the simplest case of DSSS, defined in 802.11, an 11-bit pattern known as a Barker sequence (or Barker code) is used to modulate every bit in the input data stream. The Barker sequence is 10110111000. Specifically, a "zero" data bit is modulated with the Barker sequence resulting in an output sequence of 10110111000. Likewise, a "one" data bit becomes 01001000111 after modulation (the inverted Barker code). These output patterns are known as *chipping* streams; each bit of the stream is known as a *chip*. It can be seen that a 1 Mbps input data stream becomes an 11 Mbps output data stream.

The DSSS radio, like the FHSS radio, can operate at data rates of 1 and 2 Mbps.

Table 9.16

Bit encoding for 5.5 Mbps and 11 Mbps CCK transmissions

Data Rate, Mbps	*CCK encoded bit*	*DQPSK encoded bits*
5.5	2	2
11	6	2

The chipping stream is used to phase modulate the carrier via phase shift keying. Differential Binary Phase Shift Keying (DBPSK) is used to achieve 1 Mbps and Differential Quadrature Phase Shift Keying (DQPSK) is used to achieve 2 Mbps. Fig 9.62 shows a typical 1 or 2 Mbps DSSS signal having a major lobe bandwidth of ±11 MHz (–30db). The first minor sidelobe is down at least 30 db and the second minor sidelobe is down 50 db as required by Part 15 rules.

The higher data rates specified in 802.11b are achieved by using a different pseudo-random code known as a Complimentary Sequence. Recall the 11 bit Barker code can encode one data bit. The 8 bit Complimentary Sequence can encode 2 bits of data for the 5.5 Mbps data rate or 6 bits of data for the 11 Mbps data rate. This is known as Complimentary Code Keying (CCK). Both of these higher data rates use DQPSK for carrier modulation. DQPSK can encode 2 data bits per transition. **Table 9.16** shows how 4 bits of the data stream are encoded to produce a 5.5 Mbps data rate and 8 bits are encoded to produce an 11 Mbps data rate. There are 64 different combinations of the 8 bit Complimentary Sequence that have the mathematical properties that allow easy demodulation and interference rejection. At 5.5 Mbps only four of the combinations are used. At 11 Mbps all 64 combinations are used.

As an example, for an input data rate of 5.5 Mbps, four bits of data are sampled at the rate of 1.375 Million samples per second. Two input bits are used to select 1 of 4 eight-bit CCK sequences. These 8 bits are clocked out at a rate of 11 Mbps. The two remaining input bits are used to select the phase at which the 8 bits are transmitted. **Fig 9.65A** shows a conceptual block diagram of a 5.5-Mbps CCK transmitter modulator while Fig 9.65B shows an 11-Mbps modulator.

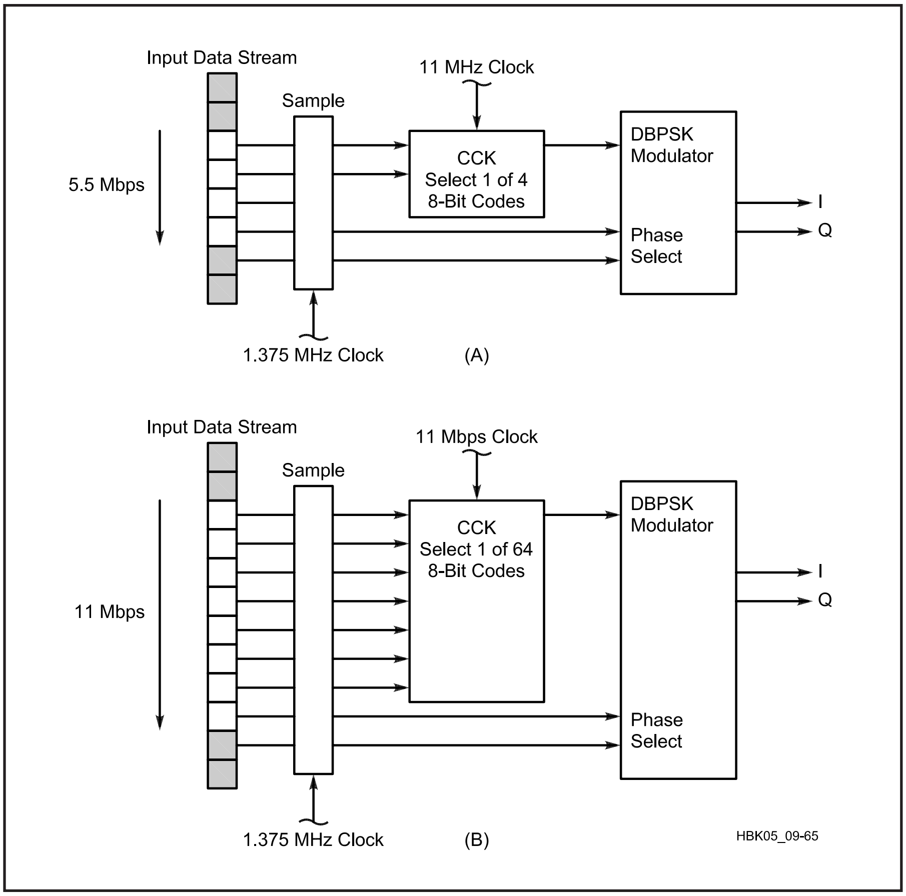


Fig 9.65 — Conceptual block diagram of a modulator for a CCK Spread Spectrum transmitter. (A) 5.5 Mbps data rate. (B) 11 Mbps data rate. See text.

Orthogonal Frequency Division Multiplexing

OFDM provides its spreading function by transmitting the data simultaneously on multiple carriers. 802.11g and 802.11a specify 20 MHz wide channels with 52 carriers spaced every 312.5 KHz. Of the 52 carriers, four are non-data pilot carriers that carry a known bit pattern to simplify demodulation. The remaining 48 carriers are modulated at 250 Kilo transitions per second. Taking all 48 transitions in parallel is known as a symbol. That is, at any given instant in time 48 bits of data are being transmitted.

