Chapter 9

Modes and Systems

The various modes, modulation types and protocols we use partly reflect the different types of information we might wish to transmit, such as data, voice, image or even multimedia communications. Other considerations are the behavior of the radio link including fading, delay, Doppler frequency shift and distortion. We are also limited by regulatory restrictions such as bandwidth and following certain conventions or protocols. Some of the bands used by amateurs are wide and well behaved, such as VHF links over short paths. 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 frequency spectrum that a signal occupies. There are narrow-band modes, such as CW and PSK31, and wideband modes, such as TV and spread spectrum. Not all modes are permitted on all amateur bands. Wideband modes can be used only where the total width of the amateur allocation is sufficient to contain the wide signal. In addition, voluntary agreements and regulatory restrictions keep some wideband modes out of certain bands or subbands so that one station'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 bandwidth rules. The occupied bandwidth is determined not only by the mode being used, but by proper operation of that mode. Many of the permitted modes can become too wide when improperly adjusted. Perhaps the greatest source of conflict between ham operators is "splatter" or "key clicks" caused by overmodulated 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 mean power of the transmitted signal.

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 a major issue for most amateurs, although they rightly take pride in the clarity of their transmissions and some amateurs take audio quality quite seriously. *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 difficult 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 regulatory 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 that 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 regulatory and technical solutions.

Emission Classifications

Emissions are designated according to their classification and their necessary bandwidth. A minimum of three symbols is used to describe the basic characteristics of radio waves. Emissions are classified and symbolized according to the following characteristics:

I. First symbol—Type of modulation of the main carrier
II. Second symbol—Nature of signal(s) modulating the main carrier
III. Third symbol—Type of information to be transmitted
Note: A fourth and fifth symbol are provided for in the ITU Radio Regulations. Use of the fourth and fifth symbol is optional.
IV. Details of signal(s)
V. Nature of multiplexing

First symbol—type of modulation of the main carrier

(1) Emission of an unmodulated carrier .... N
(2) Emission in which the main carrier is amplitude-modulated (including cases where subcarriers are angle-modulated):
- Double sideband .... A
- Single sideband, full carrier .... H
- Single sideband, reduced or variable level carrier .... R
- Single sideband, suppressed carrier .... J
- Independent sidebands .... B
- Vestigial sideband C

(3) Emission in which the main carrier is angle-modulated:
- Frequency modulation .... F
- Phase modulation .... G

Note: Whenever frequency modulation (F) is indicated, phase modulation (G) is also acceptable.

(4) Emission in which the main carrier is amplitude and angle-modulated either simultaneously or in a pre-established sequence .... D
(5) Emission of pulses[1]
- Sequence of unmodulated pulses .... P
- A sequence of pulses:
- Modulated in amplitude .... K
- Modulated in width/duration L
- Modulated in position/phase .... M
- In which the carrier is angle-modulated during the period of the pulse .... Q
- Which is a combination of the foregoing or in produced by other means V

(6) Cases not covered above, in which an emission consists of the main carrier modulated, either simultaneously or in a pre-established sequence in a combination of two or more of the following modes: amplitude, angle, pulse .... W
(7) Cases not otherwise covered X

Second symbol—nature of signal(s) modulating the main carrier

(1) No modulating signal 0
(2) A single channel containing quantized or digital information without the use of a modulating subcarrier, excluding time-division multiplex 1
(3) A single channel containing quantized or digital information with the use of a modulating subcarrier, excluding time-division multiplex 2
(4) A single channel containing analog information 3
(5) Two or more channels containing quantized or digital information 7
(6) Two or more channels containing analog information 8
(7) Composite system with one or more channel containing quantized or digital information, together with one or more channels containing analog information 9
(8) Cases not otherwise covered X

Third symbol—type of information to be transmitted[2]

(1) No information transmitted .... N
(2) Telegraphy, for aural reception .... A
(3) Telegraphy, for automatic reception .... B
(4) Facsimile C
(5) Data transmission, telemetry, telecommand .... D
(6) Telephony (including sound broadcasting) .... E
(7) Television (video) .... F
(8) Combination of the above .... W
(9) Cases not otherwise covered X

Where the fourth or fifth symbol is used it shall be used as indicated below. Where the fourth or the fifth symbol is not used this should be indicated by a dash where each symbol would otherwise appear.

Fourth symbol—Details of signal(s)

(1) Two-condition code with elements of differing numbers and/or durations .... A
(2) Two-condition code with elements of the same number and duration without error-correction ...... B
(3) Two-condition code with elements of the same number and duration with error-correction C
(4) Four-condition code in which each condition represents a signal element (of one or more bits) .... D
(5) Multi-condition code in which each condition represents a signal element (of one or more bits) .... E
(6) Multi-condition code in which each condition or combination of conditions represents a character .... F
(7) Sound of broadcasting quality (monophonic) .... G
(8) Sound of broadcasting quality (stereophonic or quadraphonic) .... H
(9) Sound of commercial quality (excluding categories given in (10) and (11) below .... J
(10) Sound of commercial quality with frequency inversion or band-splitting .... K
(11) Sound of commercial quality with separate frequency-modulated signals to control the level of demodulated signal L
(12) Monochrome .... M
(13) Color .... N
(14) Combination of the above .... W
(15) Cases not otherwise covered X

Fifth symbol—Nature of multiplexing

(1) None .... N
(2) Code-division multiplex[3] C
(3) Frequency-division multiplex .... F
(4) Time-division multiplex .... T
(5) Combination of frequency-division and time-division multiplex .... W
(6) Other types of multiplexing X

[1] Emissions where the main carrier is directly modulated by a signal which has been coded into quantized form (eg, pulse code modulation) should be designated under (2) or (3).
[2] In this context the word “information” does not include information of a constant unvarying nature such as is provided by standard frequency emissions, continuous wave and pulse radars, etc.
[3] This includes bandwidth expansion techniques.

Emission, Modulation and Transmission Characteristics

Emission designators are generally expressed as characters representing the necessary bandwidth and emission classification symbols. *Necessary bandwidth* is expressed as a maximum of five numerals and one letter. The letter occupies the position of the decimal point and represents the unit of bandwidth, as follows: H = hertz, K = kilohertz, M = megahertz and G = gigahertz. For example, a bandwidth of 2.8 kHz is expressed as 2K8 or 2K80 and a bandwidth of 150 Hz is noted as 150H.

Emission classification symbols are (1) type of modulation of the main carrier, (2) nature of the signal(s) modulating the main carrier and (3) the information to be transmitted. They may be supplemented by (4) details of signal(s) and (5) nature of multiplexing, but the FCC does not require these. These designators are found in Appendix 1 of the ITU Radio Regulations, ITU-R Recommendation SM.1138 and in the FCC rules §2.201.

TRANSMISSION IMPAIRMENTS

In addition to attenuation of a signal from a transmitting station to a receiving station, the signal is subject to a variety of impairments. These include flat fading, frequency-selective fading, wave polarization rotation fading, Doppler shift, interference from other signals, atmospheric noise, galactic noise and manmade noise. Receiver thermal noise is not usually an issue at HF because external noise often dominates but can be a limiting factor at VHF and above.

Effect of the HF Path on Pulses

Digital signals, including Morse signals copied by ear, are transmissions in which the wave abruptly changes state. That means that something about them is varied in order to 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 HF digital system. VHF and UHF signals generally use more benign paths but some of the same principles important in HF will also apply to them.

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 letter "T" by its length, provided the keying speed is known beforehand. Using the ARRL-recommended values for shaping to prevent excessive key clicks, a single Morse "E" will appear as shown on the left in **Fig 9.1** as it leaves the transmitter. A ham living nearby would observe the

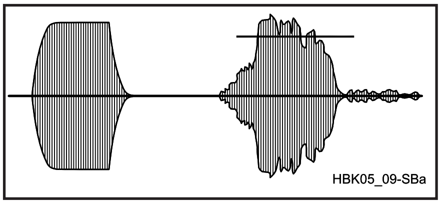

Fig 9.1—Morse "E" as transmitted (left) and received (right).

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 9.1**. Several things, however, distort the pulse. 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 simply deciding whether a pulse was sent or not, and its length. 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. Human AGC (automatic gain control) operates to provide a continually variable "decision threshold." If all else fails during on-air contacts, repeats can be requested.

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 smoothes 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 difficult to decode reliably.

The second oldest method of transmitting pulses is with frequency shift keying (FSK) where, instead of turning a signal on and off, it is made to jump between different frequencies to correspond to the "key-up" and "key-down" conditions. RTTY (radio teletype) 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 (plus 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 with FM analog transmissions, FSK and PSK tend to discriminate against amplitude noise, which is common on HF. Thus, amplitude changes resulting from fading and/or 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. You need only observe the little phase "compass" included in some PSK31 software to be aware that even when transmitting an idle tone, the received signal's phase will jump around.

Doppler shifts introduce noise into phase- and frequency-shift systems. If we are counting on seeing a certain frequency shift as a signal, you 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. The general 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 unusual multipath propagation conditions smear the characters together. A machine can operate at speeds where even normal multipath can smear the characters, not just in 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

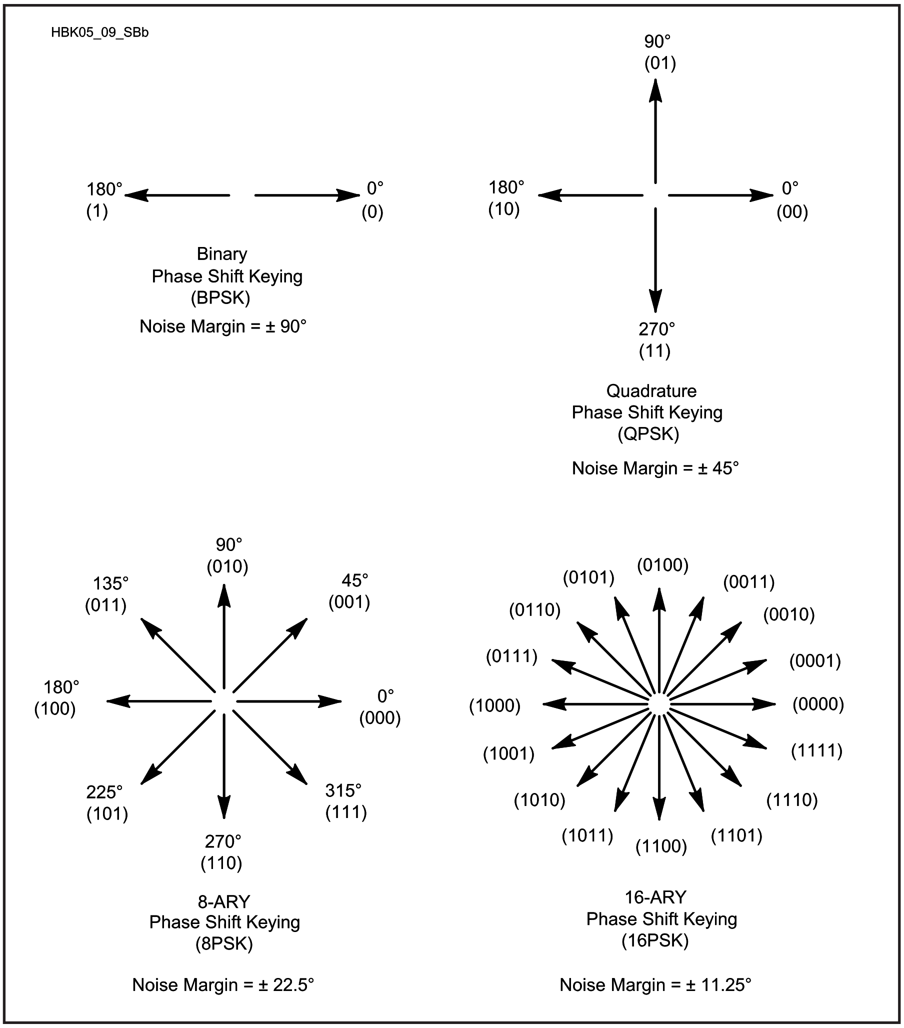

Fig 9.2—An example of multilevel phase modulation.

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 9.2**. One type of a complex system would be QAM (quadrature amplitude modulation), which requires special equipment to observe.

With complex modulation forms, noise-induced changes in both amplitude and phase will cause some pulses to fall into oblivion, where you simply cannot say what was sent. Thus, a higher signal to noise ratio is needed for complex types of modulation such as 64QAM, whereas binary modes may work closer to the noise.

For high bit rates on HF, many carriers may be used, each one carrying multiple amplitude and phase levels. There are, however, regulatory and practical limits. Thus, while it is feasible to transmit digital speech in a 3-kHz bandwidth, full-motion, high-resolution full-motion TV signals would be very difficult to transmit on HF, even if an entire amateur band were used.

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.

All present and future digital modes must address these 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 work poorly 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 are often used. The design of digital communications systems will always be an exercise in trade-offs.

EFFECT OF THE HF PATH ON ANALOG SIGNALS

Analog signals undergo the same impairments as digital signals along an ionospheric transmission path. However, the signals are normally decoded by the combination of the human ear and brain, which overcome problems, often without much notice. Frequency-selective fading of double-sideband AM signals manifests itself as distortion or mushiness at the receiver audio that may be ignored by the human operator. Even if the signal becomes temporarily unreadable, often the operator can fill in the blanks because the information is familiar or expected. Single-sideband AM (SSB) usually suffers less mushiness in the receiver due to frequency selective fading because the signal occupies a narrower bandwidth, in which low and high-frequency audio tend to fade together. The human operator can usually cope with such temporary fadeouts or can request a retransmission for any part missed.

MULTIPLE ACCESS AND MULTIPLEXING TECHNOLOGIES

To appreciate some of the more complex communications systems, you need an understanding of the different methods of sharing a carrier or accessing the frequency spectrum. *Multiple access* refers to more than one originating source having use of the media. *Multiplexing* means combining of two or more information streams into one carrier or transmission path.

Frequency Division Multiple Access (FDMA)

FDMA is probably the oldest and most familiar method of accessing the frequency spectrum, since individual signals are on different frequency channels. It is also the least efficient, since each frequency occupies a slot that is reserved for one user at a time.

Frequency Division Multiplexing (FDM)

FDM uses more than one subcarrier, imposed on a carrier, to convey different information. It traditionally was used for multiplexed telephone systems but is rarely used in the Amateur Radio Service.

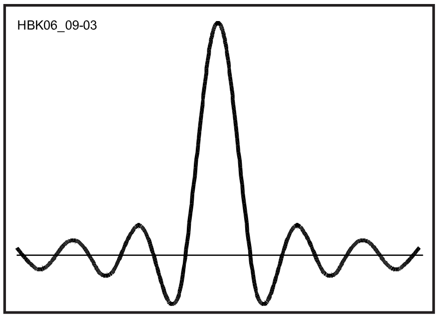

Fig 9.3—Spectrum of an individual OFDM carrier.

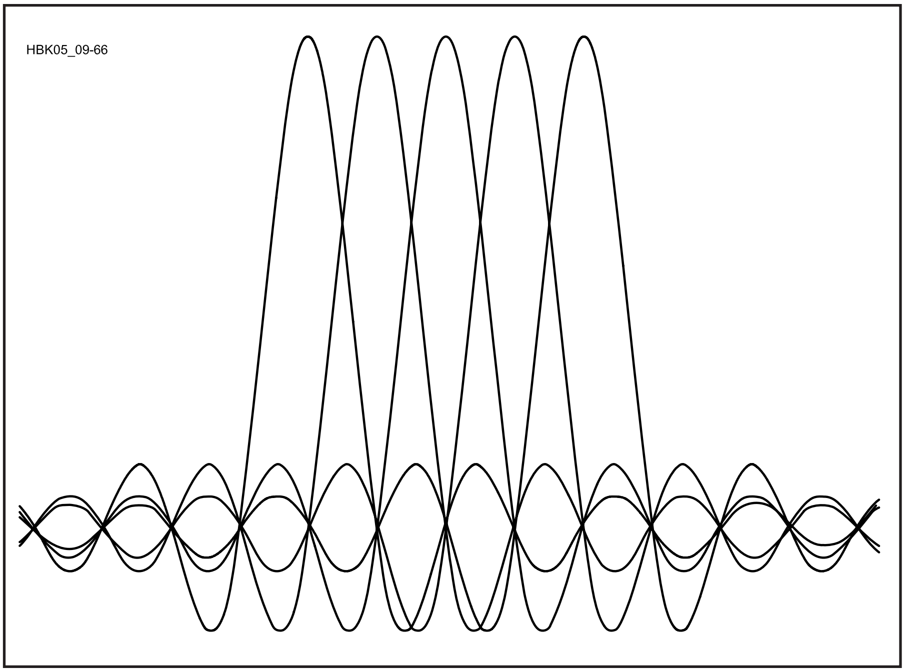

Fig 9.4—Overall OFDM spectrum.

Time Division Multiple Access (TDMA)

TDMA is simply time-sharing a frequency. In a general sense, this occurs naturally as stations in a QSO take turns transmitting. TDMA is also used in digital systems that reverse the direction of a circuit automatically to send information and acknowledgements.

Time Division Multiplex (TDM)

TDM is transmission of two or more signals over a common channel by interleaving so that the signals occur in different time slots. Some cellular telephone systems, such as Global System for Mobile Communications (GSM) use TDM. In the Amateur Radio Service it is used mostly for telemetry, such as from amateur satellites and remote repeaters.

Orthogonal Frequency Division Multiplexing (OFDM)

The term *orthogonal* is derived from the fact that multiple carriers are closely spaced in frequency, but positioned such 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. **Figs 9.3** and **9.4,** illustrate how the carriers are interleaved to prevent intercarrier interference.

Because each carrier is modulated at a relatively low rate, OFDM links suffer less intersymbol interference (ISI) on HF ionospheric paths than single-carrier modulation at a higher rate. See Smith, Doug, KF6DX, "Distortion and Noise in OFDM Systems," *QEX* Mar/Apr 2005.

Code Division Multiple Access (CDMA)

CDMA is a form of *spread spectrum* and is generated by modulating a carrier with a spreading code sequence known to both the sender and receiver. Unlike FDMA and TDMA, there is no fixed limit on the number of users but the number is not infinite.

Major Modulation Systems

The broadest category of modulation is how the *main carrier* is modulated. The major types are amplitude modulation, angle modulation and pulse modulation.

AMPLITUDE MODULATION

Amplitude modulation (AM) covers a class of modulation systems in which the amplitude of the *main* carrier is the characteristic that is varied. AM is sometimes simplistically described as varying the amplitude of the carrier from zero power to a peak power level. In fact, the carrier itself stays at the same amplitude when modulated by an analog (such as voice) baseband signal. The modulation itself produces sidebands, which are bands of frequencies on both sides of the carrier frequency. AM is basically a process of heterodyning or nonlinear mixing. As in any mixer, when a carrier and baseband modulation are combined, there are three products in the frequency range of interest: (1) the carrier, (2) the lower sideband (LSB), and (3) the upper sideband (USB). Thus, if a carrier of 10 MHz were modulated by a 1-kHz sine wave, the outputs would be as shown in **Fig 9.5**.

The bandwidth of the modulated signal in this example would be 2 kHz, the difference between the lowest and highest frequencies. In AM, the difference between the carrier and farthest-away component of the sideband is determined by the highest frequency component contained in the baseband-modulating signal.

ANGLE MODULATION

Two particular forms of angle modula-

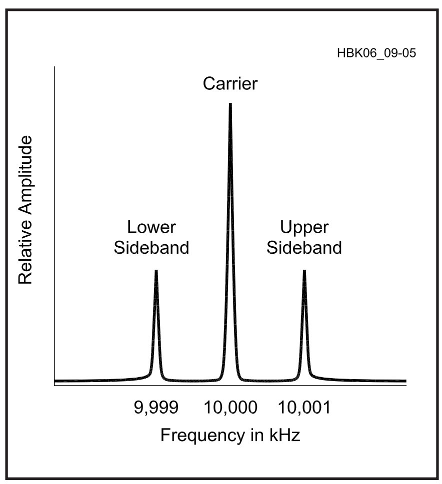

Fig. 9.5—A 10-MHz carrier AM-modulated by a 1-kHz sine wave.

tion are *frequency modulation* (FM) and *phase modulation* (PM). Frequency and phase modulation are not independent, since the frequency cannot be varied without also varying the phase, and vice versa.

The communications effectiveness of FM and PM depends almost entirely on the receiving methods. If the receiver can respond to frequency and phase changes but is insensitive to amplitude changes, it will discriminate against most forms of noise, particularly impulse noise, such as that from ignition systems.

Frequency Modulation

Fig 9.6 is a representation of frequency modulation. When a modulating signal is applied, the carrier frequency is increased during one half cycle of the modulating signal and decreased during the half cycle of the opposite polarity. This is indicated in the drawing by the fact that the RF cycles occupy less time (higher frequency) when the modulating signal is negative.

The change in the carrier frequency (*frequency deviation*) is proportional to the instantaneous amplitude of the modulating signal. Thus, the deviation is small when the instantaneous amplitude of the modulating signal is small and is greatest when the modulating signal reaches its peak, either positive or negative. The drawing shows that the amplitude of the RF signal does not change during modulation. This is an oversimplification and is true only in the overall sense, as the amplitude of both the carrier and sidebands do vary with frequency modulation. FM is capable of conveying dc levels, as it can maintain a specific frequency.

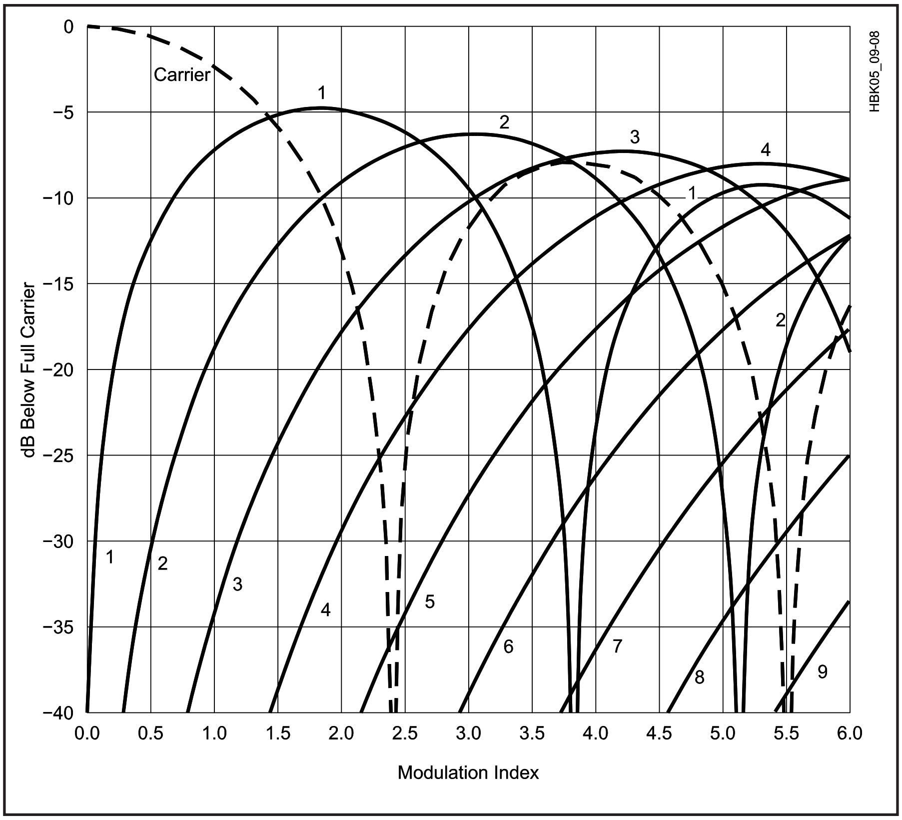

Fig 9.7—Amplitude of the FM carrier and sidebands with modulation index. This is a graphical representation of mathematical functions developed by F. W. Bessel. Note that the carrier completely disappears at modulation indexes of 2.405 and 5.52.

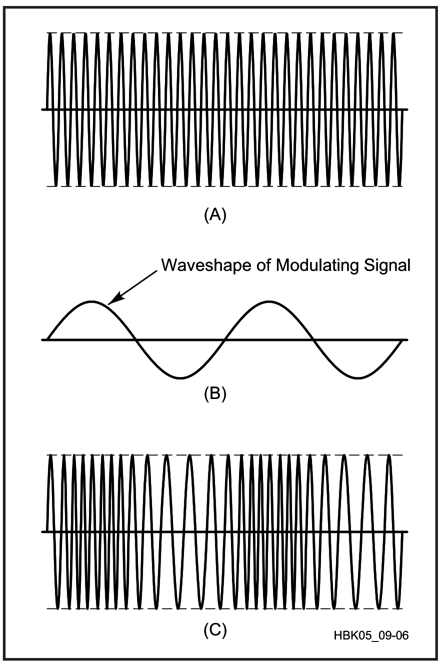

Fig 9.6—Graphical representation of frequency modulation. In the unmodulated carrier (A) each RF cycle occupies the same amount of time. When the modulating signal (B) is applied, the radio frequency is increased and decreased according to the amplitude and polarity of the modulating signal (C).

Phase Modulation

In phase modulation, the characteristic varied is the carrier phase from a reference value. In PM systems, the demodulator responds only to instantaneous changes in frequency. PM cannot convey dc levels unless special phase-reference techniques are used. The amount of frequency change, or deviation, is directly proportion to how rapidly the phase is changing and the total amount of the phase change.

Bessel Functions

Bessel functions are employed—using the carrier null method—to set deviation. Some version of the chart shown in **Fig 9.7** has appeared in the *ARRL Handbook* for 50 years. This chart is unlike previous ones in that the values are plotted here 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 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 accurately, 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-mode receiver using a narrow CW filter could be used to detect the carrier null, using the S meter and carefully tuning to 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 the deviation with narrow-band 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 even be used to calibrate a deviation meter.

You can also use the plot in Fig 9.7 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.

Operating Modes

This chapter examines various *operating modes* used in the Amateur Radio Service, including text modes, data, telemetry and telecommand, voice, image, spread spectrum and multimedia. While modes once fit into neat categories, there is now a blurring of the definitions. For example, data transmissions could include images.

TELEGRAPHY MODES

These are basically text modes; that is, transmission of letters, figures and punctuation, in a format suitable for printing at the receiving station. Morse telegraphy and radioteletype (Baudot and ASCII) are described, but you should be aware that the term "telegraphy" includes facsimile transmission as well.

Morse Telegraphy (CW)

Text messages sent by on-off keying (OOK, also known as amplitude-shift keying, or ASK) 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 they use and many take great and justifiable pride in their proficiency with it. The complete international Morse (including the new @ character ·−−·−·) code itself is defined in ITU-R Recommendation M.1677, *International Morse code*.

CW continues in use, however, not just for reasons of nostalgia. When used by an experienced operator, it can rival most any mode for "getting the message 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. Depending on ability of the operator, direct human copy works well between about 5 and 60 WPM, but 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 below the noise, while very fast speeds are useful for meteor scatter communication where bandwidth is large, but the reflection path lasts only 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 of the keying pulse. These 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 a sine wave. See **Figs 9.8** and **9.9**. 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 nar-

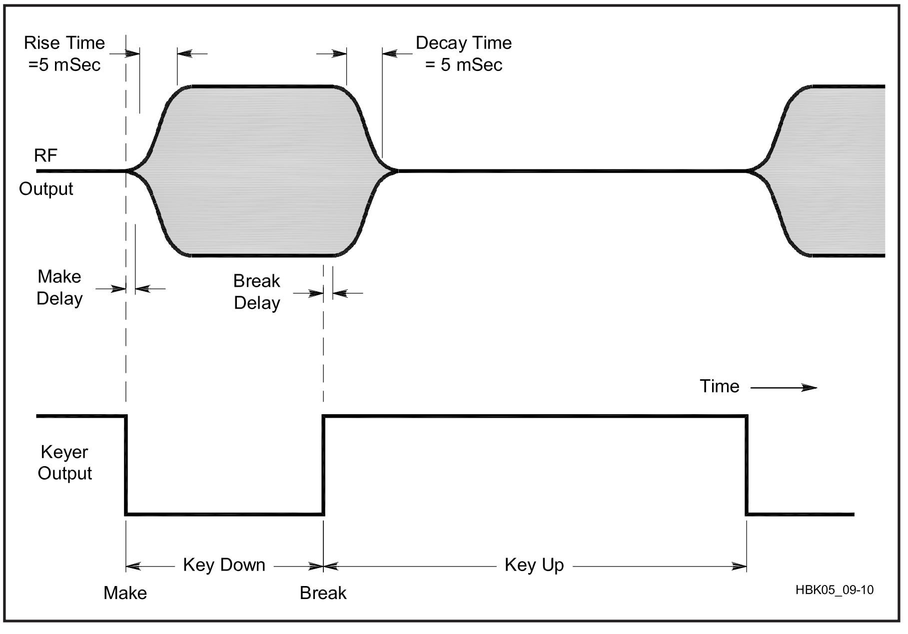

Fig 9.8—Optimum CW keying waveforms. The on-off transitions of the RF envelope should be smooth, ideally following a sine-wave curve. See text.

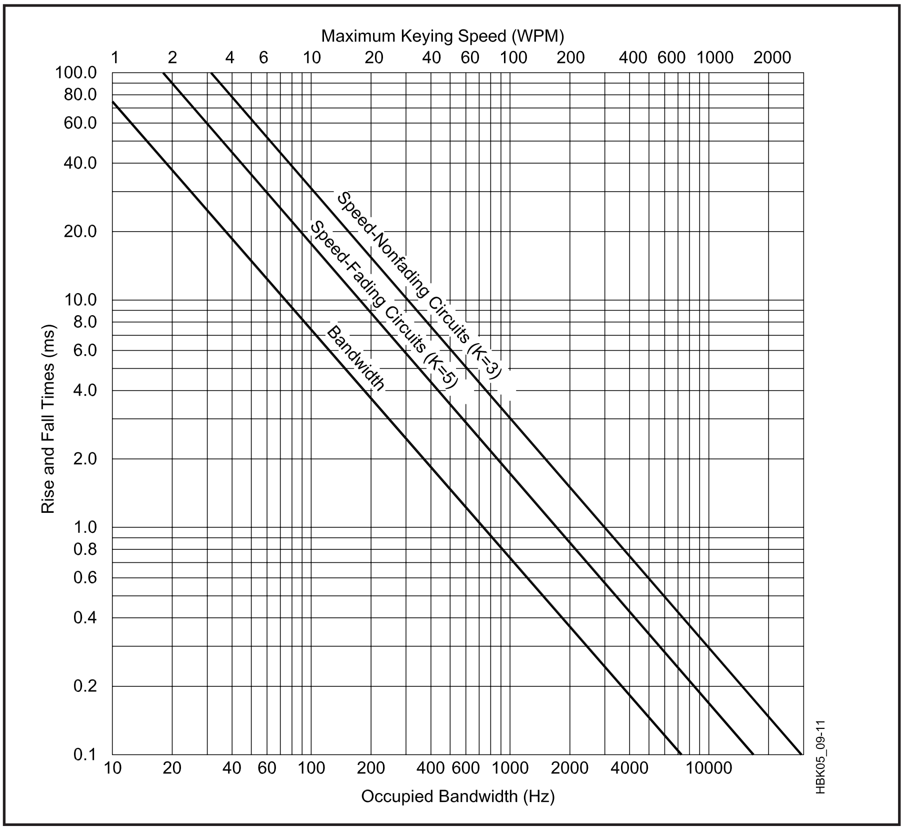

Fig 9.9—Keying speed vs rise and fall times vs bandwidth for fading and non-fading communications circuits. For example, to optimize transmitter timing for 25 WPM on a non-fading circuit, draw a vertical line from the WPM axis to the K = 3 line. From there draw a horizontal line to the rise/fall time 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.

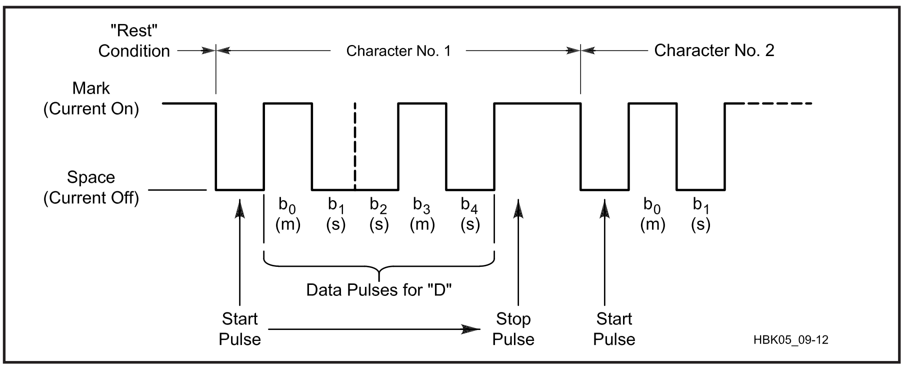

Fig 9.10—A typical Baudot timing sequence for the letter "D."

rower filter than this on 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.

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 a fraction of the bandwidth.

Baudot Radioteletype (RTTY)

One of the first data communications codes to receive widespread use had five bits (also 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 included with this book. In the United Kingdom, the almost-identical code is called *Murray* code. There are many variations in five-bit coded character sets, principally to accommodate foreign-language alphabets.

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

FCC rules provide that ITA2 transmissions must be sent using start-stop pulses, as illustrated in **Fig 9.10**. The bits in the figure are arranged as they would appear on an oscilloscope.

Speeds and Signaling Rates

The signaling speeds for RTTY are those used by the old TTYs, primarily 60 WPM or 45.45 bauds. The *baud* (Bd) 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.

Transmitter Keying

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

AFSK

Multimode communications processors (MCPs), however, generally connect to the radio AF input and output, often through the speaker and microphone connectors, and 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 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. Also remember that equipment is operating at 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 the level that is safe for CW operation.

What are High and 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

TUs (Terminal Units) have been replaced by multi-mode communications processors (MCPs), which accept AF signals from a radio and translate them into common ASCII text or graphics file formats (see **Fig 9.11**). Because the basic interface is via ASCII, MCPs are compatible with virtually any PC running a simple terminal program. Many MCPs handle CW, RTTY, ASCII, packet, fax, SSTV and new digital modes as they come into amateur use. To an increasing extent, personal computer sound cards with appropriate software are a viable and low-cost alternative to MCPs. However, sound cards have their limitations and dedicated hardware can more efficiently perform some operations.

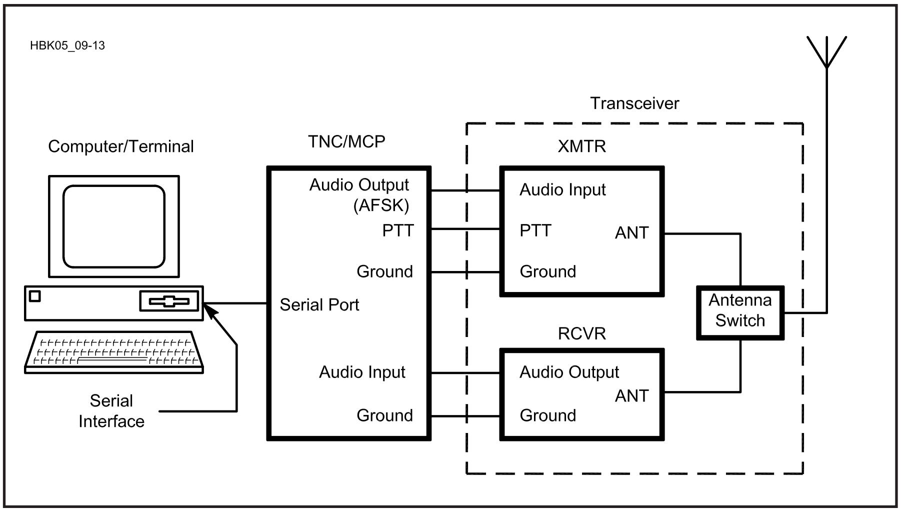

Fig 9.11—A typical multimode communications processor (MCP) station. MCPs can do numerous data modes as well as SSTV and fax.

AFSK Demodulators

An AFSK demodulator takes the shifting tones from the audio output of a receiver and produces TTY keying pulses. FM is a common AFSK demodulation method. The signal is first bandpass 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 is 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.

Diversity Reception

Although not restricted to RTTY, *diversity reception* can be achieved by using two antennas, two receivers and a dual demodulator. Some amateurs are using it with good 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 near the first antenna. A problem is to get both receivers on the same frequency without carefully tuning each one manually. Two demodulators are needed for this type of diversity. Also, some type of diversity combiner, selector or processor is needed. Many commercial or military RTTY demodulators are equipped for diversity reception.

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

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ASCII (IA5) RADIOTELETYPE

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 ITU-T Recommendation 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 that accompanies this book.