The term *orthogonal* is derived from the fact that these carriers are positioned such

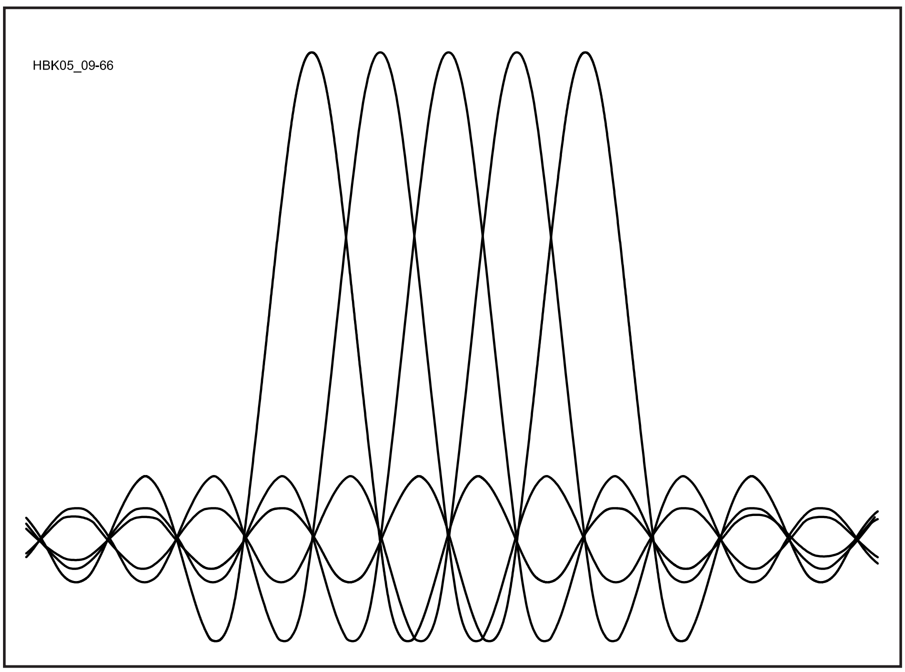


Fig 9.66 — Five OFDM modulated subcarriers with their frequencies spaced to provide orthogonality. See text.

Table 9.17

Summary of the modulation techniques used by OFDM to achieve the different data rates

Data Rate, Mbps	*Modulation*	*Coding Rate, R*
6	BPSK	1/2
9	BPSK	3/4
12	QPSK	1/2
18	QPSK	3/4
24	16QAM	1/2
36	16QAM	3/4
48	64QAM	2/3
54	64QAM	3/4

that they do not interfere with one another. The center frequency of one carrier's signal falls within the nulls of the signals on either side of it. **Fig 9.66** illustrates how the carriers are interleaved to prevent intercarrier interference.

OFDM radios can be used to transmit data rates of 6, 9, 12, 18, 24, 36, 48 and 54 Mbps as specified by both 802.11a and 802.11g. In order to transmit at faster and faster data rates in the same 20 MHz channel different modulation techniques are employed: BPSK, QPSK, 16QAM and 64QAM. In addition, some of the bits transmitted are used for error correction so the raw data rates could be reduced by up to half of what they would be without error correction. For instance, assuming BPSK (1 bit per carrier) and assuming half the bits are used for error correction (known as the coding rate, R); the resulting data rate would be 6 Mbps.

48 carriers × 1 bit per carrier × ½ R = 24 bits (effective)

24 bits × 250 Kilo transitions per second = 6 Mbps

Table 9.17 shows a complete list of the modulation methods and coding rates employed by OFDM. The higher data rates will require better signal strength to maintain error free reception due to using few error correction bits and more complex modulation methods.

Notes

[1]"Poisson, Shannon and the Radio Amateur," Proceedings of the IRE, Dec 1959.

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High Speed Multimedia Radio

In January 2001, the ARRL Board of Directors voted unanimously that the ARRL should proceed with the development of High Speed Digital Networks for the Amateur Service. The ARRL President was to appoint a group of individuals knowledgeable in the field from the international Amateur-Radio community and industry. The group would report to the Technology Task Force (TTF). The TTF established the High Speed Multimedia (HSMM) Working Group with John Champa, K8OCL, as its chairman. Champa moved quickly to identify two initial goals for the new working group, so as to immediately begin the development of such high speed digital Amateur Radio networks:

1. Encourage the amateur adoption and modification of commercial off-the-shelf (COTS) IEEE 802.11 spread spectrum hardware and software for Part 97 uses.

2. Encourage or develop other high-speed digital radio networking techniques, hardware, and applications.

These efforts were rapidly dubbed HSMM Radio. Although initially dependent on adaptation of COTS 802.11 gear to Part 97, it is obvious from these goals that HSMM radio is not a specific operating mode, but more of a direction or driving force within Amateur Radio.

Furthermore, in HSMM radio, the emphasis has shifted away from primarily keyboard radio communication, as in conventional packet radio, to multimedia radio: Simultaneous voice, video, data, and text over radio.

In HSMM radio these individual mediums have different names, much like their Internet counterparts. For example, voice modes, although technically digital voice, are most often called streaming audio. However, since it is two-way voice over an IP network similar to the direction being taken by contemporary commercial telephony technology, the same technology use to link many amateur radio repeaters over the Internet, the name voice-over-IP (VoIP) may be more appropriate.

Video modes, although sometimes called amateur digital video (ADV), are also known as streaming video. Again, perhaps the commercial term for such two-way video QSOs may be more appropriate: IPVC (IP videoconferencing).

Text exchanges via a keyboard are often used in HSMM radio, but they are similarly called by their Internet or Packet Radio name: Chat mode. File transfers using FTP can also be done just as on the Internet. This combination of Internet terminology, coupled with this dramatic shift in emphasis within amateur radio from traditional analog point-to-point radio toward networked digital radios, has resulted in many amateurs nick naming HSMM radio *The Hinternet.* Although the name implies some under-dog status to some, the name seems to be sticking.

HSMM RADIO APPLICATIONS

HSMM radio has some unique ham radio networking applications and operational practices that differentiate the Hinternet from normal Wi-Fi hotspots at coffee houses and airports which you may have read about in the popular press. HSMM radio techniques are used, for example, for system RC (remote control) of amateur radio stations.

In this day of environmentally sensitive neighborhoods, one of the greatest challenges, particularly in high density, residential areas, is constructing ham radio antennas, particularly high tower mounted HF beam antennas. In addition, such amateur installations represent a significant investment in time and resources. This burden could be easily shared among a small group of friendly hams, a radio club or repeater group.

Implementing a link to a remote HF station via HSMM radio is easy to do. Most computers now come with built-in multimedia support. Most amateur radio transceivers are capable of PC control. Adding the radio networking is relatively simple. Most HSMM radio links use small 2.4 GHz antennas mounted outdoors or pointed through a window. These UHF antennas are relatively small and inconspicuous when compared to a full-size 3-element HF yagi on tall steel tower.

For example, Darwin Thompson, K6USW, has performed remote control of a Kenwood TS-480SAT/HX transceiver, which can be controlled over a LAN and the Internet, or in this case the Hinternet. The Kenwood International website provides two programs for the TS-480SAT/HX at: **www.kenwood.net/indexKenwood.cfm?do=SupportFileCategory&FileCatID=3**.