Parity

While not strictly a part of the ASCII standard, an eighth bit may be added for parity (P) checking. FCC rules permit optional use of the parity bit. The applicable US and in-

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.
ALE—Automatic link establishment.
AMRAD—Amateur Radio Research and Development Corporation, a nonprofit organization dedicated to experimentation.
AMSAT—Radio Amateur Satellite Corporation.
AMTOR—Amateur teleprinting over radio, an amateur radioteletype transmission technique employing error correction as specified in several ITU-R Recommendations M.476-2 through M.476-4 and M.625.
ANSI—American National Standards Institute.
Answer—The station intended to receive a call. In modem usage, the called station or modem tones associated therewith.
APCO—Association of Public Safety Communications Officials.
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.
Bit/s—Bits per second.
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.
CDMA—Code division multiple access.
Chirp—Incidental frequency modulation of a carrier as a result of oscillator instability during keying.
CLOVER—Trade name of digital communications system developed by Hal Communications.
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.
DBPSK—Differential binary phase-shift keying.
DQPSK—Differential quadrature phase-shift keying.
DCE—Data circuit-terminating equipment, the equipment (for example, a modem) that provides communication between the DTE and the line radio equipment.
Domino—A conversational HF digital mode similar in some respects to MFSK16.
DRM—Digital Radio Mondiale. A consortium of broadcasters, manufacturers, research and governmental organizations which developed a system for digital sound broadcasting in bands between 100 kHz and 30 MHz.
EIA—Electronic Industries Alliance.
EIA-232—An EIA standard physical-level interface between DTE (terminal) and DCE (modem), using 25-pin connectors. Formerly RS-232, a popular serial line standard, equivalent of ITU-T V.24 and V.28.
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.
Facsimile (fax)—A form of *telegraphy* for the transmission of fixed images, with or without half-tones, with a view to their reproduction in a permanent form.
FCS—Frame check sequence. (See CRC.)
FDM—Frequency division multiplexing
FDMA—Frequency division multiple access
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.
GNU—A project to develop a free UNIX style operating system.
G-TOR—A digital communications system developed by Kantronics.
HDLC—High-level data link control procedures as specified in ISO 3309.
Hellschreiber—A facsimile system for transmitting text.
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, ITU-T 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 ITU-T 5-bit coded character set commonly called the Baudot or Murray code.

ITU—International Telecommunication Union, a specialized agency of the United Nations. (See **www.itu.int**.)
ITU-R—Radiocommunication Sector of the ITU, formerly CCIR.
ITU-T—Telecommunication Standardization Sector of the ITU, formerly CCITT.
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, ITU-T Recommendation X.25 unbalanced-mode communications.
LAPB—Link access procedure, balanced, ITU-T Recommendation 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.
Linux—A free Unix-type operating system originated by Linus Torvalds, et al. Developed under the GNU General Public License.
Loopback—A test performed by connecting the output of a modulator to the input of a demodulator.
LSB—Least-significant bit.
MFSK16—A multi-frequency shift communications system
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.
MT-63—A keyboard-to-keyboard mode similar to PSK31 and RTTY.
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-232 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 that 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 ITU-T Recommendation 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.
PACTOR®—Trade name of digital communications protocols offered by Special Communications Systems GmbH & Co KG (SCS).
Parity check—Addition of non-information 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.)
Project 25—Digital voice system developed for APCO, also known as P25.
Protocol—A formal set of rules and procedures for the exchange of information within a network.
PSK—Phase-shift keying.
PSK31—A narrow-band digital communications system developed by Peter Martinez, G3PLX.
Q15X25—A DSP-intensive mode intended as an error-free mode more reliable on HF than packet.
QAM—Quadrature Amplitude Modulation. A method of simultaneous phase and amplitude modulation. The number that precedes it, eg, 64QAM, indicates the number of discrete stages in each pulse.
QPSK—Quadrature phase-shift keying.
RAM—Random access memory.
Router—A network packet switch. In packet radio, a network-level relay station capable of routing packets.
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.
SCAMP—Sound Card Automated Message Protocol, an inexpensive alternative to hardware for passing e-mail traffic on narrow-bandwidth channels.
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.
TDM—Time division multiplexing
TDMA—Time division multiple access
Telecommand—The use of telecommunication for the transmission of signals to initiate, modify or terminate functions of equipment at a distance.
Telemetry—The use of telecommunication for automatically indicating or recording measurements at a distance from the measuring instrument.
Telephony—A form of telecommunication primarily intended for the exchange of information in the form of speech.
Telegraphy—A form of telecommunication in which the transmitted information is intended to be recorded on arrival as a graphic document; the transmitted information may sometimes be presented in an alternative form or may be stored for subsequent use.
Teleport—A radio station that acts as a relay between terrestrial radio stations and a communications satellite.
Television—A form of telecommunication for the transmission of transient images of fixed or moving objects.
Throb—A multi-frequency shift mode like MFSK16.
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—An ITU-T Recommendation defining physical-level interface circuits between a DTE (terminal) and DCE (modem), equivalent to EIA-232.
V.28—An ITU-T Recommendation 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—An ITU-T packet-switching protocol Recommendation.

ternational standards (ANSI X3.16-1976; ITU-T Recommendation 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. By sacrificing parity, the eighth bit can be used to extend the ASCII 128-character code to 256 characters.

ASCII Serial Transmission

Serial transmission standards for ASCII (ANSI X3.15 and X3.16; ITU-T Recommendation 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 9.12A**.

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.

ASCII Data Rates

Data-communication signaling rates depend largely on the medium and the state of the art when the equipment was selected. 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.

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

Amateur ASCII RTTY Operations

On April 17, 1980, the FCC first permitted ASCII in the Amateur Radio Service. Amateurs have been slow to abandon Baudot in favor of asynchronous serial ASCII. Rather than transmitting start-stop ASCII, this code has become embedded in more sophisticated data transmission modes, which are described later in this chapter.

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AMTOR

AMTOR is derived from ITU-R Recommendation M.476, and is known "narrowband direct printing" (NBDP) and commercially as "SITOR." It has been largely overtaken by newer protocols.

AMTOR 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 Mode A or Mode B, the second type of time diversity is supplied by the redundancy of the code itself.

Mode B (FEC)

When transmitting to no particular station (for example calling CQ, in a net operation or during bulletin transmissions) there is no (one) receiving station to request repeats. 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 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. 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.

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.

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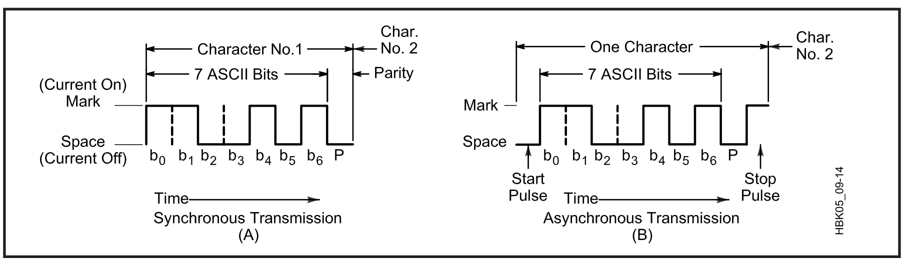

Fig 9.12—Typical serial synchronous and asynchronous timing for the ASCII character "S."

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PSK31

Peter Martinez, G3PLX, who was instrumental in bringing us AMTOR, also 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 9.13**). Much like Morse code, the more commonly used letters in PSK31 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 systems.

The shifting carrier phase generates sidebands 31.25 Hz from the carrier. These are used to synchronize the receiver with the transmitter. The bandwidth of a PSK31 signal is shown in **Fig 9.14**.

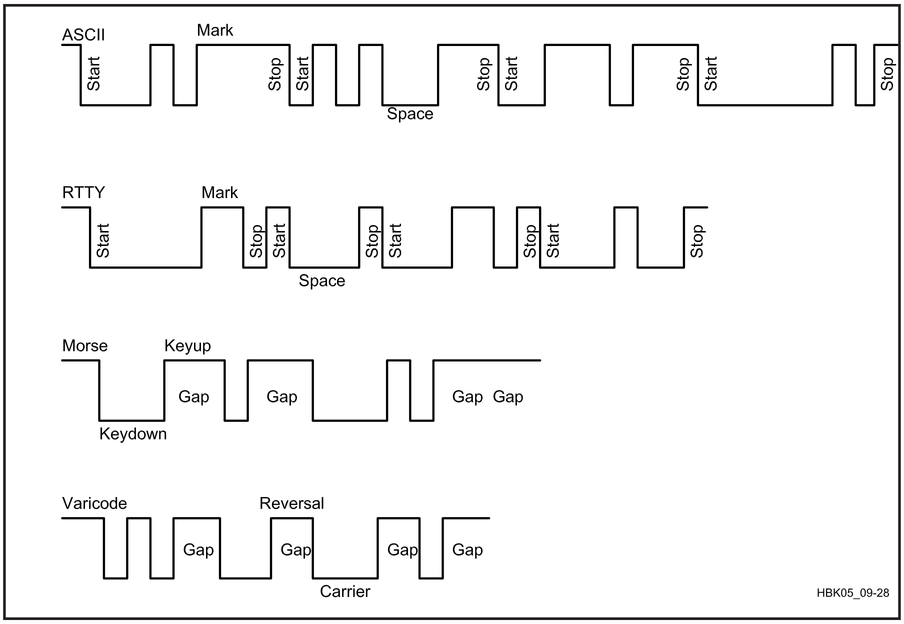

Fig 9.13—Codes for the word "ten" in ASCII, Baudot, Morse and Varicode.

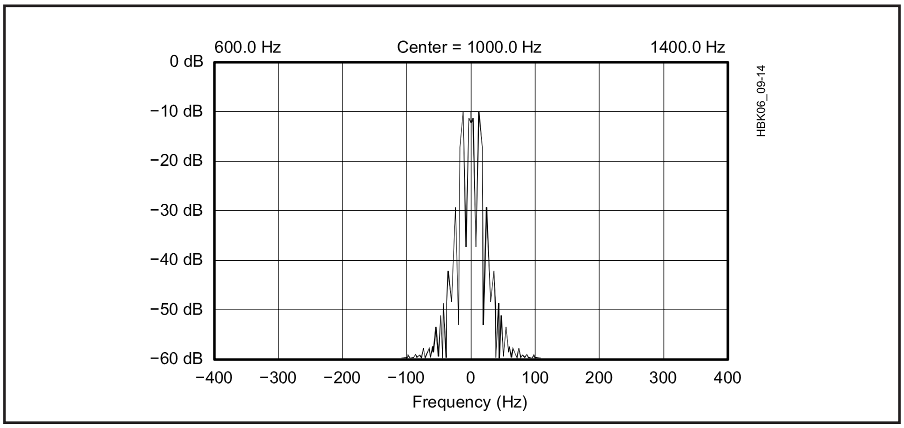

Fig 9.14—The spectrum of a PSK31 signal.

PSK31 Error Correction

Martinez added error correction to PSK31 by using QPSK (quaternary 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 Windows operating system and a 16-bit sound card to receive and transmit PSK31. Additional information and software is available for free download over the Web. Use a search engine to find PSK31 information and links to downloads.

An interesting wrinkle is to generate text, transmit it via PSK31 or some other RTTY or data mode, receive it and use a speech synthesizer to read the message. An example of this technique was described by W3NRG in the October 2004 issue of *CQ Magazine* (p 48). Synthesized speech takes some getting used to, as everybody sounds pretty much alike, and the personality of the speaker does not come through.

PSK31 BIBLIOGRAPHY

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DATA MODES

The difference between text and data modes is not abrupt but a blur. *Data* could be used to mean text, numbers, telecommand, telemetry and in some cases images. The third letter of the emission symbol "D" is used in common for data, telecommand and telemetry.

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 a small group of experimenters 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.

Packet radio 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.

Packet uses time efficiently, since packet bulletin-board systems (PBBSs) permit packet operators to store information for later retrieval by other amateurs. And 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 9.15**).

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 TNC accepts data from a computer or data terminal and assembles it into packets (see **Fig 9.16**). 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 com-

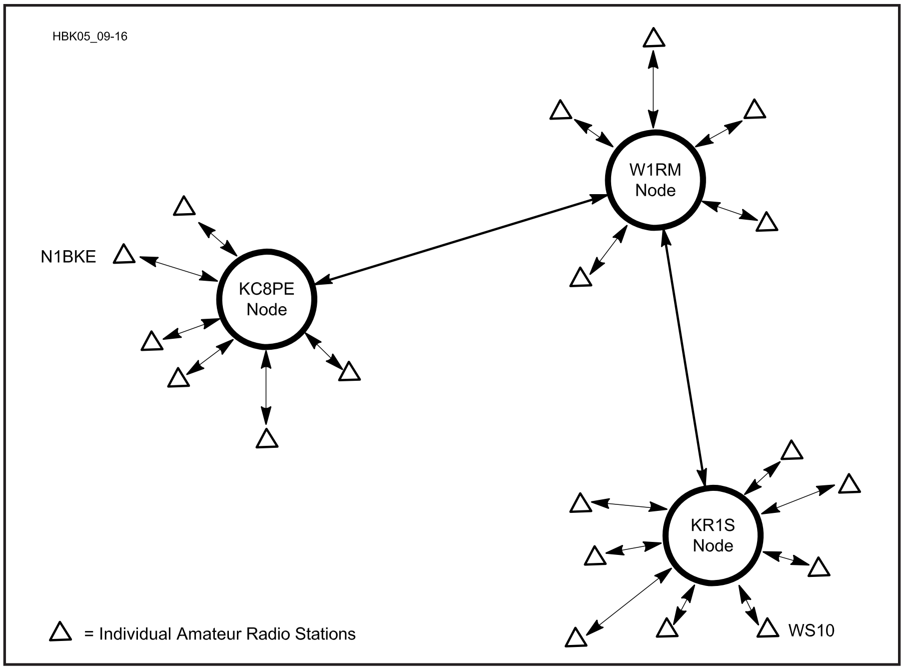

Fig 9.15—*DX PacketClusters* are networks comprised of individual nodes and stations with an interest in DXing and contesting. In this example, N1BKE is connected to the KC8PE node. If he finds a DX station on the air, he'll post a notice—otherwise known as a *spot*—which the KC8PE node distributes to 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.

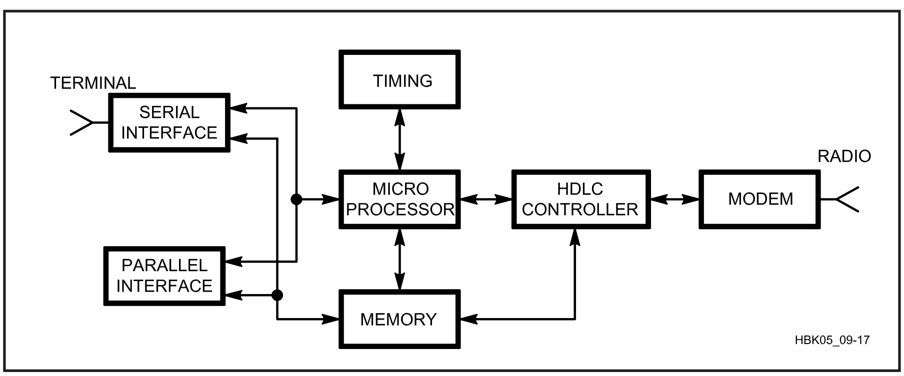

Fig 9.16—The functional block diagram of a typical TNC.

puter or terminal can understand. The part of the TNC that performs this tone-translating function is known as a *modem* (see **Fig 9.17**).

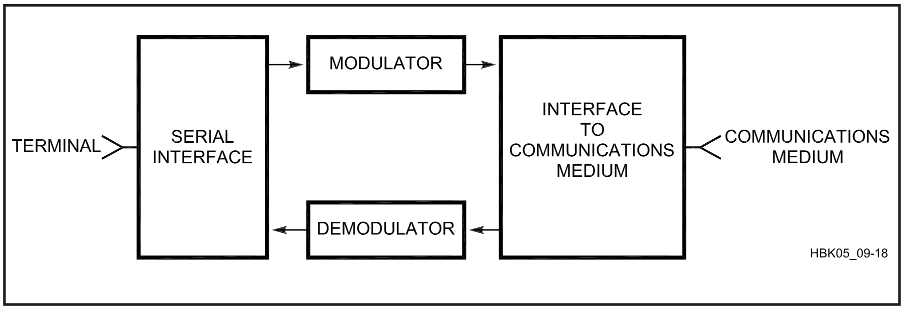

Fig 9.17—A block diagram of a typical modem.

Fig 9.18—Four popular 1200 bit/s packet TNCs: (clockwise, from bottom left) the MFJ-1270C, AEA PK-88, Kantronics KPC-3 and the DRSI DPK-2.

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

You have plenty of TNCs to choose from. The amount of money you'll spend depends directly on what you want to accomplish. Most TNCs are designed to operate at 300 and 1200 bit/s, or 1200 bit/s exclusively (see **Fig 9.18**). 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.

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

TNC Emulation and Internal TNCs

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

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

Transceiver Requirements

Packet activity on the HF bands typically takes place at 300 bit/s using common SSB transceivers. The transmit audio is fed from the TNC to the microphone jack or auxiliary audio input. Receive audio is obtained from the radio's 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.

In the mid 1990s 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.

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.

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. 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. 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.

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.

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* for short.

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

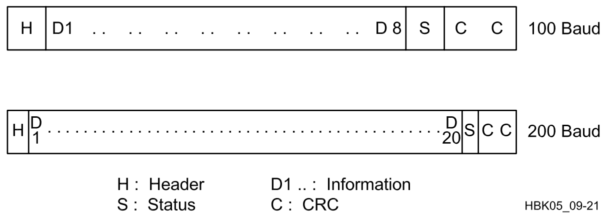

Fig 9.19—PACTOR data packet format.

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PACTOR

PACTOR (PT), now often referred to as PACTOR-I, is an HF radio transmission system developed by German amateurs Hans-Peter Helfert, DL6MAA, and Ulrich Strate, DF4KV. It was designed to overcome the shortcomings of AMTOR and packet radio. It performs well under both weak-signal and high-noise conditions. PACTOR-I has been overtaken by PACTOR-II and PACTOR-III but remains in use.

TRANSMISSION FORMATS

Information Blocks

All packets have the basic structure shown in **Fig 9.19**, and their timing is as shown in **Table 9.1**:

- *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 9.2**
- *CRC*: The CRC is calculated according to the CCITT standard, for the data, status and CRC.

Table 9.1

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 9.2

PACTOR Status Word

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

Data Format Bits

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

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

Acknowledgment Signals

The PACTOR acknowledgment signals are shown in **Table 9.3**. 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. 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 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 9.4**

Speed Changes

With good conditions, PACTOR's normal signaling rate is 200 bauds, but the system automatically changes from 200 to 100 bauds 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 immediately assembles the previous packet in 100-baud format and sends it. Thus, one packet is repeated in a change from 200 to 100 bauds.

The RX can acknowledge a good 100-baud packet with CS4. TX immediately switches to 200 bauds 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 bauds. The RX station responds with a final CS.

PACTOR-II

This is a significant improvement over PACTOR-I, yet it is fully compatible with the older mode. Also invented in Germany, PACTOR uses 16PSK to transfer up to 800 bit/s at a 100-baud rate. This keeps the bandwidth less than 500 Hz.

PACTOR-II uses digital signal processing (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 bit/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.

Table 9.3
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.

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. PACTOR-II stations acknowledge each received transmission block. 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.

PACTOR-III

PACTOR-III is a software upgrade for existing PACTOR-II modems that provides a data transmission mode for improved speed and robustness. PACTOR-III is not a new modem or hardware device. Most current PACTOR-II modems are upgradeable to use PACTOR-III via a software update, since PACTOR-II firmware accommodates the new PACTOR-III software. Both the transmitting and receiving stations must support PACTOR-III for end-to-end communications using this mode.

PACTOR-III's maximum uncompressed speed is 2722 bit/s. Using online compression, up to 5.2 kbit/s is achievable. This requires an audio passband from 400 Hz to 2600 Hz (for PACTOR-III speed level 6).

On an average channel, PACTOR-III is more than three times faster than PACTOR-II. On good channels, the effective throughput ratio between PACTOR-III and PACTOR-II can exceed five. PACTOR-III is also slightly more robust than PACTOR-II at their lower SNR edges.

The ITU emission designator for PACTOR-III is 2K20J2D. Because PACTOR-III builds on PACTOR-II, most specifications like frame length and frame structure are adopted from PACTOR-II. The only significant difference is PACTOR III's multi-tone waveform that uses up to 18 carriers while PACTOR-II uses only two carriers. PACTOR-III's carriers are located in a 120-Hz grid and modulated with 100 symbols per second DBPSK or DQPSK. Channel coding is also adopted from PACTOR-II's Punctured Convolutional Coding.

PACTOR-III Link Establishment

The calling modem uses the PACTOR-I FSK connect frame for compatibility. When the called modem answers, the modems negotiate to the highest level of which both modems are capable. If one modem is only capable of PACTOR-II, then the 500 Hz PACTOR-II mode is used for the session. With the *MYLevel* (MYL) command a user may limit a modem's highest mode. For example, a user may set MYL to "1" and

Table 9.4
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

Table 9.5
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

only a PACTOR-I connection will be made, set to "2" and PACTOR-I and II connections are available, set to "3" and PACTOR-I through III connections are enabled. The default MYL is set to "2" with the current firmware and with PACTOR-III firmware it will be set to "3". If a user is only allowed to occupy a 500 Hz channel, MYL can be set to "2" and the modem will stay in its PACTOR-II mode. The PACTOR-III Protocol Specification is available on the Web at **www.scs-ptc.com/pactor.html**.

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Ford, Steve, WB8IMY, ARRL's *HF Digital Handbook,* Third Ed., ARRL. 2004.

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*).

G-TOR is short for *Golay-TOR*, an innovation of Kantronics, Inc. It was inspired by HF Automatic Link Establishment (ALE) concepts and is structured to be compatible with ALE.

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 bauds
- 2.4-second 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 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 bauds, depending upon channel conditions. (G-TOR initiates contacts and sends ACKs only at 100 bauds.) 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 you 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.

Data Compression

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

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

Golay Coding

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

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

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tions/techchar/.

Ford, Steve, WB8IMY, ARRL's *HF Digital Handbook,* Third Ed., ARRL. 2004.

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 in a very narrow frequency spectrum. 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.

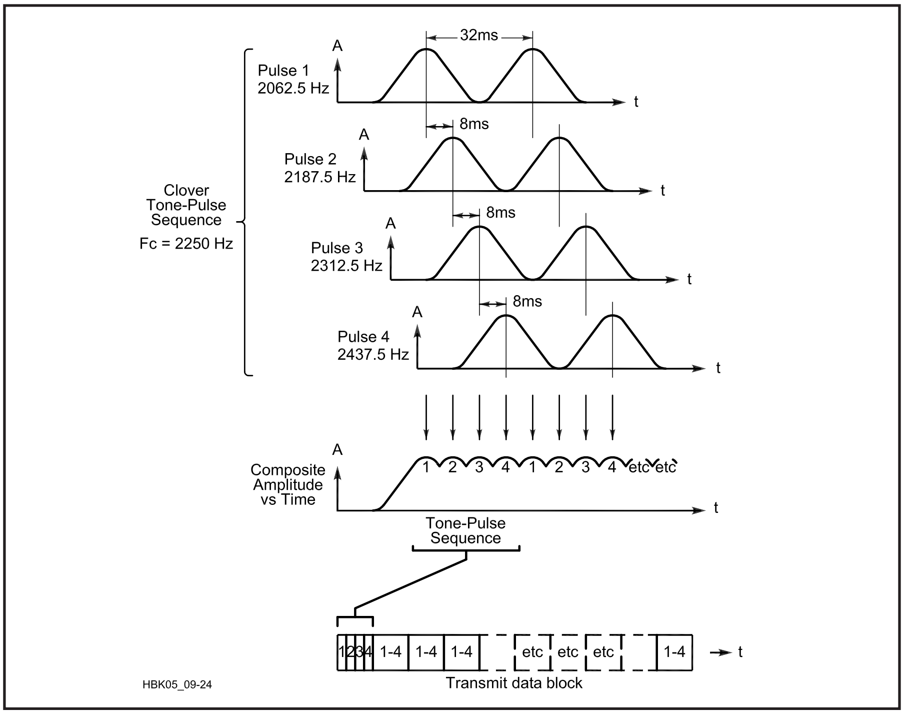

Fig 9.20—Amplitude vs time plots for CLOVER-II's four-tone waveform.

The CLOVER Waveform

Multilevel tone, phase and amplitude modulation give CLOVER a large selection of data modes that may be used (see **Table 9.5**). The adaptive ARQ mode of CLOVER senses current ionospheric 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 byte/s to 70 byte/s.

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 9.20**, **9.21** and **9.22**. The time-domain shape of each tone pulse is intentionally shaped to produce a very compact frequency spectrum. 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.

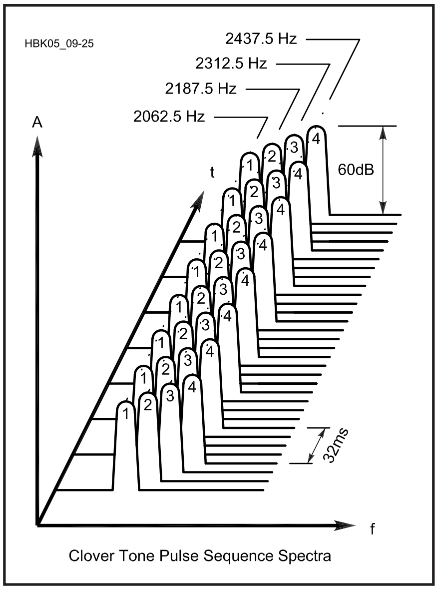

Fig 9.21—A frequency-domain plot of a CLOVER-II waveform.

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 versus the number of errors that can be corrected without resorting to retransmission of the entire data block.

Note that while the "In Block Data Rate" numbers listed in the table go as high as 750 bit/s, 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 9.6** and **Table 9.7** 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

Table 9.6
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 9.7
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

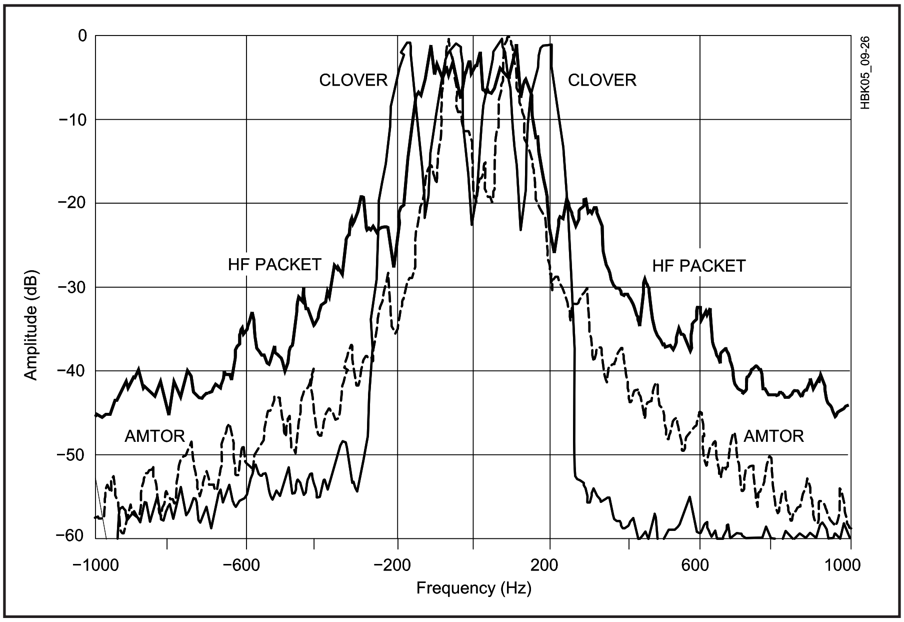

Fig 9.22—Spectra plots of AMTOR, HF packet-radio and CLOVER-II signals.

repeat transmission. This is a very powerful error-correction technique that is not available in some other common HF data modes.

CLOVER ARQ

Reed-Solomon data coding is the primary means by which errors are corrected in CLOVER "FEC" 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).

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 eight different waveform combinations that are actually used for FEC and/or ARQ modes.

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 bit/s.

Allowing for overhead, CLOVER-2000 can deliver error-corrected data over a standard HF SSB radio channel at up to 1994 bit/s, or 249 characters (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 (eg, 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 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 bit/s for BPSM to 1992 bit/s 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. 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 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 orderwire 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.

CLOVER Bibliography

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SCAMP AND RDFT

SCAMP (Sound Card Amateur Message Protocol) is intended as a low-cost alternative to commercial modems (TNCs). A paper describing SCAMP was presented by Rick Muething, KN6KB, at the 2004 ARRL/TAPR Digital Communications Conference. It is a new digital sound card protocol suitable for both HF and VHF for transmission of text messages with binary attachments. It is compatible with Winlink 2000 and is designed for manually initiated message forwarding. SCAMP is not a keyboard (chat) mode.

SCAMP incorporates the work by Barry Sanderson, KB9VAK, on Redundant Digital File Transfer (RDFT) and adds an ARQ wrapper around RDFT to ensure error-free transmission. There are four redundancy levels: 10%, 20%, 40% and 70%, the latter being the most robust and requires the most transmission time. Audio data is sent at a standard rate of 11.025 kHz, with 16-bit samples using a PC sound card. SCAMP occupies a bandwidth of about 2 kHz and a net throughput of 2 to 4 kbytes/minute, depending on conditions. It employs an automated "channel-busy" detector for reduction of QRM and to protect against QRM from "hidden transmitters." For more details, please see the SCAMP Bibliography, below.

On-the-air peer-to-peer testing began in November 2004 and the first transcontinental transmission was made in December 2004 between N6KZB in Temecula, CA, and W3QA in West Chester, PA. Beta testing began of SCAMP with WinLink 2000 in March 2005.

SCAMP and RDFT Bibliography

ARRL Letter, The Vol 23, No 48, "SCAMP On-Air Testing Commences," Dec 10, 2005.

Muething, "SCAMP (Sound Card Amateur Message Protocol)," *proc., 2004 ARRL/TAPR Digital Communications Conference.*

Muething, "SCAMP Protocol Specification," **winlink.org/Presentations/SCAMPspec.pdf**.

AUTOMATIC LINK ESTABLISHMENT

The US military services have found it difficult to maintain a sufficient number of qualified radio operators to operate MF/HF radios. So the Defense Department contracted with MITRE Corporation for the development of a method of operating MF/HF radios without skilled operators. MITRE studied what skilled operators do and developed Automatic Link Establishment (ALE) to operate radios and make contact with another station without human intervention and under computer control. ALE automatically finds the best frequency among a prearranged list using techniques such as selective calling, handshaking, link quality analysis, polling, sounding, etc.

ALE is used by the Military Affiliate Radio Service (MARS). It has also been adopted by some radio amateurs.

ALE Waveform

The ALE waveform is designed to be compatible with the audio passband of a standard SSB radio. It has a robust waveform for reliability during poor path conditions. It consists of 8-ary frequency-shift keying (FSK) modulation with eight orthogonal tones, a single tone for a symbol. These tones represent 3 bits of data, with least significant bit to the right, as follows:

Frequency	*Data*
750 Hz	000
1000 Hz	001
1250 Hz	011
1500 Hz	010
1750 Hz	110
2000 Hz	111
2250 Hz	101
2500 Hz	100

The tones are transmitted at a rate of 125 tones per second, 8 ms per tone. The resultant transmitted bit rate is 375 bit/s. The basic ALE word consists of 24 bits of information. Details can be found in Federal Standard 1045, Detailed Requirements, **www.its.bldrdoc.gov/fs-1054a/45-detr.htm**.

ALE Bibliography

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Brain, Charles, G4GUO, *PC-ALE Project*, **www.chbrain.dircon.co.uk/peale.html**.

Menold, Ronald, AD4TB, "ALE—The Coming of Automatic Link Establishment," *QST* February 1995, p. 68 (Technical Correspondence).

National Communications System, "Telecommunications: HF Radio Automatic Link Establishment," Federal Standard 1045A, October 1993.

Internetworking

Although it has been a goal of some radio amateurs to develop a digital communications network independent of the Internet, interconnection with the Internet provides a good bridge between isolated amateur radio nets. Several methods of transferring data, e-mail or linking repeaters have been developed.

WINLINK 2000

WinLink 2000 is a *Windows* application that permits messages to be transferred automatically between the Internet and remote amateur stations, which may be on recreational vehicles or at sea. The Internet is used as a backbone to allow WinLink mailbox operation (MBO) stations to share their databases. Its original author was Victor Poor, W5SSM. See: **winlink.org/**.

IRLP

Created by David Camerpon, WE7LTD, the Internet Radio Linking Project (IRLP) uses Voice over Internet Protocol (VoIP) to form a voice communications network of servers and nodes between amateur repeaters and/or simplex stations. See: **www.irlp.net/**.

EchoLink

EchoLink was developed by Jonathan Taylor, K1RFD, to link a personal computer to communicate by VoIP with several thousand repeaters having *EchoLink* capabilities. Or, it can be used to permit amateur stations within range of your station to connect with the Internet. See: **www.echolink.org/**.

eQSO

eQSO, created by Paul Davies, MØZPD, was designed to operate like a worldwide amateur radio net. See: **www.eqso.net/**.

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Linden, Louis, KI5TO, "Winlink 2000 in the Jungle," *QST*, November 2004.

TELEMETRY, TRACKING AND TELECOMMAND

According to FCC Part 97 rules, *telemetry* is a one-way transmission of measurements at a distance from the measuring instrument, whereas *telecommand* is a one-way transmission to initiate, modify or terminate functions of a device at a distance. Actually, the two go hand in hand, since it is important to have telemetry first, then modify the remote device, then look once again in the telemetry to see if the desired action took place.

Telemetry, tracking and telecommand (often seen as TT&C) are attracting increasing attention because Amateur Radio rules permit higher power transmitters than allowed under Part 15 of the FCC rules. TT&C is distinct from traditional forms of Amateur Radio (telegraphy, voice and image intended to be heard or seen by human operators), since it receives information from an object and commands the object to take an action. Although pulse modulation systems are common, TT&C also uses familiar communications modes, such as television (in this case used as a form of telemetry), packet radio (such as ASCII used for telemetry coding, commands or uploading programs).

This section provides only a sampling of telemetry, tracking and telecommand systems involving Amateur Radio. *APRS* (Automatic Position Reporting System) is a marriage of an application of the Global Positioning System and Amateur Radio to relay position and tracking information. Telemetry and telecommand are also used to manage remote terrestrial stations, as well as amateur satellites. Radio Control (R/C) of remote objects has long been a part of Amateur Radio because of the versatility offered by Part 97 rules to licensed operators. R/C is not limited to model cars, boats and airplanes but is vital for the growing field of robotics.

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 lifetime. 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 voice net. Stations that have information to contribute simply transmit it, and all stations monitor and collect all data on frequency. APRS also 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 Tracks, Maps and Mobiles* by Stan Horzepa, WA1LOU, published by ARRL.

Radio Control (R/C)

Amateur Radio gave birth to the radio control (R/C) hobby as we know it today. FCC §97.215 rules specifically permit "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. FCC §97.215 states:

"Telemetry transmitted by an amateur station on or within 50 km of the Earth's surface is not considered to be codes or ciphers intended to obscure the meaning of communications."

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 con-

Fig 9.23—Photo of three R/C model electric cars.

trol 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 9.23** 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 9.24** are the most popular. They can be unpowered (gliders) or powered by either electric or gas engines. 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 currently authorizes 91 R/C frequencies between 27 MHz and 76 MHz. Some frequencies are for all models, some are only for aircraft and others for surface (cars, boats) models only. Some frequencies are used primarily for toys and others for hobbyist models. Amateur Radio R/C

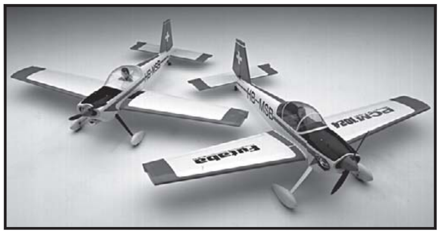

Fig 9.24—Photo of two R/C aircraft models.

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 Academy of Model Aeronautics (**www.modelaircraft.org**) *Membership Manual* provides a detailed list of all R/C frequencies in current use as well as other useful information. The *ARRL Repeater Directory* lists current Amateur Radio R/C frequencies.

Fig 9.25 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 50.8-51.0 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

(A)

(B)

Fig 9.25—A, photo of Futaba's Conquest R/C aircraft transmitter. B shows the matching airborne system.

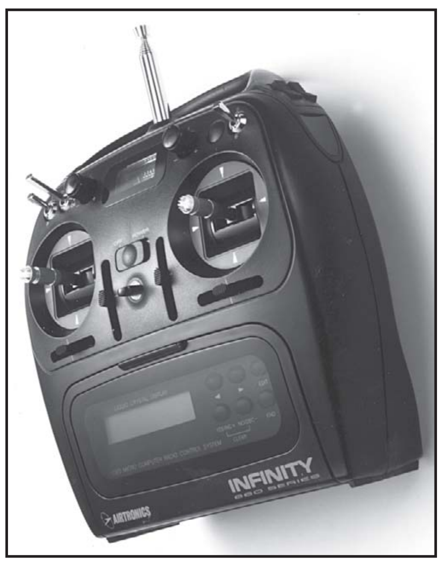

Fig 9.26—Photo of Airtronics Infinity 660 R/C aircraft transmitter.

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 9.26** 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.

Many R/C operators use the Amateur Radio channels to avoid crowding on the non-ham 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.

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.