The ARHP-10 is the radio host program. It operates the computer attached to the transceiver. Just follow the instructions included with the software to make the cables to interface the radio to your computer. The ARCP-480 is the radio control software. ARCP-480 operates the computer at the other end of the remote control link. By attaching a suitable headset to this remote PC, the operator now has full control of the transceiver via the HSMM radio link and can use voice-over-Internet-protocol (VoIP) to transmit and receive audio.

A ham does not have to have an antenna-unfriendly homeowners association (HOA) or a specific deed restriction problem to put RC via HSMM radio to good use. This system RC concept could be extended to other types of amateur radio stations. For example, it could be used to link a ham's home to a shared high performance amateur radio DX station, EME station, or OSCAR satellite ground station for a special event or on a regular basis.

SHARED HIGH-SPEED INTERNET ACCESS

Sharing high-speed Internet access (Cable, DSL, etc.) with another ham is a popular application for HSMM radio. Half of the US population is restricted to slow dial-up Internet connections (usually around 20-40 Kbps) over regular analog telephone lines. Getting a high-speed Internet connection, even a shared one, can dramatically change the surfing experience! Just remember that if you use an HSMM radio to share high speed access to the Internet, which Amateur Radio has content restrictions, for example no commercial for-profit business e-mails, etc. Pop-up ads, although a nuisance, are not illegal. An example might be an amateur television station (ATV) transmitting an outdoor scene and inadvertently picking-up a billboard in the station camera. Such background sources are merely incidental to your transmission. They are not the primary purpose of your communications, plus they are not intended for rebroadcast to the public.

Just as on the Internet, it is possible to do such things as playing interactive games, complete with sound effects and full motion animation with HSMM radio. This can be lots of fun for new and old hams alike, plus it can attract others in the "Internet Generation" to get interested in amateur radio and perhaps become new radio club members. In the commercial world these activities are called "WLAN Parties". Such e-games are also an excellent method for testing the true speed of your station's Hinternet link.

HSMM RADIO IN EMERGENCY COMMUNICATIONS

There are a number of significant reasons and exciting new examples why HSMM radio is the way of the future for many Emergency Communications (EmComm) situations. These may or may not be under ARES or RACES auspices.

1. The amount of digital radio traffic on 2.4 GHz is increasing and operating under low powered, unlicensed Part 15 limitations cannot overcome this noise.
2. EmComm organizations increasingly need high-speed radio networks which can simultaneously handle voice, video, data, and text traffic.
3. The cost of a commercially installed high-speed data network can be more than emergency organizations and communities can collectively afford.
4. EmComm managers also know that they need to continuously exercise any emergency communications system and have trained operators for the system in order for it to be dependable.

Being able to send live digital video images of what is taking place at a disaster site to everybody on the HSMM radio network could be invaluable in estimating the severity of the situation, planning appropriate responding resources, and other reactions. The Emergency Operations Center (EOC) could actually see what is happening while it is happening. Submitting a written report while simultaneously talking to the EOC using Voice over IP (VoIP) would provide additional details.

With HSMM radio often all that would be needed to accomplish this in the field is a laptop computer equipped with a wireless local area network card (PCMCIA) *with* an external antenna jack. In HSMM radio jargon such a card is simply called a RIC (radio interface card). Connect any digital camera with a video output port or any webcam, and a headset to the laptop's sound card. Then connect the RIC to a short yagi antenna (typically 18 inches of antenna boom length) and point the antenna back to the EOC.

HSMM RADIO RELAY

There are a number of ways to extend the HSMM link. The most obvious means would appear to be to run higher power and place the antennas as high as possible, as is the case with VHF/UHF FM repeaters. In some densely populated urban areas of the country this approach with 802.11, at least in the 2.4 GHz band, may cause some interference with other users. Other means of getting greater distances using 802.11 on 2.4 GHz or other amateur bands should be considered. One approach is to use highly directive, high-gain antennas, or what is called the directive link approach.

Another approach used by some HSMM radio networks is what is called a low-profile radio network design. They depend on several low power sources and radio relays of various types. For example, two HSMM radio repeaters (known commercially as access points, or APs, about $100 devices) may be placed back-to-back in what is known as bridge mode. In this configuration they will simply act as an automatic radio relay for the high-speed data. Using a series of such radio relays on a series of amateur towers between the end-points of the link, it is possible to cover greater distances with relatively low power and yet still move lots of multimedia data.

BASIC HSMM RADIO STATION

How does one set up an HSMM radio base station? It is really very easy. HSMM radio amateurs can go to any electronics outlet or office supply store and buy commercial off-the shelf (COTS) Wireless LAN gear, either IEEE 802.11b or IEEE 802.11g. They then connect external outdoor antennas. That is all there is to it.

There are some purchasing guidelines to follow. First, decide what interfaces you are going to need to connect to your computer. Equipment is available for all standard computer interfaces: Ethernet, USB, and PCMCIA. If you use a laptop in your station, get the PCMCIA card. Make certain it is the type with an external antenna connection. If you have a PC, get the Wireless LAN adaptor type that plugs into either the USB port or the RJ45 Ethernet port. Make certain it is the type that has a removable rubber duck antenna or external antenna port! The included directions will explain how to install these devices.

This is the core of any HSMM radio station. It is a computer-operated HSMM 2.4 GHz radio transceiver and it will probably cost about $60 to $80. Start off teaming up with a nearby ham radio operator. Do your initial testing in the same room together. Then as you increase distances going toward your separate station locations, you can coordinate using a suitable local FM simplex frequency. Frequently hams will use 146.52 MHz or 446.00 MHz, the National FM Simplex Calling Frequencies for the 2-m and 70-cm bands, for voice coordination. More recently, the HSMM radio operators have tended to use 1.2 GHz FM transceivers and handheld (HT) radios. The 1.2 GHz amateur band more closely mimics the propagation characteristics of the 2.4 GHz amateur band. The rule of thumb being, if you can not hear the other station on the 1.2 GHz FM radio, you probably will not be able to link up the HSMM radios.

Hams frequently ask why 802.11 transmitter output and receiver sensitivity are stated typically in dBm. The simple answer is that this convention simplifies certain calculations. For transmitter output, convert dBm to power using the formula for dB. The reference power level is 1 mW. That means that +10 dBm = 10 mW and +20 dBm = 100 mW.

For receive, it's a bit more complicated if you want the more familiar units. First, calculate power level and then covert that to voltage across 50 Ω. A good RIC receiver is able to receive down to –96 dBm. That would be equal to 0.00000000025 mW, which is 3.54 μV across 50 Ω.

HSMM RADIO REPEATERS

Access Points

What hams would call a repeater, and computer buffs would call a hub, the Wi-Fi industry refers to as a *radio access point*, or simply *AP*. This is a device that allows several amateur radio stations to share the radio network and all the devices and circuits connected to it.