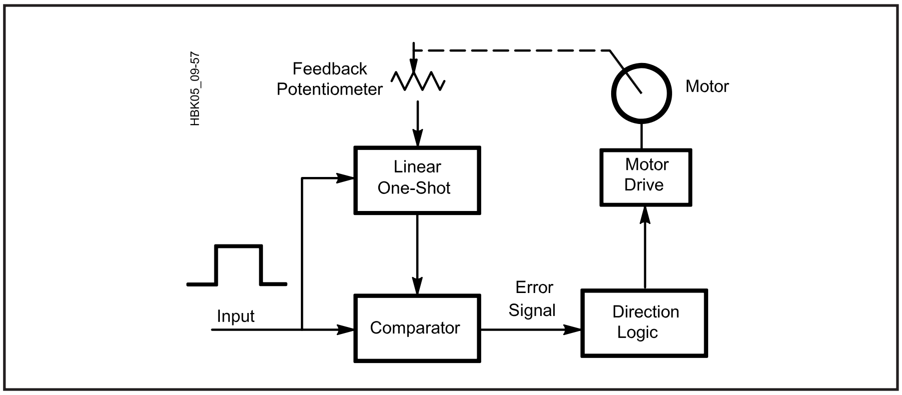

Fig 9.27—Diagram of a pulse-feedback servo.

Fig 9.27 is a block diagram of a pulse-feedback servo. The leading edge of the input pulse triggers a linear one-shot multivibrator. The width of the one-shot output pulse is compared to the input pulse. Any pulse width difference is an error signal that is amplified to drive the motor. The motor drives a feedback potentiometer that controls the one-shot timing. When this feedback loop reduces the error signal to a few microseconds, the drive motor stops. The servo position is a linear function of the input pulse width. The motor-drive electronics are usually timed for pulse repetition rates of 50 Hz or greater and a pulse width range of 1 to 2 ms. A significantly slower repetition rate reduces the servomechanism slew rate but not the position accuracy.

In addition to motor driven servos, the concept of pulse-width comparison can be used to operate solid-state or mechanical relay switches. The same 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 wired to operate a single control servomechanism. If such a pulse is used to modulate 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).

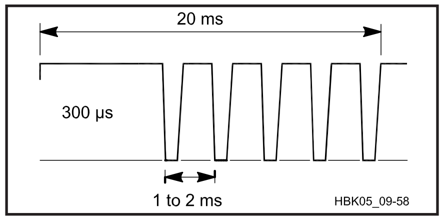

Fig 9.28—Diagram of a four-channel PPM RF envelope.

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 9.28** 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 9.29**. 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 carrier.

Received pulse decoding can also use digital logic semiconductors. **Fig 9.30** shows a simple four-control-channel decoder circuit using a 74C95 CMOS logic IC. The IC is a 4-bit shift register operated in the right-shift mode. Five data pulses spaced 1 to 2 ms apart, followed by a synchronization pause, contain the encoded pulse-width information in one frame. During the sync pause, the RC circuit discharges and sends a logic-one signal to the 74C95 serial input terminal. Subsequent negative going data pulses remove the logic-one signal from the serial input and sequentially clock the logic one through the four D-flip-flops. The output of each flip-flop is a positive going pulse, with a width corresponding to the time between the clocking pulses. The output of each flip-flop is a demultiplexed signal that is used to control the corresponding servo.

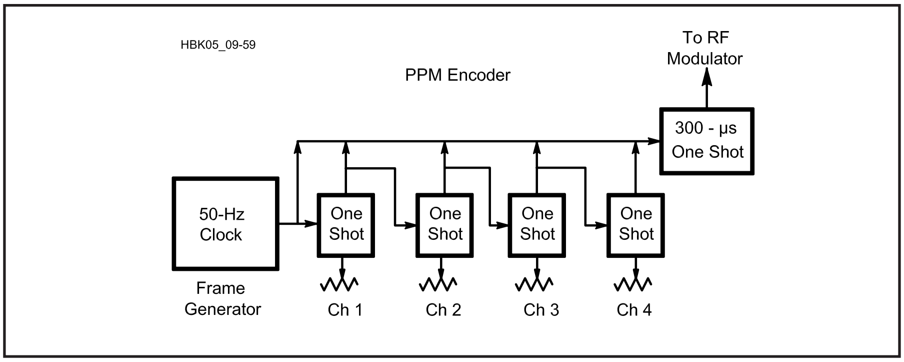

Fig 9.29—Diagram of a PPM encoder.

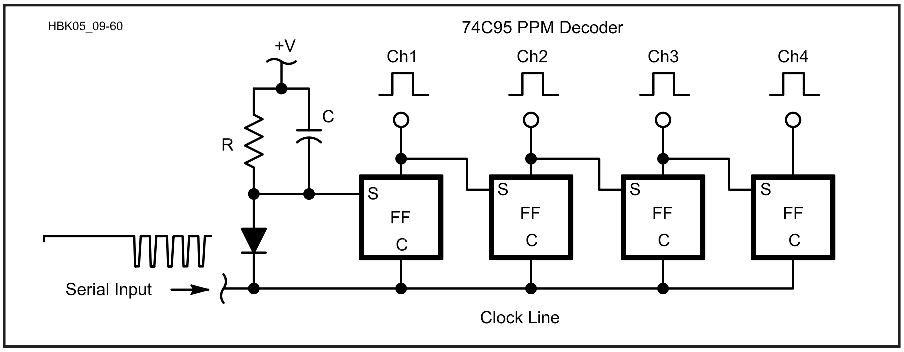

Fig 9.30—Diagram of a 74C95 PPM decoder.

Pulse Code Modulation

PCM uses true digital code to transfer R/C signals. The pulse width data of each control channel is converted to a binary word. The digital word information of each control channel is coded and multiplexed to permit transmission of multiple channels of control on a single RF carrier. On the receiving end, the process is reversed to yield the servo control signals.

There is no standard for how the digital word is coded for transmission. Therefore PCM R/C transmitters and receivers from different makers are not interchangeable. Some older PCM systems provide only 256 discrete positions for 90° of servo motion, thereby limiting servo resolution. Newer systems use more digital bits for each word and provide smooth servo motion with 512 and 1024 discrete positions. All PCM and PPM systems use the same servo input-signal and supply voltages. Therefore the servos of different manufacture are interchangeable once compatible wiring connectors have been installed.

AMATEUR SATELLITE TT&C

TT&C plays a vital part of the launching and management of amateur satellites. Satellites have onboard intelligence and are increasing able to make their own decisions but Article 25 of the international Radio Regulations requires the following:

> *"Administrations authorizing space stations in the amateur-satellite service shall ensure that sufficient earth command stations are established before launch to ensure that any harmful interference caused by emissions from a station in the amateur-satellite service can be terminated immediately."*

The U.S. implementation of the Radio Regulations in Part 15 of the FCC rules has these provisions:

> ***"§97.211 Space telecommand station.***
> *(a) Any amateur station designated by the licensee of a space station is eligible to transmit as a telecommand station for that space station, subject to the privileges of the class of operator license held by the control operator.*
> *(b) A telecommand station may transmit special codes intended to obscure the meaning of telecommand messages to the station in space operation.*
> .....
> *(d) A telecommand station may transmit one-way communications."*

Telemetry from amateur satellites, such as from "engineering beacons" is available to all amateurs. Computer programs are available from AMSAT for decoding the telemetry to monitor the health of the spacecraft and other measurements. However, telecommand of amateur satellites is closely held in order to maintain effective control.

Amateur Satellite TT&C References

AMSAT, **www.amsat.org**.

Davidoff, Martin, K2UBC, *The Radio Amateur's Satellite Handbook,* ARRL, Rev first ed, 2003.

Voice Modes

AMPLITUDE MODULATION (AM)

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 the radio frequency carrier.

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 situation was a mess, and QRM was king! By 1929 it was permissible to use AM voice in limited portions of our amateur spectrum and, on some bands, only the most qualified licensees had the privilege.

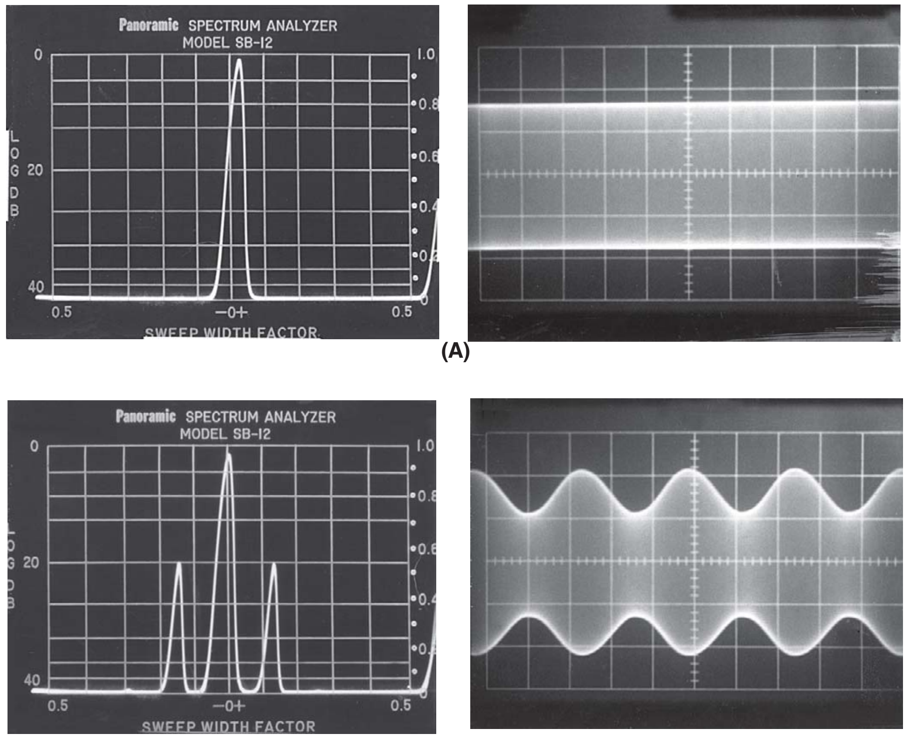

Fig 9.31—Electronic displays of AM signals in the frequency and time domains. A shows an unmodulated carrier or single-tone SSB signal. B shows a full-carrier AM signal modulated 20% with a sine wave.

Users of AM had to learn that an RF wave could have only a certain amount of audio imposed upon it before overmodulation occurred. Trying to go above 100% modulation produced severe distortion and splatter. 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. AMers 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 even 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 obsolescing 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 that 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** 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 9.31B**. 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 9.31B. The center peak, which is identical with the original unmodulated wave shown in Fig 9.31A, 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 that 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.

SINGLE SIDEBAND (SSB)

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 with 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,

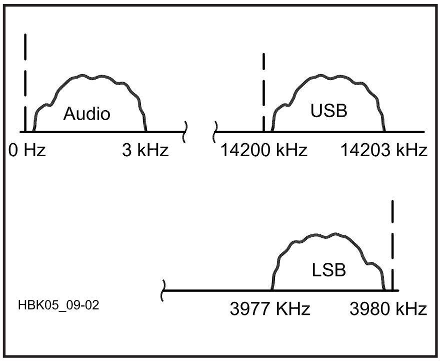

Fig 9.32—How an occupied radio frequency spectrum shifts with application of an audio (baseband) signal. The dotted line represents the RF carrier point, or in the case of the 3-kHz audio signal, the reference frequency, 0 kHz.

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 while 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 DSB AM. On HF, it is possible for the carrier to fade in an DSB AM signal, leaving less than is needed for envelope detection. Medium wave 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 9.32**. 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.

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*

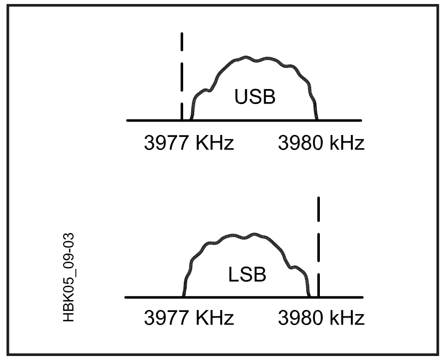

Fig 9.33—A method of changing sidebands with virtually no change in the frequency spectrum occupied. Note that the carrier point position *has* changed concurrent with the change from LSB to USB.

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 bandwidth on five center frequencies: 5332, 5348, 5368, 5373 and 5405 kHz. Only USB voice (2K8J3E emission) is permitted.

Most 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 you wish to change from LSB to USB without changing the spectrum occupied, you must retune your dial down about 3 kHz, as a careful study of **Fig 9.33** 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, you must not operate above approximately 14.347 MHz, since the transmission will be outside the band if you operates much higher. Operation with a suppressed carrier exactly on 4.0 MHz could be done on LSB if the signal is very clean, but is not recommended. Most modern rigs prevent out of allocated band transmissions but *do not* preclude the above cases of improper operation.

Today there are many new modes for text, speech and image transmission, and more will be developed in the future. 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 that 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 you can use one for these modes 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, *flat-topping* must be prevented. This results from overdriving the amplifier so that it goes above the design power limit and becomes non-linear.

SSB transmitters and most linear amplifiers use automatic level control (ALC) to prevent overdrive and flat topping. However, there are limits to ALC and flat topping 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.

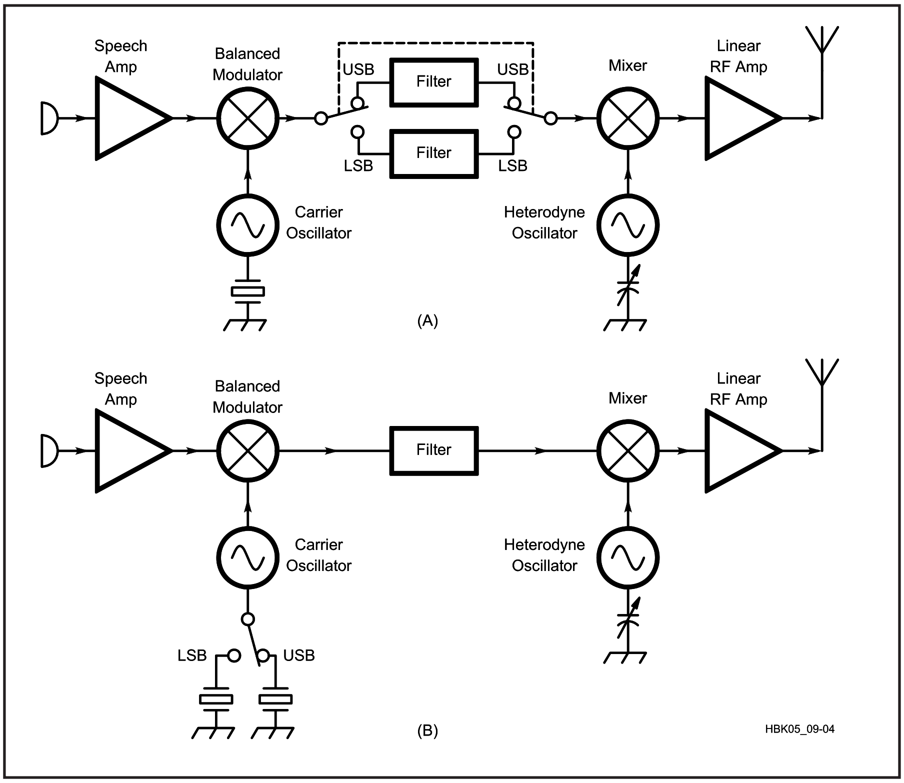

Fig 9.34—Block diagrams of filter-method SSB generators. They differ in the manner that the upper and lower sideband are selected.

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.

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 suppression. The bandwidth of an SSB filter is selected for the specific application. For voice communications, typical values are 1.8 to 3.0 kHz.

Fig 9.34 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 9.35 shows another method to obtain an SSB signal. The audio and carrier signals are each split into equal components with a 90° phase difference (called *quadrature*) and applied to balanced modulators. When the DSB outputs of the modulators are combined, one sideband is reinforced and the other is canceled. The figure shows sideband selection by means of transposing the audio leads, but the same result can be achieved by switching the carrier leads. The phase shift and amplitude balance of the two channels must be very accurate if the unwanted sideband is to be adequately attenuated. **Table 9.8** shows the required phase accuracy of one channel (AF or RF) for various levels of opposite sideband suppression.

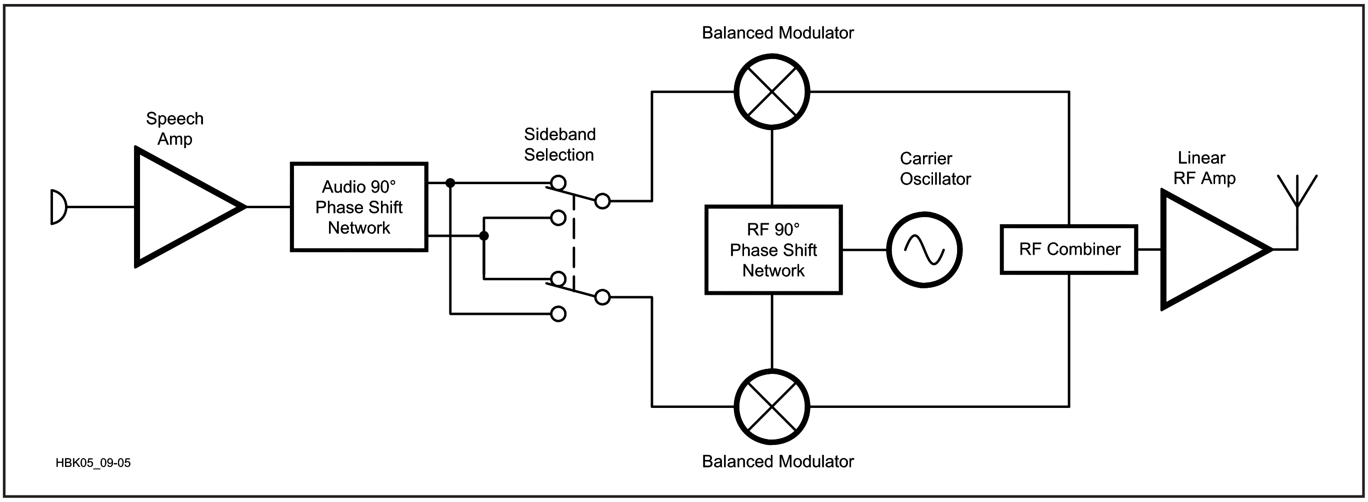

Fig 9.35—Block diagram of a phasing SSB generator.

Table 9.8
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

The numbers given assume perfect amplitude balance and phase accuracy in the other channel.

The 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. 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 low cost.

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 (Digital Signal Processing), 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.

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.

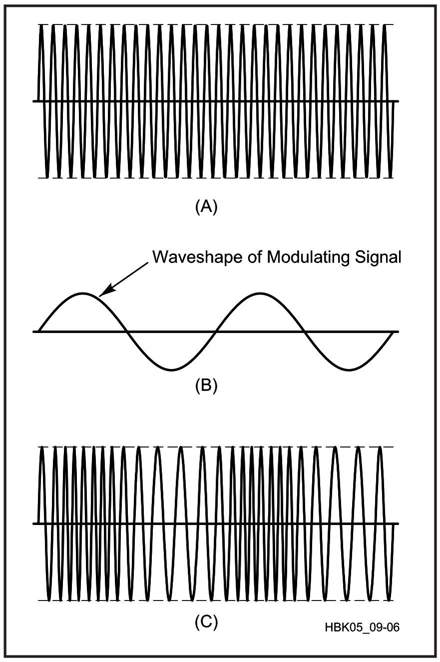

Fig 9.36 – Graphical representation of frequency modulation. In the unmodulated carrier (A) each RF cycle occupies the same amount of time. When the modulating signal (B) is applied, the radio frequency is increased and decreased according to the amplitude and polarity of the modulating signal (C).

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 9.36**. 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 9.37**, 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.

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 wideband 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 that 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 commercial 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 that extend to infinity but, fortunately, these drop off in amplitude rather quickly. As Armstrong surmised, ignoring sidebands that 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 hertz, 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. This is characteristic of an FM signal. This constant amplitude signal has the advantage of being easy to amplify without the need for 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. For normal speech, the power advantage 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 because they are 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 was covered earlier 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

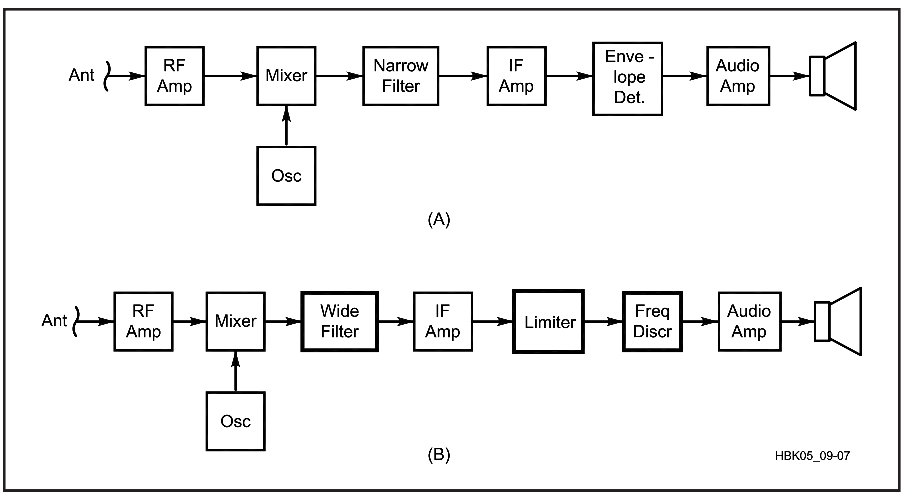

Fig 9.37—At A, block diagram of an AM receiver. At B, an FM receiver. Dark borders outline the sections that are different in the FM set.

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.

DIGITAL VOICE MODES

There is a risk in saying anything about an area that is developing rapidly both inside and outside Amateur Radio. Amateurs are watching digital voice developments in other radio services but not all are suitable models for Amateur Radio applications.

On MF and HF, transmission of digital voice is difficult owing to multipath propagation, QRM and noise. Several digital voice systems have been developed and more are expected. The most prominent contender is *Digital Radio Mondiale* (DRM). DRM is a non-profit consortium of broadcasters, manufacturers, educational and governmental organizations devoted to developing a single standard for digital sound broadcasting in long, medium and short wave bands. As of 2005, thousands of *software* radios were being used to hear regular DRM broadcasts from a growing number of countries. The software radio consists of a modified HF receiver, a sound card and computer software downloaded from DRM. See **www.drmx.org/** for details. As of mid 2005, there were few if any *hardware* DRM receivers available even at relatively high prices.

The DRM standard has several modes, some aimed at high fidelity music and others suitable for voice. 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 quality desired. See **Fig 9.38**. DRM produces excellent quality but is more subject to the effects of interference and propagation than DSB AM.

Another digital sound system used in the broadcasting service is called *IBOC*—In-Band On-Channel. The basic idea is to send a digital signal underneath an existing AM or FM program without one interfering with the other. Although used in the United States, IBOC hasn't caught on for international broadcasting. An article on IBOC at New York station WOR was presented in the March 2003 issue of *QST* (p 28).

The International Telecommunication Union approved a standard known as ITU-R Recommendation BS.1514, *System for digital sound broadcasting in the broadcasting bands below 30 MHz.* It describes DRM and IBOC, and compares the systems.

Amateur Radio Digital Voice

For HF Amateur Radio, digital voice has the potential to provide better quality than SSB. It could have other yet-to-be-exploited possibilities, such as adapting to conditions from a "robotic" sounding speech under marginal propagation to "arm-chair copy" when conditions are good. It is possible to imbed some ancillary information in the digital stream so the receiver will be able to display call signs, graphics and other information of interest to the stations in QSO.

In 2000, the ARRL Board of Directors created a Digital Voice Working Group to investigate and promote digital voice in the Amateur Radio Service. The pioneering work done by Charles Brain, G4GUO, and Andy Talbot, G4JNT, was published in the May-June 2000 issue of *QEX*. Their system was based on use of the AMBE 2020 encoder-decoder. It uses Orthogonal Frequency Division Multiplexing (OFDM) with 36 carriers in a band of 300-2500 Hz. At least one commercial version of AMBE 2020/G4GUO system is available in the amateur market. See Hallas, Joel, W1ZR, "AOR ARD9800 Digital Voice Modem," *QST*, February 2004.

The January-February 2003 issue of *QEX* (p 49) described a special Amateur Radio adaptation of the DRM system to fit inside a 3-kHz bandwidth. This system has taken on the name *HamDream* and some information can be found on the Web at **www.qslnet.de/member/hb9tlk/**.

These OFDM standards use many carriers spaced about 50 Hz apart, each using 16QAM (Quadrature Amplitude Modulation with 16 discrete states in each symbol) or some similar modulation scheme. To mitigate the effects of multipath propagation, the symbol rate must be limited to a few hundred bauds. Thus, the high bit rate needed for voice requires both multiple carriers and complex modulation. There are tradeoffs between complexity, weak signal sensitivity, reliability under difficult conditions, speech quality and latency. The most obvious way to generate and demodulate such a signal is to use a computer and a sound card.

While the same digital voice encoder-decoders could be used at MF/HF as well as VHF and above, it may be desirable to optimize the system for best performance in each frequency range. At MF/HF, the emphasis is naturally on reliability in the presence of fading and interference, while at VHF and UHF, it is possible to design for quality of speech reproduction and possibly multimedia (voice/data/image). See **High Speed Multimedia Radio** later in this chapter.

There is much room for innovation and experimentation in this field. A great deal of work will go into developing whatever digital voice mode 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* and *QEX* material on digital speech. Also check the following Web sites: **www.arrl.org/tis/info/digivoice.html**; **www.dougsmith.net**; **www.temple.edu/k3tu/digital_voice.htm**; **www.tapr.org/tapr/dv/**; **www.rac.ca/opsinfo/infodig.htm#Digital%20Speech** and **www.DRM.org.**

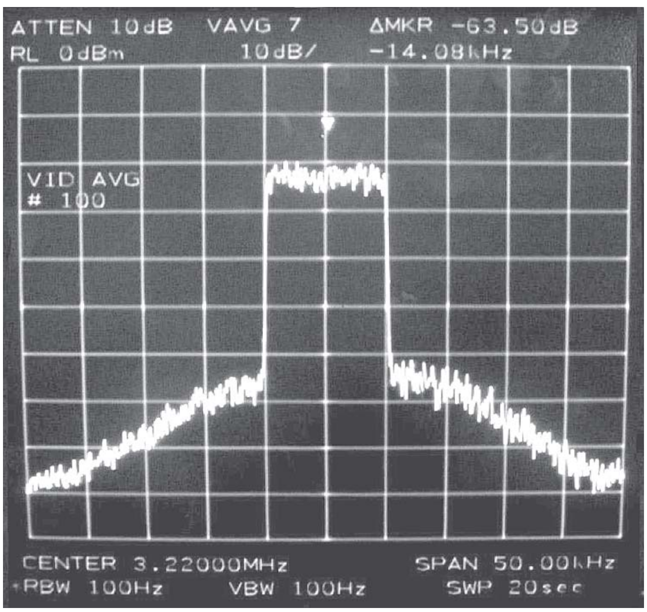

Fig 9.38—A DRM HF digital broadcast signal. Per-division resolution is 5 kHz horizontal and 10 dB vertical.

Image Modes

FACSIMILE

This section, by Dennis Bodson, W4PWF, Steven Karty, N5SK, and Ralph Taggart, WB8DQT, covers the several facsimile systems most commonly used in Amateur Radio 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 (Slow Scan TV), provides the potential for worldwide communications on the HF bands. Fax is the oldest of the image-transmitting technologies and has been for years 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 geostation-

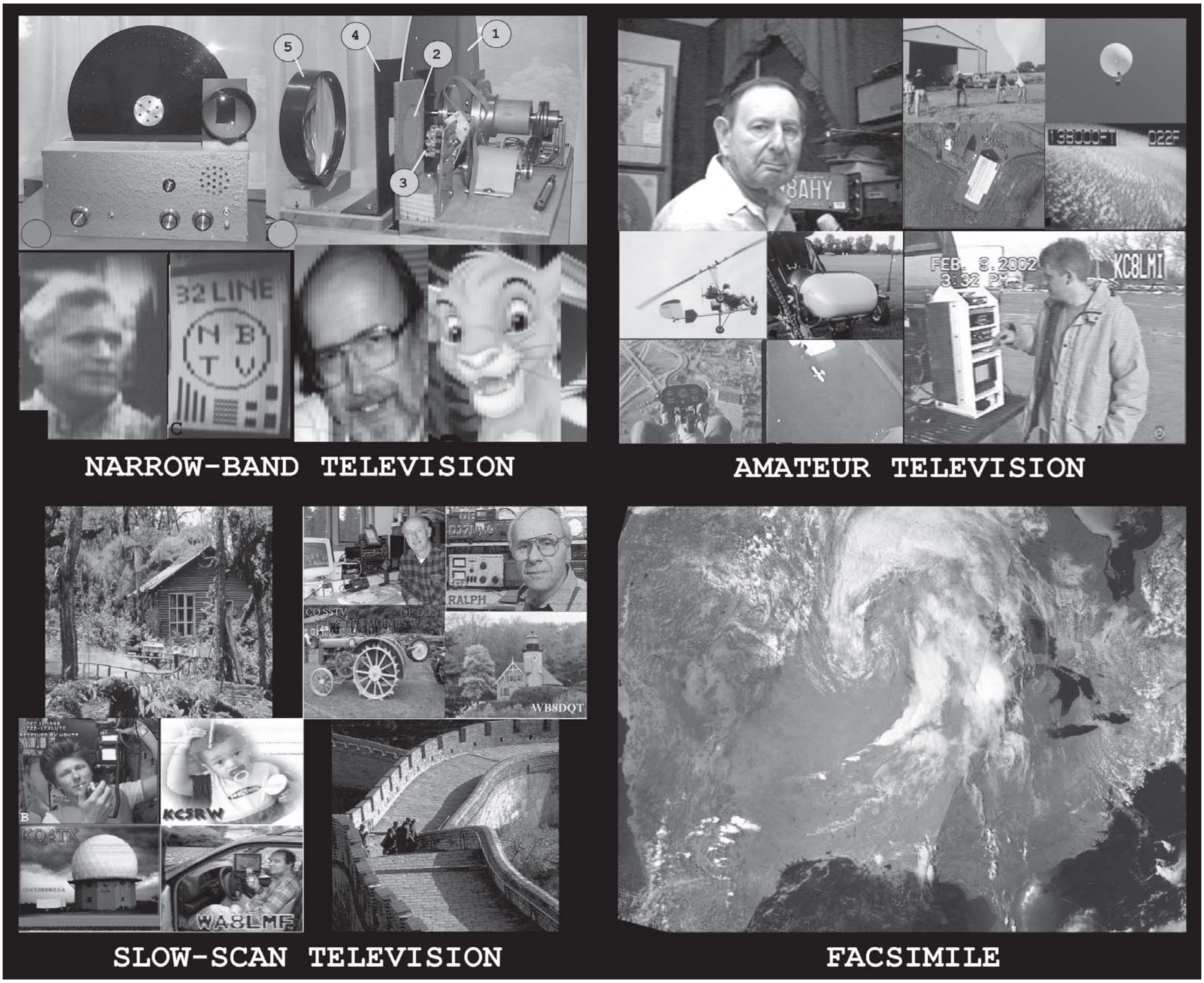

Fig 9.39—Amateur Image Communications encompass a wide range of activities, a few of which are illustrated here. Narrowband Television (NBTV) experimenters explore the history and technology of the earliest days of television by restoring or recreating mechanical TV gear while exploring the possibilities 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*.

ary satellites. Many of these images are retransmitted using fax on the HF bands.

The resolution of typical fax images greatly exceeds what can be obtained using SSTV or even conventional television (typical images will be made up of 800 to 1600 scanning lines). This high resolution is achieved by slowing down the rate at which the lines are transmitted, resulting in image transmission times of 4 to 10 minutes.

Modern personal computers have virtually eliminated bulky mechanical 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

The computer allows reception and transmission of various fax modes, where parameters such as line-per-minute rates and indices of cooperation can be altered by simply pressing a key or by pointing and clicking a mouse. Many fax programs are available as either commercial software or shareware. Usually, the shareware packages (and often trial versions of the commercial packages) are available by downloading from the Internet.

A good starting point is the ARRL software repositories. To get to them, set your browser to the *ARRL Web* and go to the FTP (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," p 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 3.X*. While specifically set up to decode and store weather-satellite APT pictures, it can also be used for HF-fax reception.

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

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

One well-known fax page on the Internet, complete with downloadable software, is posted and maintained by Marius Rensen; it contains listings of commercial fax transmissions for you to test your software or just SWL for interest. 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 always a good idea to virus check 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 with higher resolution.

HAL Communications Corporation has developed an interesting system that 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 that 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 the image density of the page being 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.

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.

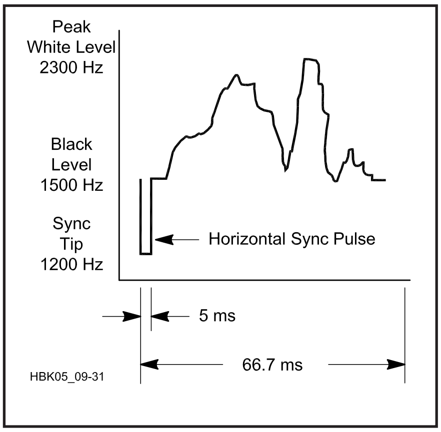

Fig 9.40—Early SSTV operators developed a basic 8-second black and white transmission format. The sync pulses are often called "blacker than black." A complete picture would have 120 lines (8 seconds at 15 ms per line). Horizontal sync pulses occur at the beginning of every line; a 30 ms vertical sync pulse precedes each frame.

Facsimile References

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

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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 small numbers of commercial equipment were very expensive and home-brewing was much too complicated for most people. Early attempts at computer-based systems were rather crude and frustrating to use.

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

The early SSTV 8-second transmission standard is illustrated in **Fig 9.40**. 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 History

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

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

In the late 1980s, yet another incompatible mode was introduced. The AVT mode is different from all the rest in that it has *no horizontal sync*. It relies on very accurate oscillators at the sending and receiving stations to maintain synchronization. If the beginning-of-frame sync is missed, it's all over. There is no way to determine where a scan line begins. However, it's much harder to miss the 5-s header than the 300-ms VIS code. Redundant information is encoded 32 times and a more powerful error-detection scheme is used. It's only necessary to receive a small part of the AVT header in order to achieve synchronization. After this, noise can wipe out parts of the image, but image alignment and colors remain correct. **Table 9.9** 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 9.41 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 in memory. 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 still prefer special dedicated hardware, but most of the recent growth of SSTV has been from these lower cost PC-based systems using sound cards and software.

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 modern SSTV stations look like **Fig 9.42**. Some sort of interface is used to get audio in and out of the computer. These can be external interfaces connected to a serial or printer port, an internal computer card specifically designed for SSTV or even a peripheral audio card.

Table 9.9
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.

Perhaps the single most significant breakthrough in computer-based SSTV is the wide range of Windows- and DOS-based programs using the PC's soundcard as the main transmit/receive interface. Many operators nowadays use the popular freeware program *MMSSTV* by JE3HHT (see **mmhamsoft.ham-radio.ch/mmsstv/**), with a simple hardware interface to go into and to come out of the PC's soundcard. Information on current computer SSTV software is available at **www.tima.com/~djones**. The subject of computer SSTV software and interfacing is also discussed at length in the *Image Communications Handbook* published by ARRL.

A simple "clipper" hardware interface to the computer's soundcard can be built with less than $15 worth of RadioShack parts. **Fig 9.43** shows such an interface circuit used for receiving and transmitting. Connect the output of T2 to the phone patch input (sometimes labeled LINE INPUT) of your transceiver, if it has one. Otherwise, you'll have to use the microphone input. R3 is set to the proper level for the audio going to the transmitter. You must set the audio signal into the transceiver at a level it can handle without distortion.

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

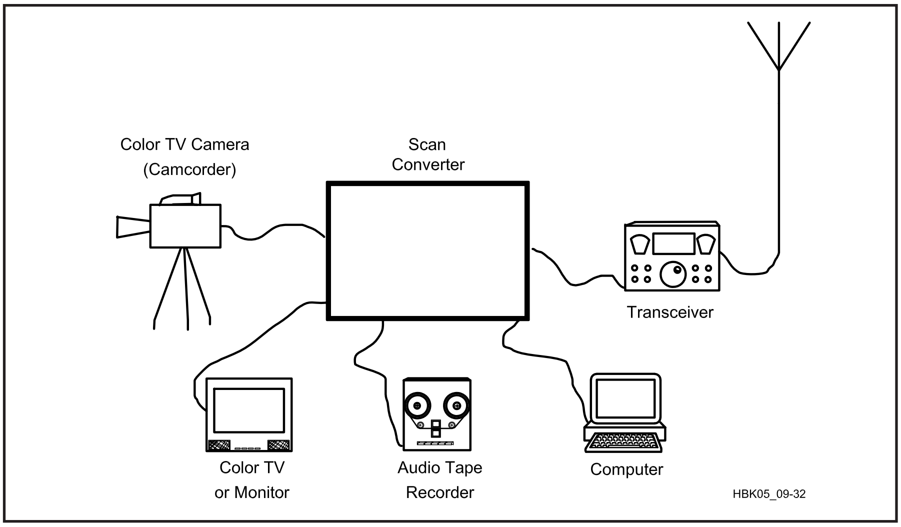

Fig 9.41—Diagram of an older SSTV station based on a scan converter.