An 802.11b AP will sell for about $80 and an 802.11g AP for about $100. The AP acts as a central collection point for digital radio traffic, and can be connected to a single computer or to another radio or wired network.

The AP is provided with an SSID, which is the station identification it constantly broadcasts. For ham purposes, the SSID can be set as your call sign, thus providing automatic, and constant station identification. To use an AP in a radio network the wireless computer users have to exit ad-hoc mode and enter what is called the infrastructure mode, in their operating software.

Infrastructure mode requires that you specify the radio network your computer station is intended to connect to, so set your computer station to recognize the SSID you assigned to the AP (yours or another ham's AP) to which you wish to connect.

Point-To-Point Links

The AP can also be used as one end of a radio point-to-point network. If you wanted to extend a radio network connection from one location to another, for example in order to remotely operate an HF station, you could use an AP at the network end and use it to communicate to a computer at the remote station location.

An AP allows for more network features and improved information security than provided by ad-hoc mode. Most APs provide DHCP service, which is another way of saying they will automatically assign an internet (IP) address to the wireless computers connected to the radio network. In addition, they can provide filtering which allows only known users to access the network.

Back panel of a typical HSMM-style repeater. This device is known commercially as a wireless access point (AP). It is essentially a computer wireless network hub to enable multiple radio stations to share the various resources of the network. This particular model is a Cisco Model 1200. Note that the left, or secondary antenna's rubber duck has been removed from the TNC connector to show that the connector is of the reverse polarity (RP) type. This is designated as a female TNC/RP connector. Manufacturers of 802.11 gear typically install a RP-type of some type connector to prevent FCC Part 15 unlicensed users from employing their equipment in a non-certified manner. Of course, this is not an issue for licensed Part 97 users, however as in this case, a male RP-type plug will be required in order to connect the device to an outside antenna. The provision of a secondary antenna is to provide space diversity, which helps reduce the negative impact of multipath propagation of the radio signals. The secondary antenna may be ignored when connecting the primary antenna to a single outside antenna, especially if it is a highly directive antenna, which would help reduce multipath effects. (Photo: John Champa, K8OCL).

SS and HSMM Glossary

Ad Hoc Mode—An operating mode of a client RIC that allows it to associate directly with any other RIC without having to go through an Access Point. See Infrastructure mode.

APRS—Automatic Position Reporting System

Association—The service used to establish access point/station mapping and enable station use of the WLANs services in infrastructure mode.

Authentication—Process by which the wireless communications system verifies the identity of a user attempting to use a WLAN prior to the user associating with the AP.

Band-limited Gaussian Channel—A "brickwall" linear filter that is equal to a constant over some frequency band and equal to zero elsewhere, and by white Gaussian noise with a constant power spectrum over the channel bandwidth.

Barker Code—An 11-bit digital sequence used to modulate (spread) the input data stream. A one bit is represented by the sequence 10110111000 and a zero bit is represented by the sequence 01001000111.

CCK—Complimentary Code Keying. A spreading technique in which the input data stream is modulated with a digital sequence (the complimentary code) depending on the value of the data stream. In 802.11b, for example, the complimentary code consists of 64 eight-bit values. Six data bits from the input stream are used to select which of the complimentary codes is used to modulate the data. See Barker Code.

Correlation—A measure of how closely a signal matches a delayed version of itself shifted n units in time.

COTS—Commercial Off The Shelf equipment.

DBPSK—Differential Binary Phase Shift Keying. A method of modulating data onto a carrier by changing the phase of the carrier relative to its current phase. A binary "1" is represented by a +90 degree phase shift and a binary "0" is represented by a 0 degree phase shift.

DHCP—Dynamic Host Configuration Protocol. A protocol used by a client computer to obtain an IP address for use on a network.

DSSS—Direct Sequence Spread Spectrum. A spread spectrum system in which the carrier has been modulated by a high speed spreading code and an information data stream. The high speed code sequence dominates the "modulating function" and is the direct cause of the wide spreading of the transmitted signal. (Title 47, Chapter I, Part 2, subpart A, section 2.1 Terms and Definitions).

DQPSK—Differential Quadrature Phase Shift Keying. A method of modulating data onto a carrier by changing the phase of the carrier similar to DBPSK except that two bits can be represented by a single phase shift such as following this scheme:

2-Bit Value	*Phase Shift (degrees)*
00	0
01	+90
10	–90
11	180

FHSS—Frequency Hopping Spread Spectrum. A spread spectrum system in which the carrier is modulated with the coded information in a conventional manner causing a conventional spreading of the RF energy about the frequency carrier. The frequency of the carrier is not fixed but changes at fixed intervals under the direction of a coded sequence. The wide RF bandwidth needed by such a system is not required by spreading of the RF energy about the carrier but rather to accommodate the range of frequencies to which the carrier frequency can hop. The test of a frequency hopping system is that the near term distribution of hops appears random, the long term distribution appears evenly distributed over the hop set, and sequential hops are randomly distributed in both direction and magnitude of change in the hop set. (Title 47, Chapter I, Part 2, subpart A, section 2.1 Terms and Definitions).

GFSK—Gaussian Frequency Shift Keying. A method of modulating data onto a carrier by changing the frequency of the carrier. The frequency is gradually changed as opposed to FSK in which the frequency is abruptly changed. This has the advantage of reducing the occupied bandwidth of the signal. 2GFSK represents a bit by changing between two frequencies. 4GFSK represents two bits by changing among four frequencies.

GPS—Global Positioning System

IEEE—Institute of Electrical and Electronic Engineering

IEEE 802.11—An IEEE standard specifying FHSS and DSSS in the 2.4 GHz band at 1 Mbps and 2 Mbps data rates. 802.11 is also used as a general term for all spread spectrum devices operating under Part 15. For example "The 802.11 network" could be referring to a collection of RICs and APs using 802.11b and 802.11g based devices.

MOBILE HSMM OPERATING

When hams use the term mobile HSMM station what they are normally talking about is a wireless computer set-up in their vehicle to operate in a stationary portable fashion. Nobody is suggesting that you try to drive a vehicle and look at a computer screen at the same time. That could be very dangerous, and is probably illegal. So unless you have somebody else to drive the vehicle keep your eyes on the road and not on the computer screen.

What sort of equipment is needed to operate an HSMM mobile station?

- Some type of portable computer, such as a laptop. Some hams use a PDA, notebook, or other small computing device. The operating system can be *Microsoft Windows*, *Linux*, or *Mac OS*, although *Microsoft XP* offers some new and innovative WLAN functionality.
- Some type of radio software hams would call an automatic monitor, and computer buffs would call a sniffer utility. The most common type being used by hams is Marius Milner's *Network Stumbler for Windows* frequently called, *NetStumbler*. All operating systems have monitoring programs available. *Linux* has *Kismet*; *MAC OS* has *MacStumbler*. Marius Milner has a version for the PocketPC which is called *MiniStumbler*.
- A RIC (Radio Interface Card = PCMCIA WiFi computer adapter card with external antenna port) which is supported by the monitoring utility you are using. The most widely supported RIC is the Orinoco line. The Orinoco

IEEE 802.11a—An IEEE standard specifying OFDM in the 5.8 GHz band at 6, 12, 16, 24, 36, 48, and 54 Mbps data rates.

IEEE 802.11b—An IEEE standard specifying DSSS in the 2.4 GHz band at 5.5 and 11 Mbps data rates in addition to being backward compatible with DSSS at 1 and 2 Mbps specified in 802.11.

IEEE 802.11g—An IEEE standard specifying OFDM in the 2.4 GHz band 6, 12, 16, 24, 36, 48, and 54 data rates in addition to being backward compatible with DSSS at 1, 2, 5.5, and 11 Mbps specified in 802.11b.

IEEE 802.11n—An unreleased (as of this publication date) IEEE standard specifying data rates up to 250 Mbps and being backward compatible with 802.11a and 802.11g.

IEEE 802.16—An unreleased IEEE standard specifying wireless last-mile broadband access in the Metropolitan Area Network (MAN). Also known as WiMAX.

ISM—Industrial, Scientific, and Medical. Specific frequency bands authorized by Part 18 rules for non-communication equipment such as microwave ovens, RF lighting, etc. The ISM spectrum where spread spectrum is allowed is located at 2.4 – 2.5 GHz and 5.725 – 5.875 GHz band.

Infrastructure Mode—An operating mode of a client RIC that requires all communications to go through an Access Point.

NMEA 183—National Marine Electronics Association interface standard which defines electrical signal requirements, data transmission protocol and time, and specific sentence formats for a 4800-baud serial data bus.

OFDM—Orthogonal Frequency Division Multiplexing. A modulation method in which the communication channel is divided into multiple subcarriers each being individually modulated. While not meeting the Part 2 definition of spread spectrum the FCC has given specific authorization for OFDM systems.

Orthogonal—A mathematical term derived from the Greek word orthos, which means straight, right, or true. In terms of RF, orthogonal applies to the frequencies of the subcarriers which are selected so that at each one of these subcarrier frequencies, all the other subcarriers do not contribute to the overall waveform. In other words, the subcarrier channel is independent of the other channels. Some hams explain OFDM as being a bunch of independent PSK channels, side-by-side.

PCMIA—Personal Computer Manufacturer Interface Adaptor.

Pigtail—A short piece of coaxial cable with a appropriate connectors to match the RIC antenna port and an external antenna system.

QAM—Quadrature Amplitude Modulation. A method of modulating data onto a carrier by changing both the phase and amplitude of the carrier. In its simplest form, 2QAM, the modulation is identical to BPSK. 16QAM represents 4 bits by changing among 16 phase/amplitude states. 64QAM represents six bits by changing among 64 phase/amplitude states.

RIC—Radio Interface Card. The radio equivalent of a Network Interface Card (NIC).

RLAN—Radio Local Area Network. See also WLAN.

RMAN—Radio Metropolitan Area Network

Spread Spectrum—An information bearing communications system in which: (1) Information is conveyed by modulation of a carrier by some conventional means, (2) the bandwidth is deliberately widened by means of a spreading function over that which would be needed to transmit the information alone. (Title 47, Chapter I, Part 2, subpart A, section 2.1 Terms and Definitions).

SSID—Service Set Identifier. A unique alphanumeric string used to identify a WLAN, or in the case of HSMM, RLAN, by using the individual callsign and perhaps the name of the amateur radio club or repeater group.

UNII—Unlicensed National Information Infrastructure. The UNII spectrum is located at 5.15 - 5.35 GHz, 5.725 - 5.825 GHz, and the recently added 5.470-5.725 GHz band.

USB—Universal Serial Bus.

VPN—Virtual Private Network.

WEP—Wired Equivalent Privacy. An encryption algorithm used by the authentication process for authenticating users and for encrypting data payloads over a WLAN.

WEP Key—An alphanumeric character string used to identify an authenticating station and used as part of the data encryption algorithm.

Wi-Fi—Wireless Fidelity. Refers to products certified as compatible by the Wi-Fi Alliance. See **www.wi-fi.org**. This term is also applied in a generic sense to mean any 802.11 capability.

WISP—Wireless Internet Service Provider

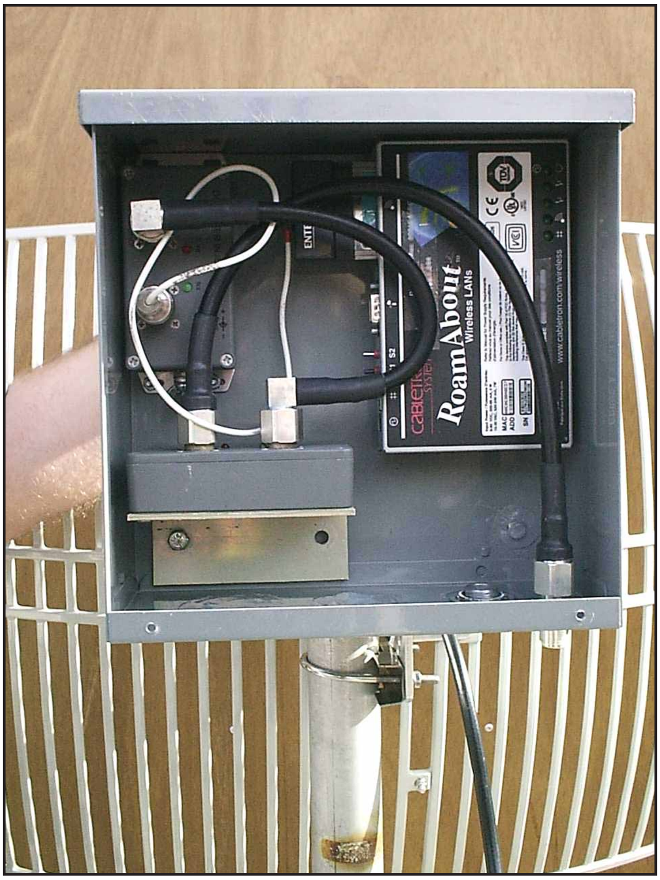
WLAN—Wireless Local Area Network.

line is inexpensive and fairly sensitive.