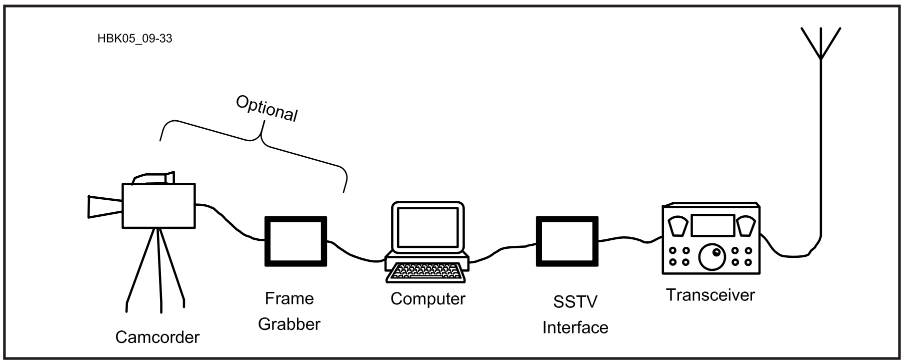

Fig 9.42—A modern SSTV station that utilizes the soundcard in a PC.

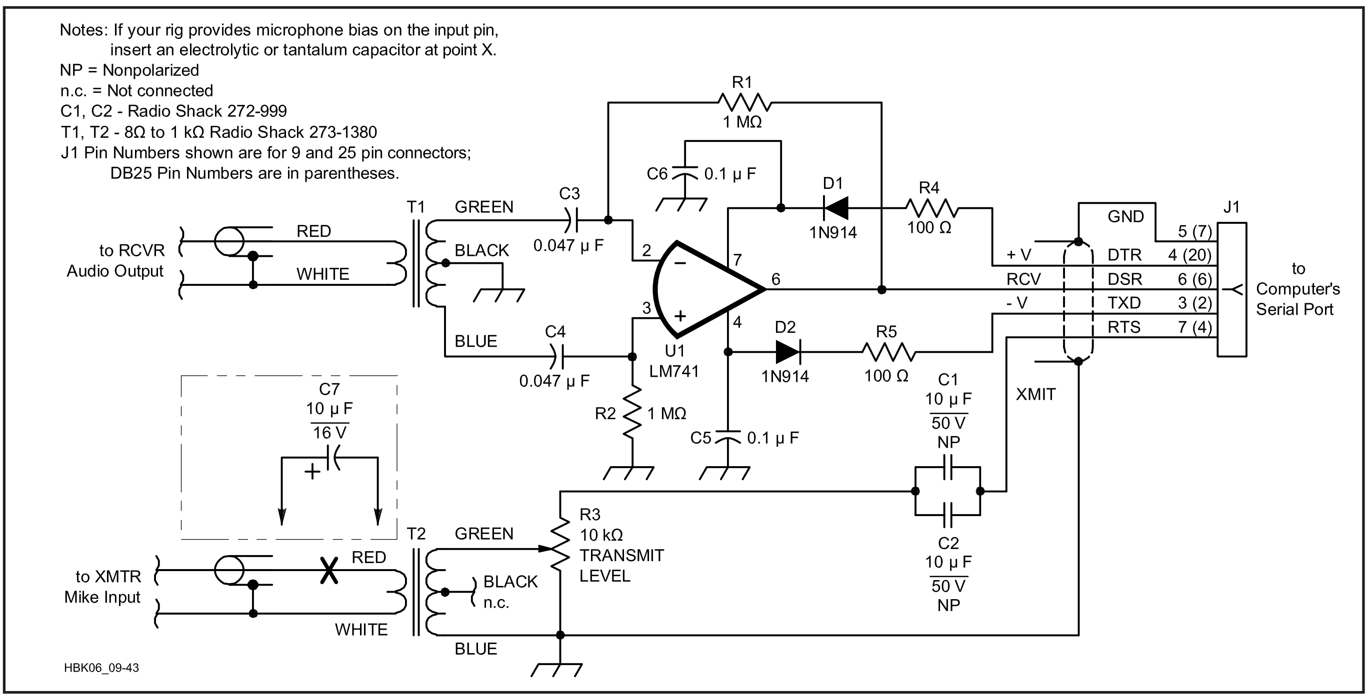

Fig 9.43—Schematic of the simple SSTV receive and transmit circuit from July 1998 *QST*. T1 and T2 are RadioShack 273-1380 audio-output transformers; the 20-µF, 50-V capacitor is a parallel combination of two RadioShack 272-999 10-µF, 50-V non-polarized capacitors; equivalent parts can be substituted. Unless otherwise specified, resistors are 1/4-W, 5%-tolerance carbon composition or film units. At J1, numbers in parentheses are for 25-pin serial port connectors; other numbers are for 9-pin connectors.

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 (see references) should be adequate for most cases.

Circuit component values aren't critical, nor is the circuit's physical construction. A PC board is available from FAR Circuits, but perf-board construction employing short leads works fine.

Digital Slow-Scan Television

DSSTV is a method of transmitting computer image files, such as JPEG or GIF over Amateur Radio, as described in an article by Ralph Taggart, WB8DQT, in the Feb 2004 issue of *QST*. The signal format phase modulates a total of eight subcarriers (ranging from 590 to 2200 Hz at intervals of 230 Hz. Each subcarrier has nine possible modulation states. This signal modulation format is known as *redundant digital file transfer* (RDFT) developed by Barry Sanderson, KB9VAK. RDFT is also used with SCAMP, described earlier in this chapter.

SSTV 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 free programs that only require trivial hardware interfaces to receive and transmit slow-scan 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 Bibliography

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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 that sent complete, sequential frames of red, then green and blue. Now obsolete.

Front porch—he 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.

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Table 9.10

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

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 that is mixed with the video. This section was written by Tom O'Hara, W6ORG.

Fig 9.44—Students enjoy using ATV to communicate between science and computer classes.

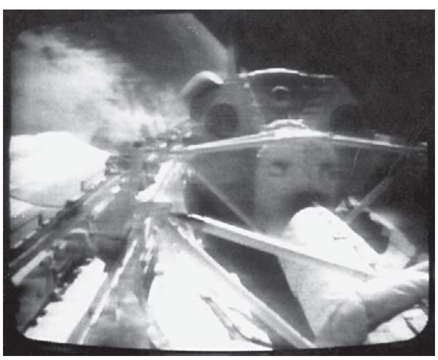


Fig 9.45—The ATV view shows the aft end of the Space Shuttle cargo bay during mission STS-9.

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 9.44** and **9.45**). Whatever the camera "sees" and "hears" is faithfully transmitted, including color and sound information. Picture quality is about equivalent to that of a VCR, depending on video signal level and any interfering carriers. All of the sync and signal-composition information is present in the composite-video output of modern cameras and camcorders. Most camcorders have an accessory cable or jacks that provide separate video and audio outputs. Audio output may vary from one camera to the next, but usually it has been amplified from the built-in microphone to between 0.1 to 1 V P-P (into a 10-kΩ load).

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

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

Table 9.11

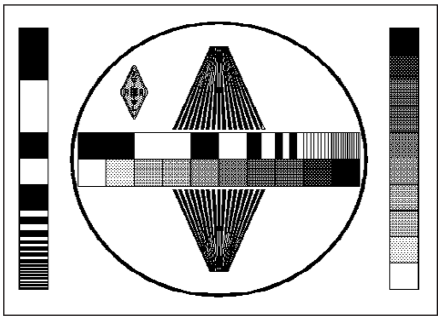
Bit encoding for 5.5 Mbps and 11 Mbps CCK transmissions.

Data Rate Mbps	CCK encoded bit	DQPSK encoded bit
5.5	2	2
11	6	2

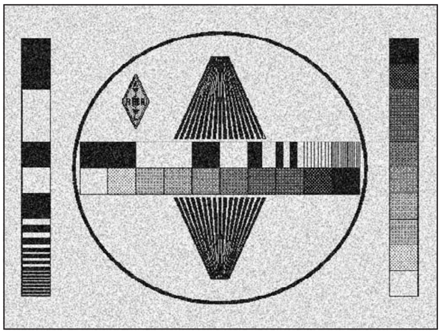
How Far Does ATV Go?

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 9.11**.) 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.

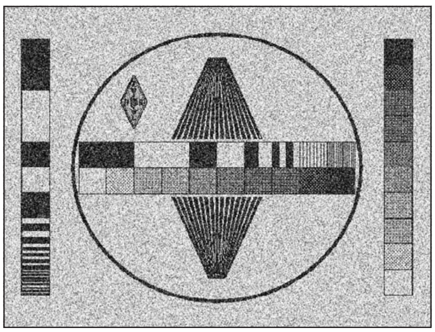
A snow-free, or "P5," picture rating (see **Fig 9.46**) requires at least 200 µV (–61 dBm) of signal at the input of the ATV receiver, depending on the system noise figure and bandwidth. The noise floor increases with bandwidth. Once the receiver system gain and noise figure reaches this



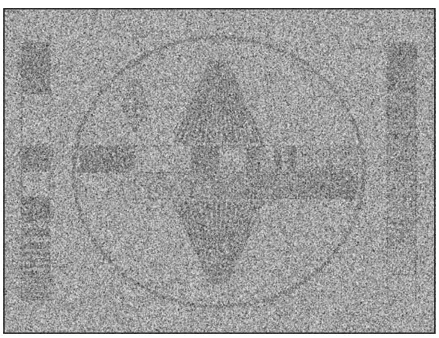
P5 —Excellent



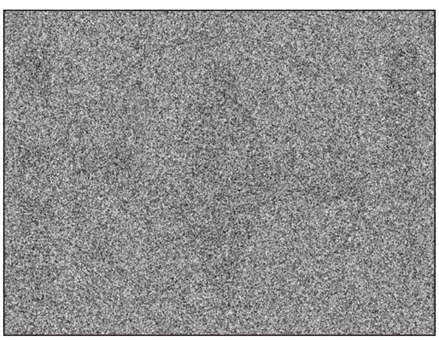
P4—Good



P3—Fair



P2—Poor



P1—Barely perceptible

Fig 9.46—An ATV quality reporting system.

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

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

Getting the Picture

Since the 70-cm band corresponds to cable TV 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 intermodulation 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 transmitting, 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

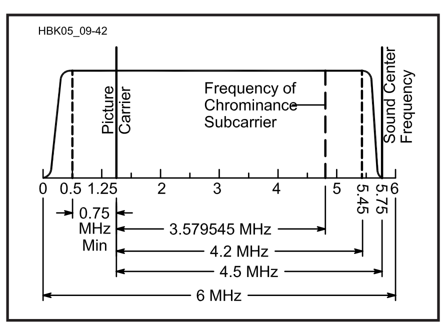


Fig 9.47—A 6-MHz video channel with the video carrier 1.25 MHz up from the lower edge. The color subcarrier is at 3.58 MHz and the sound subcarrier at 4.5 MHz above the video carrier.

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 9.47**.) 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 band plan options, there is room for no more than two simultaneous ATV channels in the 33- and 70-cm bands without interference. 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 stations. 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 bands (the 923.25 and 1253.25 MHz output frequencies are most popular). This 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 needed 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.

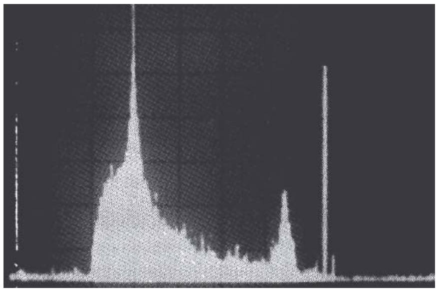


Fig 9.48—A spectrum-analyzer photo of a color ATV signal. Each vertical division represents 10 dB; horizontal divisions are 1 MHz. Spectrum power density varies with picture content, but typically 90% of the sideband power is within the first 1 MHz.

Fig 9.49—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.

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 ±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 9.48**, 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).

Narrowband 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 narrowband 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 reinsert 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 EIA-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 9.49**). The PC Electronics VOR-2 board has an automatic nine-minute timer, and it also has an end-of-transmission hang timer that switches to another video source for ID.

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 9.50**). 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 would be 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.

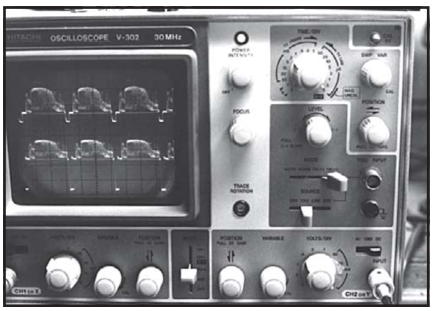


Fig 9.50—An oscilloscope used to observe a video waveform. The lower trace is the video signal as it comes out of the sync stretcher. The upper trace is the signal from the Mirage D1010-N amplifier.

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 9.51 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

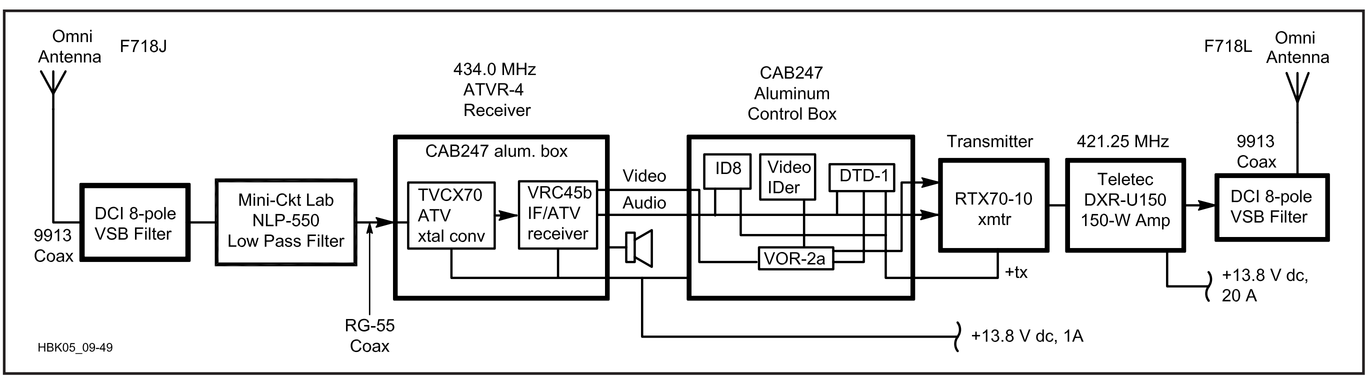


Fig 9.51—A block diagram of a 70-cm in-band ATV repeater. The antennas are Diamond omnidirectional verticals, which require 20 ft (minimum) of vertical separation to prevent receiver desensitization. The VSB filters are made by DCI; they have the proper band-pass characteristics and only 1 dB insertion 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.

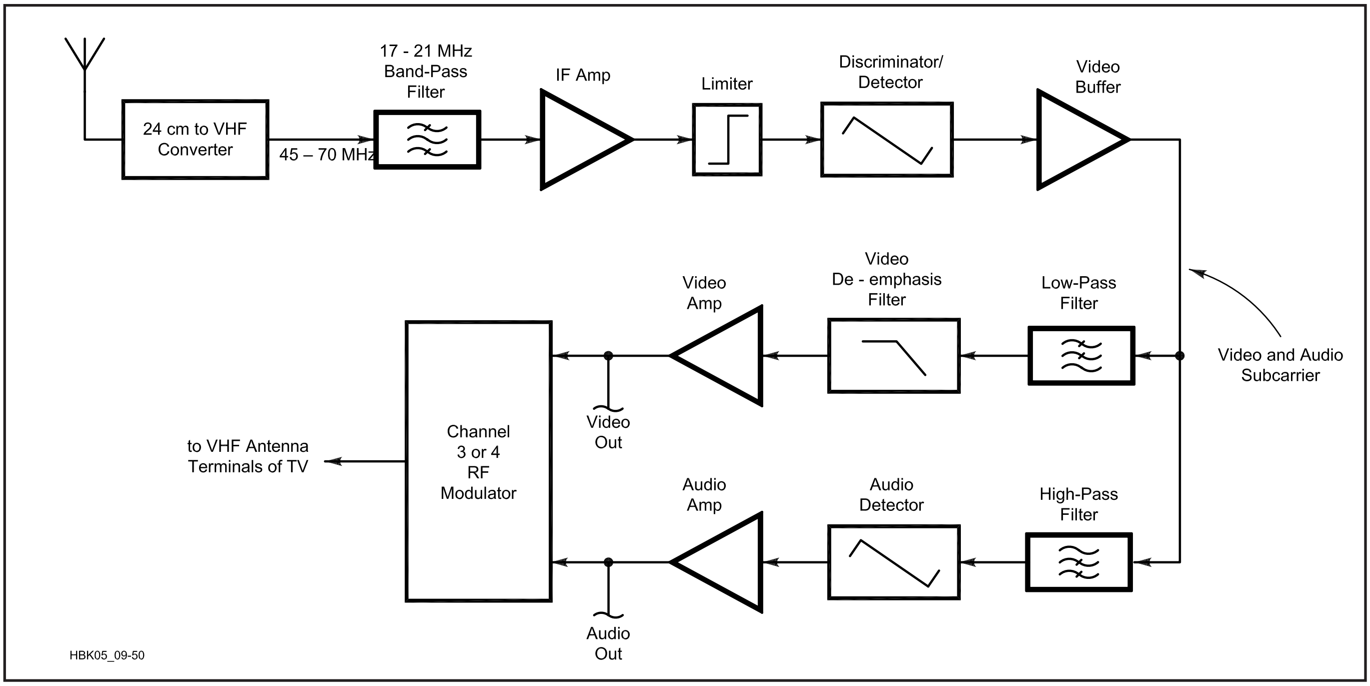


Fig 9.52—Block diagram of an FMATV receiver.

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 9.52** 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 Carson's 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 equip-

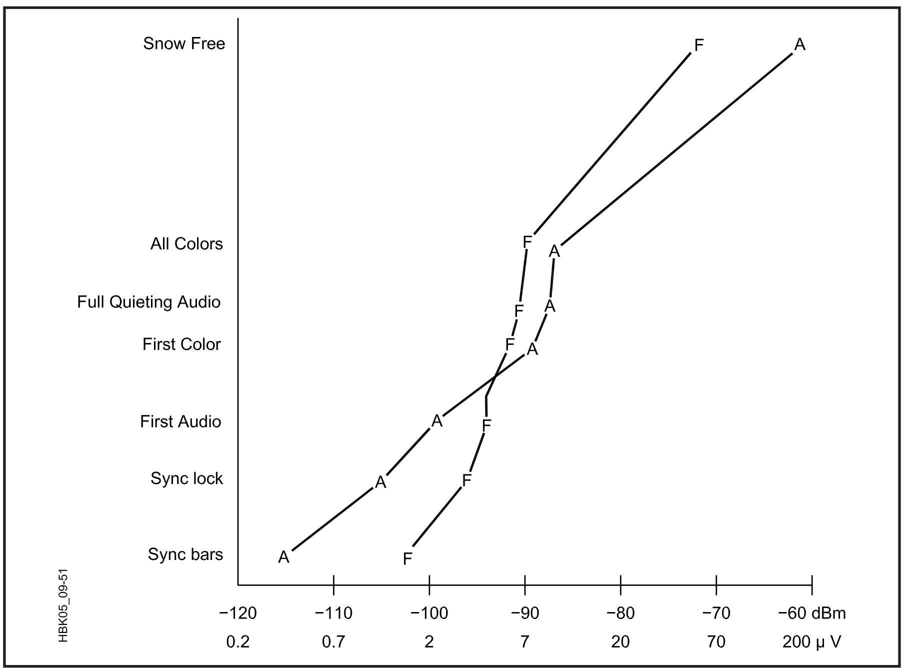


Fig 9.53—Two approaches to ATV receiving. This chart compares AM (A) and FM (F) ATV as seen on a TV receiver and monitor. Signal levels are into the same downconverter with sufficient gain to be at the noise floor. The FM receiver bandwidth is 17 MHz, using the US standard.

ment 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 four 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 9.53** compares AM and FMATV across a wide 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 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 use 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 then 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). These have been modified for higher power and other features, as well as having all four of the channels in the ham band using interface boards from PC Electronics.