- An external antenna attached to your RIC. This is often a magnetically mounted omni-directional vertical antenna on the vehicle roof, but small directional antennas pointed out a window or mounted on a small ground tripod are also frequently used.
- A pigtail or short strain relief cable will be needed to connect from the RIC antenna port to the N-series, RP/TNC or other type connector on the external antenna.
- A GPS receiver that provides NMEA 183 formatted data and computer interface cable. This allows the monitoring utility to record where HSMM stations are located in on a map just as in APRS. GPS capability is optional, but just as with APRS capability, it makes the monitored information much more useful for locating HSMM stations

Warning

While operating your HSMM mobile station, if you monitor an unlicensed Part 15 station (non-ham), some types of Wi-Fi equipment will automatically associate or link to such stations, if they are not encrypted, and many are not (i.e., WEP is not enabled). Although Part 15 stations share the 2.4-GHz band on a non-interfering basis with hams, they are operating in another service. In another part of this section we will provide various steps you can take to prevent Part 15 stations from automatically linking with HSMM stations. So in like manner, except in the case of a communications emergency, we recommend that you do not use a Part 15 station's Internet connection for any ham purpose.



View of HSMM equipment (802.11b) inside an antenna-mounted NEMA-4 box. Mounting the equipment at the end of the dish antenna's pigtail significantly reduces feed line losses and greatly enhances the performance of the station. The box contains both a bridge, and a 500 mW bidirectional amplifier or BDA (lower left). Amplifier power is provided by the power insertion module seen in the upper left corner of the enclosure. (Photo: John Champa, K8OCL).

FM voice repeater, amateur television (ATV), and HSMM antennas mounted on a hydraulically operated mast. This portable installation was used to provide shared high-speed Internet access and other special communications support to the many hams attending the 2003 Pacificon Hamvention in San Ramon, CA. The HSMM station is also used to provide streaming video or amateur digital video (ADV) to the Mount Diablo ARC's analog FM ATV repeater on the nearby mountain. (Photo: John Champa, K8OCL)

HSMM AREA SURVEYS

Both licensed amateurs and unlicensed (Part 15) stations use the 2.4-GHz band. To be a good neighbor, find out what others are doing in your area before designing your community HSMM radio network. This is easy to do using IEEE 802.11 modulation. Unless it has been disabled, an active repeater (AP) is constantly sending out an identification beacon known as the SSID. In HSMM practice this is simply the ham station call sign (and perhaps the local radio club name) entered into the software configuration supplied with the CD that comes with the repeater. So every HSMM repeater is also a continuous beacon.

A local area survey using appropriate monitoring software, for example free *NetStumbler* software downloaded and running on your PC (**www.netstumbler.com/index.php**), is recommended prior to starting up any HSMM operations. Slew your station's directional antenna through a 360° arc, or drive your HSMM mobile station (as described earlier) around your local area.

This HSMM area survey will identify and automatically log most other 802.11 station activity in your area. There are many different ways to avoid interference with other users of the band when planning your HSMM operating. For example, moving your operating frequency 2-3 channels away from the other stations is often sufficient. Why several channels and not just one? Because the channels have considerable overlap. Why this situation exists is beyond the scope of this section, but here is the situation: The channels are only 5 MHz wide, but the DSSS or OFDM modulation of 802.11 is 22 MHz wide. Commercial users often recommend moving 5 channels away from the nearest AP to completely avoid interference. There are six channels within the amateur 2.4 GHz band, but there are problems for hams with two of them. Channel 1 centered on 2412 MHz overlaps into OSCAR satellite downlink frequencies. Channel 6 centered on 2437 MHz is by far the most common out-of-the-box default channel for the majority of WLAN equipment sold in the US, so that often is not the best choice. Subsequently, most HSMM radio groups end up using either channel 3 or channel 4, depending on their local situation. Again, an area survey is recommended before putting anything on the air.

However, because of the wide sidebands used in these inexpensive broad banded 802.11 modulations, even moving 2-3 channels away from such activity may not be enough to totally avoid interference, especially if you are running what in HSMM is considered high power (typically 1800-mW RF output—more on that subject later). You may have to take other steps. For example, you may use a different polarization with your antenna system. Many HSMM stations use horizontal polarization because much 802.11 activity in their area is primarily vertically polarized.

HSMM ANTENNA SYSTEMS

There are a number of factors that determine the best antenna design for a specific HSMM radio application. Most commonly, HSMM stations use horizontal instead of vertical polarization.

Furthermore, most HSMM stations use highly directional antennas, instead of omnidirectional antennas. Directional antennas provide significantly more gain and thus better signal-to-noise ratios, which in the case of 802.11 modulations, means higher rate data throughput. Higher data throughput, in turn, translates into more multimedia radio capability.

Highly directional antennas also have many other advantages. Such antennas can allow two hams to shoot over, or shoot around, or even shoot between other wireless stations on the band.

However, the nature of 802.11 modulations coupled with the various configurations of many COTS devices allows hams to economically experiment with many other fascinating antenna designs. Such unique antenna system designs can be used to simply help avoid interference, or to extend the range of HSMM links, or both.

Space Diversity

Some APs and some RICs have space diversity capability built-into their design. However, it is not always operated in the same fashion, so check the literature or the website of your particular device's manufacturer to be certain how the dual antenna ports are used. For example, many APs come equipped with two rubber ducky antennas and two antenna ports. One antenna port may be the primary and the other port the secondary input to the transceiver. Which signal input is used may depend on which antenna is providing the best S/N ratio at that specific instant. Experimentation using two outside high-gain antennas spaced 10 or more wavelengths apart (that is only about one meter on the 2.4-GHz band) may be very worthwhile in improving data throughput on long links. Such extended radio paths tend to experience more multi-path signal distortion. This multi-path effect is caused by multiple signal reflections off various objects in the path of the linking signal. The use of space diversity techniques may help reduce this effect and thus improve the data rate throughput on the link. Again, the higher the date rates the more multimedia radio techniques that can be used on that network.

Circular Polarization

The use of circular polarization created by using helical antennas, patch fed-points on dish antennas, or other means, warrants further study by radio amateurs. Remember this is high-speed digital radio. To avoid symbol errors, circularly polarized antennas should be used at both ends of the link. Also, be certain that the antennas are of the same handedness, for example right-hand circular polarization (RHCP). The ability of circular polarization to enhance propagation of long-path HSMM radio signals should not be overlooked.

Circularly Polarized Space Diversity

A combination or hybrid antenna design combining both circularly polarized antennas and space diversity could yield some extraordinary signal propagation results. For example, it has been suggested that perhaps using a RHCP for one antenna and LHCP for the other antenna, especially using spacing greater than 10 wavelengths, in such a system could provide a nearly "bullet-proof" design. Only actual field testing of such designs under different terrain features would reveal such potential.

Mixed Antenna Design Problem

In conventional wide-bandwidth analog radio antennas systems as long as both antennas at both ends of a radio link are of broad bandwidth design and both antennas are of the same polarization, all is fine. While this may be true for wide bandwidth analog signals such as amateur television VSB (vestigial sideband) signals or FM ATV signals, it may not be true for broad bandwidth high-speed digital signals.

First, 802.11 modulations produce very broadband signals, typically 22 MHz. Secondly, the evidence to date seems to indicate that the use of a same polarized antenna with one type of feed point at one end of the link and the use of a same polarized antenna with a different type of feed point at the other end of the link, may introduce a problem with high-speed digital signals. A common example of this potential mixed antenna issue would be if one HSMM station uses a horizontally polarized linear Yagi, while the other HSMM station at the opposite end of the link uses a horizontally polarized loop Yagi. Here is another typical situation. Let us say, for example, one ham at one end of the radio path uses a dish antenna with a horizontal dipole feed-point. The other ham at the opposite end of the path uses a horizontally polarized loop yagi. Both antennas have gain, both antennas are broadband width designs, and both antennas are horizontally polarized. Nonetheless, the hams may experience higher BER (bit error rate) because of symbol errors caused by the different manner in which the two antennas manipulate the digital radio signal wave front. Further radio amateur experimentation with HSMM radio signals is warranted to determine the full impact on the radio link of using mixed antenna types.

RUNNING HIGHER POWER

Hams often ask why operate 802.11 modes under licensed Part 97 regulations when we may also operate such modes under unlicensed Part 15 regulations, and without the content restrictions imposed on the Amateur Radio service?

A major advantage of operating under Amateur Radio regulations is the feasibility of legally operating with more RF power output and larger, high-gain directive antennas. These added capabilities enable hams to increase the range of their operations. The enhanced signal-to-noise ratio provided by running high power will also allow better data packet throughput. This enhanced throughput, in turn, enables more multimedia experimentation and communication capability over such increased distances.

In addition, increasing the effective radiated power (ERP) of an HSMM radio link provides for more robust signal margins and consequently a more reliable link. These are important considerations in providing effective emergency communications services and accomplishing other important public service objectives in a band increasingly occupied by unlicensed stations and other noise sources.

It should be noted that the existing FCC Amateur Radio regulations covering spread spectrum (SS) at the time this is being written were implemented prior to 802.11 being available. The provision in the existing regulations calling for automatic power control (APC) for RF power outputs in excess of 1 watt is not considered technologically feasible in the case of 802.11 modulations for various reasons. As a result the FCC has communicated to the ARRL that the APC provision of the existing SS regulations are therefore not applicable to 802.11 emissions under Part 97.

However, using higher than normal output power in HSMM radio, in the shared 2.4-GHz band, is also something that should be done with considerable care, and only after careful analysis of link path conditions and the existing 802.11 activity in your area. Using the minimum power necessary for the communications is the law and has always been a good operating practice for hams.

There are also other excellent and far less expensive alternatives to running higher power when using 802.11 modes. For examples, amateurs are also allowed to use higher gain directional antennas. Such antennas increase both the transmit and receive effectiveness of the transceiver. Also, by placing equipment as close to the station antenna as possible, a common amateur OSCAR satellite and VHF/UHF DXing technique, the feedline loss is significantly reduced. This makes the HSMM station transceiver more sensitive to received signals, while also getting more of its transmitter power to the antenna.

Only after an HSMM radio link analysis (see the link calculations portion at **www.arrl.org/hsmm/** or go to **logidac.com/gfk/80211link/pathAnalysis.html**) clearly indicates that additional RF output power is required to achieve the desired path distance, should more power output be considered.

At that point in the situation analysis, if higher power is required, what is needed is called a bi-directional amplifier (BDA). This is a super fast switching pre-amplifier / amplifier combination that is usually mounted at the end of the antenna pig-tail near the top of the tower or mast. A reasonably priced 2.4-GHz, 1800-mW output BDA is available from the FAB Corporation (**www.fab-corp.com**). It is specifically designed for amateur HSMM radio experimenters. Be certain to specify HSMM when placing your order. Also, to help prevent unauthorized use by unlicensed Part 15 stations, the FAB Corp may request a copy of your amateur license to accompany the order, and they will only ship the BDA to your licensee address as recorded in the FCC database.

This additional power output of 1800 mW should be sufficient for nearly all amateur operations. Even those supporting EmComm which may require more robust signal margins than normally needed by amateurs, seldom will require more power output than this level. If still greater range is needed, there are other less expensive ways to achieve such ranges as described in the section HSMM Radio Relays.

When using a BDA and operating at higher than normal power levels on the channels 2 through 5 recommended for amateur radio use (these channels are arbitrary channels intended for Part 15 operation and are not required for amateur radio use, but they are hard-wired into the gear so we are stuck with them), you should also be aware of the sidebands produced by 802.11 modulation. These sidebands are in addition to the normal 22 MHz wide spread spectrum signal. Accordingly, if your HSMM radio station is next door to an OSCAR ground station or other licensed user of the band, you may need to take extra steps in order to avoid interfering with them. The use of a tuned output filter may be appropriate in order to avoid causing QRM. Even when operating on the recommended channels in the 2-5 range, whenever you use higher than normal power, some of your now amplified sidebands may go outside the amateur band which stops at 2450 MHz. So from a practical point of view, whenever the use of a BDA is required to achieve a specific link objective, it is a good operating practice to install a tuned filter on the BDA output. Such filters are not expensive and they're readily available from several commercial sources. It should also be noted that most BDAs currently being marketed, while suitable for 802.11b modulation, they are often not suitable for the newer, higher speed 802.11g modulation.

There is one further point to consider. Depending on what other 802.11 operating may be taking place in your area, it may be a good practice to only run higher power when using directional or sectional antennas. Such antennas allow hams to operate over and around other licensed stations, but also including unlicensed Part 15 activity in your area which you may not wish to disrupt (a local school WLAN, WISP, etc). Again, before running high power, it is recommended that an area survey be conducted using a mobile HSMM rig as described earlier to determine what other 802.11 activity is in your area and what channels are in use.

INFORMATION SECURITY

An HSMM radio station could be considered a form of software defined radio. Your computer running the appropriate software combined with the RIC makes a single unit which is now your station HSMM transceiver. However, unlike other radios, your HSMM radio is now a networked radio device. It could be connected directly to other computers and to other radio networks, and even to the Internet. So each HSMM radio (PC + RIC + software) needs to be protected. There are at least two basic steps that should be taken with regards to all HSMM radios:

The PC should be provided with an anti-virus program**.** This anti-virus must be regularly updated to remain effective. Such programs may have come with the PC when it was purchased. If that is not the case, reasonably priced anti-virus programs are readily available from a number of sources.

Secondly, it is important to use a firewall software program on your HSMM radio. It is recommend that the firewall be configured to allow all outgoing traffic, but to restrict all incoming traffic without specific authorization. Commercial personal computer firewall products are available from Symantec, ZoneLabs and McAfee Network Associates.