Fig 9.54—N8QPJ mounted an ATV setup aboard this model Humvee.

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 9.54**.

DIGITAL AMATEUR TELEVISION (DATV)

German amateurs have lead the way in digital ATV. For the past few years, Uwe Kraus, DJ8DW, and others have had a stand at HamRadio—the large European Amateur Radio gathering in Friedrichshafen, Germany. The motivation for DATV is about the same as for commercial digital television, particularly high quality pictures even with weak signals and a distinctively smaller bandwidth than that occupied by analog TV. A breakthrough occurred in September 1998 when the DATV team transmitted digital pictures over a 62-mile path with a 2 MHz bandwidth at 434 MHz using MPEG-1 encoding.

See: **www.von-info-ch/hb9afo/histoire/news043.htm** and **www.von-info.ch/hb9afo/datv_e.htm**.

Further ATV Reading

Amateur Television Quarterly Magazine.

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Seiler, Thomas, HB9JNX/AE4WA, et al, "Digital Amateur TeleVision (D-ATV), *proc. ARRL/TAPR Digital Communications Conference*, **www.baycom.org/~tom/ham/dcc2001/datv.pdf**

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

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Spread Spectrum

Contributors to this section were André Kesteloot, N4ICK, John Champa, K8OCL, and Kris Mraz, N5KM. *The ARRL Spread Spectrum Sourcebook* contains a more complete treatment of the subject. The following information takes the subject from early experiments by the Amateur Radio Research and Development Corporation (AMRAD) to contemporary Amateur Radio use of spread spectrum technology for high-speed multimedia (HSMM) applications.

Spread spectrum originated in the 1930s, shrouded in secrecy. In 1942, Hollywood movie actress Hedy Lamarr and composer George Antheil were granted a patent for spread spectrum. Despite the fact that John Costas, W2CRR, published a paper on non-military applications of spread spectrum communications in 1959, spread spectrum was used almost solely for military purposes until the late 1970s. In 1981, the FCC granted 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. These FCC grants were intended to encourage the development of spread spectrum, which was an important element in commercial wireless systems that emerged in the 1990s.

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:

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

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 proportional to the channel bandwidth.

Within reason, however, you 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 narrowband, spread-spectrum users will also benefit, as narrowband 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.

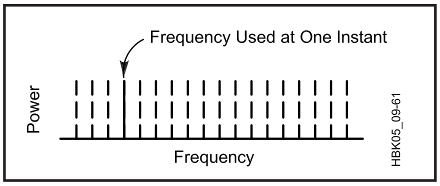


Fig 9.55—Power vs frequency for frequency-hopping spread spectrum signals. Emissions jump around to discrete frequencies in pseudo-random fashion.

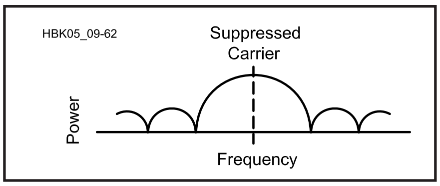


Fig 9.56—Power vs frequency for a direct-sequence-modulated spread spectrum signal. The envelope assumes the shape of a (sin x/x)^2 curve. With proper modulating techniques, the carrier is suppressed.

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 schemes are in existence, amateurs can use any of them as long as the 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 9.55**. Should some signal interfere with the receiver site on three of those discrete frequencies, the system will still have achieved reliable transmission 97% of the time. Because of the built-in redundancy in human speech, as well as the availability of error-correcting

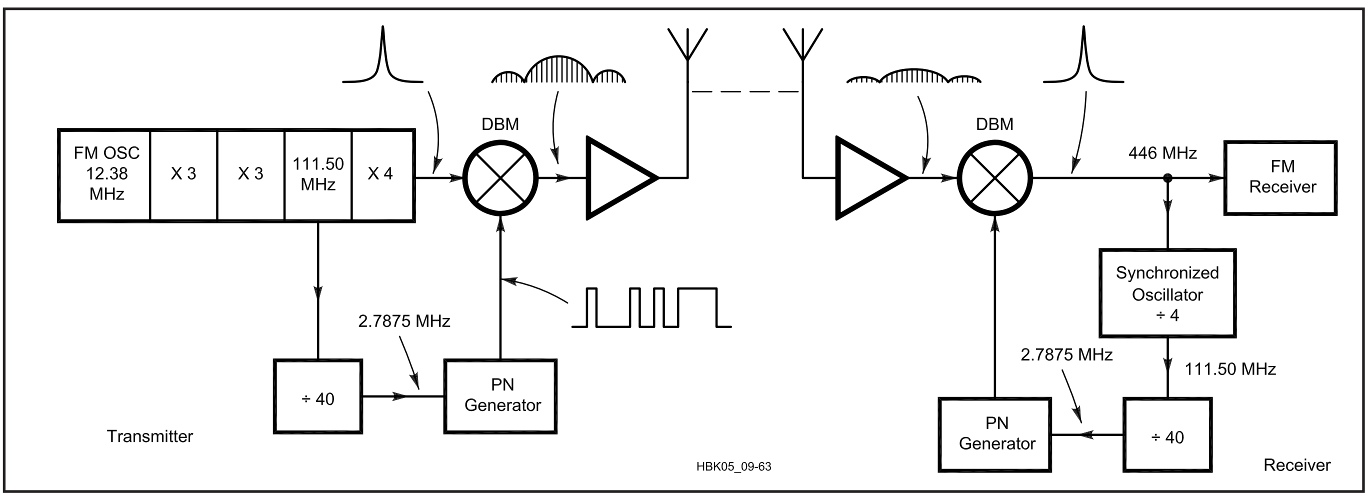


Fig 9.57—A block diagram of the practical spread spectrum link. The success of this arrangement lies in the use of a synchronized oscillator (right) to recover the transmitter clock signal at the receiving site.

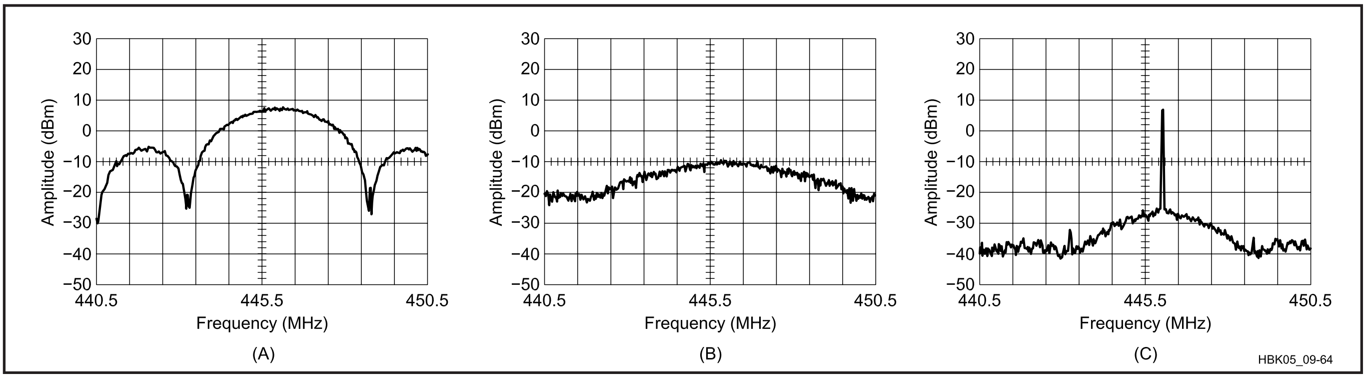


Fig 9.58—(A) The envelope of the unfiltered biphase-modulated spread spectrum signals as viewed on a spectrum analyzer. In this practical system, band-pass filtering is used to confine the spread spectrum signal to the amateur band. (B) At the receiver end of the line, the filtered spread spectrum signal 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.)

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 wideband transmission, as shown in **Fig 9.56**. At the receiver, a similar pseudo-random signal is reintroduced and the spread spectrum signal is correlated, or despread, while narrowband 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.

So far as the Amateur Radio community is concerned, particular benefit will be derived from the interference rejection just mentioned, since it offers both robustness and reliability of transmissions, as well as a 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

Experimentation sponsored by AMRAD began in 1981 led 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, André Kesteloot, N4ICK, offered a simple solution to the problem of synchronization. The block diagram is shown in **Fig 9.57**, and **Fig 9.58** 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. 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.

In 1997 TAPR started the development of a 1-W, 128-kbit/s, 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 commercial off-the-shelf (COTS) Part 15 spread spectrum devices used under § 97.311 of the FCC rules.

Emergence of Commercial Part 15 Equipment

Just as military surplus radio equipment fueled Amateur Radio in the 1950s, and commercial FM radios and repeaters snowballed the popularity of VHF/UHF amateur repeaters in the 1960s and 1970s, the availability of commercial wireless LAN (WLAN) equipment is driving the direction and popularity of Amateur Radio use of spread spectrum in the 2000s. FCC Part 15 documents the technical rules for commercial spread-spectrum equipment. The Institute of Electrical and Electronics Engineers (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 band at data rates of 1 and 2 Mbit/s. Next came the release of 802.11b, which provided the additional data rates of 5.5 and 11 Mbit/s but only for DSSS. FHSS was not carried forward. This was followed by 802.11g, which does not use SS but uses OFDM for data rates of 6, 9, 12, 18, 24, 36, 48 and 54 Mbit/s 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 certain parts of the 5 GHz band. It provides the same data rates as 802.11g. The currently unreleased 802.11n standard promises data rates in excess of 108 Mbit/s.

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 W 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 Mbit/s. The binary data stream modulates the carrier frequency using frequency shift keying. At 1 Mbit/s 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 Mbit/s by using 4GFSK modulation with shifts of ±75 kHz and ±225 kHz.

Direct Sequence Spread Spectrum

DSSS uses a fast digital sequence 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 Mbit/s input data stream becomes an 11 Mbit/s output data stream.

The DSSS radio, like the FHSS radio, can operate at data rates of 1 and 2 Mbit/s. 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 Mbit/s, and Differential Quadrature Phase Shift Keying (DQPSK) is used to achieve 2 Mbit/s. Fig 9.58 shows a typical 1 or 2 Mbit/s DSSS signal having a major lobe bandwidth of ±11 MHz (–30 dB). 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-Mbit/s data rate or 6 bits of data for the 11-Mbit/s data rate. This is known as *Complimentary Code Keying (CCK).* Both of these higher data rates use DQPSK for carrier modulation. DQPSK can encode two data bits per transition. **Table 9.12** shows how four bits of the data stream are encoded to produce a 5.5- Mbit/s data rate and eight bits are encoded to produce an 11-Mbit/s data rate. There are 64 different combinations of the 8-bit Complimentary Sequence that have mathematical properties that allow easy demodulation and interference rejection. At 5.5 Mbit/s, only four of the combinations are used. At 11 Mbit/s, all 64 combinations are used.

As an example, for an input data rate of 5.5 Mbit/s, four bits of data are sampled at the rate of 1.375 million samples per second. Two input bits are used to select one of four eight-bit CCK sequences. These eight bits are clocked out at a rate of 11 Mbit/s. The two remaining input bits are used to select the phase at which the eight bits are transmitted. **Fig 9.59A** shows a conceptual block diagram of a 5.5-Mbit/s CCK transmitter modulator, while Fig 9.59B shows an 11-Mbit/s modulator.

Orthogonal Frequency Division Modulation

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 thousand transitions per

Table 9.12

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

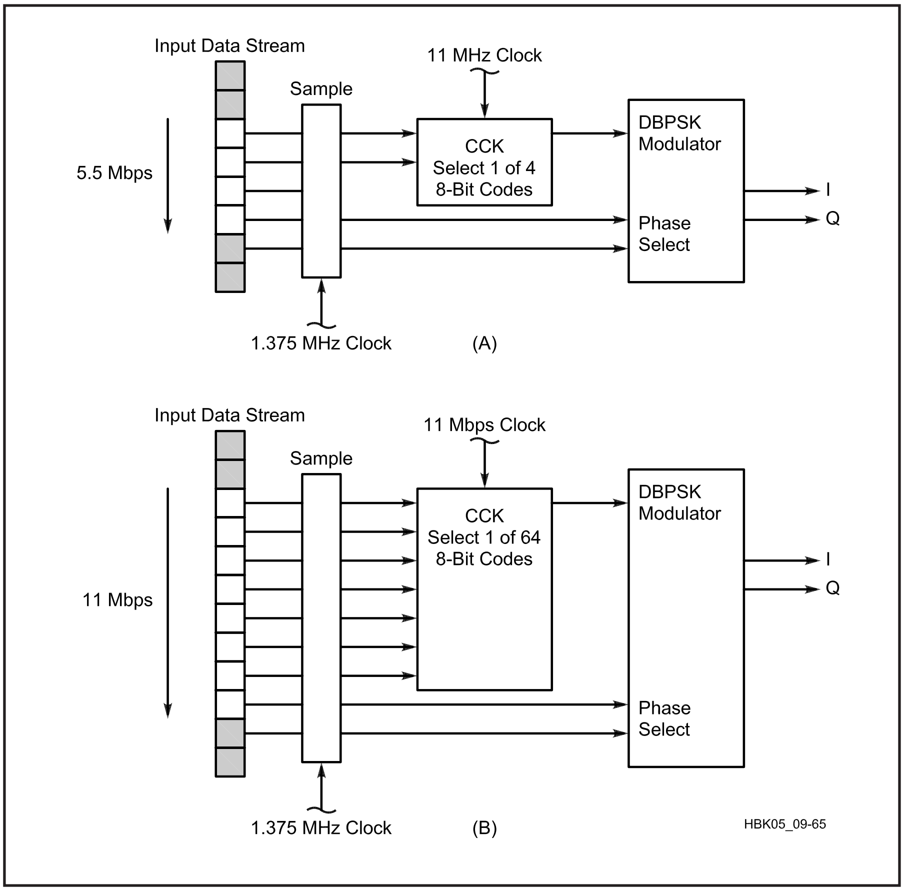


Fig 9.59—Conceptual block diagram of a modulator for a CCK Spread Spectrum transmitter. (A) 5.5 Mbit/s data rate. (B) 11 Mbit/s data rate. See text.

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 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.

OFDM radios can be used to transmit data rates of 6, 9, 12, 18, 24, 36, 48 and 54 Mbit/s 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 (one 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 Mbit/s.

48 carriers × 1 bit per carrier × 1/2 R = 24 bits (effective)

24 bits × 250 kilo transitions per second = 6 Mbit/s.

Table 9.13 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.

Table 9.13
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

Spread Spectrum References

Dixon, *Spread Spectrum Systems*, second edition, 1984, Wiley Interscience, New York.

Dixon, *Spread Spectrum Techniques*, 1976, IEEE Press, New York.

Kesteloot, Ed., *The ARRL Spread Spectrum Sourcebook* (Newington, CT: ARRL, 1990). Includes Hershey, *QST* and *QEX* material listed separately here.

"Poisson, Shannon and the Radio Amateur," *Proceedings of the IRE*, Dec 1959.

Multimedia Systems

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 appointed 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 identified two initial goals for the 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. This includes 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 a 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 a 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* program 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* program 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 to 40 kbit/s) 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. 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

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.

AP—Access Point

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).

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 Mbit/s and 2 Mbit/s 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.

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

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

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 that 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 can be invaluable in estimating the severity of the situation, planning appropriate responding resources and other reactions. The Emergency Operations Center (EOC) can 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 is needed to accomplish such immediacy 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 be to run higher power and to 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

rates in addition to being backward compatible with DSSS at 1, 2, 5.5, and 11 Mbit/s specified in 802.11b.

IEEE 802.11n—An IEEE standard specifying data rates up to 250 Mbit/s and being backward compatible with 802.11a and 802.11g.

IEEE 802.16—An 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 0183—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.

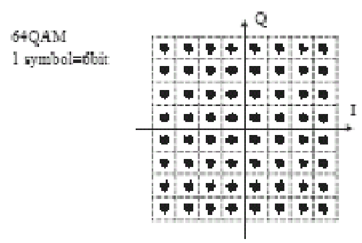
OFDM—Orthogonal Frequency Division Multiplexing. A modulation method in which the communication channel is divided into multiple subcarriers each being individually modulated. While not meeting the Part 2 definition of spread spectrum the FCC has given specific authorization for OFDM systems.

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.

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 call sign 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.

WiMAX—Familiar name for the IEEE 802.16 standard.

WISP—Wireless Internet Service Provider

WLAN—Wireless Local Area Network.

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 do you 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.

The core of any HSMM radio station 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, 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 is that if you cannot hear the other station on the 1.2-GHz FM radio, you probably will not be able to link up the HSMM radios either.

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.

Fig 9.60—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).*

HSMM RADIO REPEATERS

Access Points

What hams would call a repeater, and computer buffs would call a hub, the WiFi industry refers to as a *wireless 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 point-to-point radio network. If you want 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 is 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.

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 would be very dangerous. So unless you have somebody else driving 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 just called, *NetStumbler*. All operating systems have monitoring programs that are available. *Linux* has *Kismet*; *MAC OS* has *MacStumbler*. Marius Milner has a version for the PocketPC, which he calls *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 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 tri-pod 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 0183 formatted data and computer interface cable. This allows the monitoring utility to record where HSMM stations are located 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 WiFi equipment will automatically associate or link to such stations, if they are not encrypted, and many are not (that is, 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.

Fig 9.61—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).*

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 the 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 (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 with 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 multipath signal distortion. This multipath 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 using helical antennas, patch feed-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, so long as both antennas at both ends of a radio link have

Fig 9.62—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)*

broad bandwidths and 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 indicates 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 the 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 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 W 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 feed-line 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 analysis showing that 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 watt 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 necessary 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 another 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 that you don't want 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 already 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 software 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. The firewall should 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, you could use a different antenna polarization than the Part 15 station, or you could operate with a directional antenna oriented toward the desired coverage area rather than using an omni-directional antenna.

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, since 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 overpower 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 Mbit/s 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-GHz band is also being investigated. The COTS 802.11a modulation gear has OFDM channels that 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 that, 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 an 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 that 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 by visiting *ARRLWeb* at **arrl.org/hsmm/**.

For more details about 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 p 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*.

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