Check this URL for a list of freeware firewalls for your personal computer: **www.webattack.com/freeware/security/fwfirewall.shtml**.

Check this URL for a list of shareware firewalls for your personal computer: **www.webattack.com/Shareware/security/swfirewall.shtml**.

Once a group of HSMM stations has setup and configured a repeater (AP) into a radio local area network (RLAN) then additional steps may need to be taken to restrict access to the repeater. Only Part 97 stations should be allowed to associate with the HSMM repeater. Remember, in the case of 802.11 modulations, the 2.4 GHz band is shared with Part 15 unlicensed 802.11 stations. How do you keep these unlicensed stations from automatically associating (auto-associate) with your licensed ham radio HSMM network?

Many times the steps taken to avoid interference with other stations also limits those other stations' capability to auto-associate with the HSMM repeater, and to improve the overall security of the HSMM station. For example, using a different antenna polarization than the Part 15 station, operating with a directional antenna oriented toward the desired coverage area rather than using an omnidirectional antenna, etc.

The most effective method to keep unlicensed Part 15 stations off the HSMM repeater is to simply enable the Wired Equivalent Protection (WEP) already built into the 802.11 equipment. The WEP encrypts or scrambles the digital code on the HSMM repeater based on the instruction or "key" given to the software. Such encryption makes it impossible for unlicensed stations not using the specific code to accidentally auto-associate) with the HSMM repeater.

The primary purpose of this WEP implementation in the specific case of HSMM operating is to restrict access to the ham network by requiring all stations to authenticate themselves. Ham stations do this by using the WEP implementation with the appropriate ham key. Hams are permitted by FCC regulations to encrypt their transmission in specific instances; however, ironically at the time of this writing, this is not one of them. Accordingly, for hams to use WEP for authentication and not for encryption, the key used to implement the WEP must be published. The key must be published in a manner accessible by most of the amateur radio community. This fulfills the traditional ham radio role as a self-policing service. The current published ham radio WEP key is available at the home page of the ARRL Technology Task Force High Speed Multimedia Working Group: **www.arrl.org/hsmm/**.

Before implementing WEP on your HSMM repeater be certain that you have checked the website to ensure that you are using the current published WEP key. The key may need to be occasionally changed.

HSMM FREQUENCIES

Up to this point all the discussion has been regarding HSMM radio operations on the 2.4-GHz amateur band. However, 802.11 modulations can be used on any amateur band above 902 MHz.

On the 902-MHz band using 802.11 modulations would occupy nearly the entire band. This may not be a problem in your area depending on the nature of the other existing users of the band in your area, either licensed or unlicensed. FM repeaters may not have a problem with sharing the frequency with 802.11 operations as they would likely just hear an 802.11 modulated signal as weak background noise, and the 802.11 modulation, especially the OFDM channels used by 802.11g, would simply work around the FM interference with little negative impact. There is some older 802.11 gear (FHSS) available on the surplus market for amateur experimentation. Alternatively, some form of frequency transverter may be used to take 2.4 GHz to the 902-MHz band.

The 1.2-GHz band has some potential for 802.11 experimenting. Some areas have several FM voice repeaters and even ATV FM repeaters on the band, but again these relatively narrow bandwidth signals would likely hear any 802.11 modulations as simply background noise. Looking at the potential interference from the HSMM perspective, even in the case of the FM ATV, it is unlikely the signal would significantly disrupt the 802.11 modulation unless the two signals were on exactly the same center frequency or at least with complete overlap in bandwidth. Keep in mind that the FM ATV signal is only several megahertz wide, but the 802.11 modulation is 22 MHz wide. For the analog signal to wipe out the spread spectrum signal, it would need to over-power or completely swamp the 802.11 RIC receiver's front end.

The 3.5-GHz band offers some real possibilities for 802.11 developments. Frequency transverters are available to get to the band from 2.4 GHz and there is little other activity on the band at this time. Developments in Europe of 802.16 with 108-mbps data throughput may make 3.5-GHz gear available for amateur experimentation in the US. Hams are investigating the feasibility of using such gear when it becomes available in the US for providing a RMAN or radio metropolitan area networks. The RMAN would be used to link the individual HSMM repeaters (AP) or RLANs together in order to provide county-wide or regional HSMM coverage, depending on the ham radio population density.

The 5.0-GHz band is also being investigated. The COTS 802.11a modulation gear has OFDM channels which operate in this amateur radio band. The 802.11a modulation could be used in a ham RLAN operating much as 802.11g is in the 2.4-GHz band. It is also being considered by some HSMM groups as a means of providing MAN links. This band is also being considered by AMSAT for what is known as a C-N-C transponder. This would be an HSMM transponder onboard probably a Phase 3 high-altitude or a Phase 4 geostationary OSCAR with uplink and downlink bandpass both within the 5-GHz amateur band. Some other form of modulation other than 802.11 would likely have to be used because of timing issues and other factors, but the concept is at least being seriously discussed.

RMAN link alternatives are also being tested by hams. One of these is the use of virtual private networks (VPN) similar to the method currently used to provide worldwide FM voice repeater links via the Internet. Mark Williams, AB8LN, of the HSMM Working Group is leading a team to test the use of various VPN technologies for linking HSMM repeaters.

HF is not being ignored either. It is possible that a modulation form which, while it is neither SS nor HSMM, might be able to produce data rates fast enough to efficiently handle e-mail type traffic on the HF bands while still occupying appropriate bandwidth. Such modulation would be helpful in an emergency with providing an outlet for RMAN e-mail traffic. Neil Sablatzky, K8IT, is leading a team of ham investigators on the HF and VHF bands.

Finally there are commercial products being developed such as the Icom D-STAR system which could readily be integrated into a RMAN infrastructure.

HSMM REFERENCES

Use of HSMM over Amateur Radio is a developing story. You can keep up with developments and find useful links to other data by visiting *ARRLWeb* at **arrl.org/hsmm/**.

For more details on using HSMM radio for remote control of stations, see the article "Remote-Control HF Operation over the Internet," by Brad Wyatt, K6WR, *QST*, November 2001 pp 47-48.

For guidelines on using e-games on-the air in Amateur Radio, see the HSMM column titled "Is (sic) All Data Acceptable Data" by Neil Sablatzky, K8IT, in the Fall 2003 issue of *CQ VHF*.

For more information regarding HSMM on future OSCAR satellites, see the *Proceedings of the AMSAT-NA 21st Space Symposium*, November 2003, Toronto, Ontario, Canada, especially the paper by Clark, Tom, W3IWI, "C-C RIDER, A New Concept for Amateur Satellites", available from ARRL.

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