

## Modulation Methods, Part 1:

### CW and AM

Transmitting information on some form of carrier (usually electromagnetic radiation) depends on a process called modulation. The ability to generate the required amounts of energy at any frequency is, of course, necessary for transmission of intelligence via electromagnetic radiation. If this energy cannot have information applied to it in some way, it is useless for most communications purposes. Parameters that can be controlled are the amplitude or power level, frequency of the radiation, and the phase of the waveform of the energy with respect to a known reference.

In addition, some means of extracting this information from the transmitted radiation is needed. This process is usually called demodulation or detection. For this discussion, we will assume that a carrier consisting of electromagnetic radiation in the radio-frequency spectrum will be used. Any frequency can be used, but we will assume it is one between 10 kHz and 300,000 MHz.

These limits are those presently allocated for communications purposes. Ten kHz is low enough in frequency to be audible as a high-pitched tone, if a headphone or speaker is used. Above 300,000 MHz, which is about as high as can be readily handled by microwave techniques, the radio spectrum is called the submillimeter region.

Above about 30,000,000 MHz (10 microns wavelength), it is the far infrared region of the spectrum. This radiation can be felt as heat rays. Visible light starts at about 430,000,000 MHz (0.7 microns wavelength), perceived by the eye as red light. Lasers operate in the far infrared to visible spectrum, and these can also be modulated.

These frequencies allow almost unlimited modulation bandwidth and are used for fiber-optic communications. Even though we will confine this discussion to radio frequencies, be aware that other forms of radiation can also be modulated. The same theoretical concepts will apply, although the physical methods and techniques will generally be very different from those used in the radio spectrum.

#### Digital-Type Modulation

The simplest and oldest form of modulation is a digital type, that of turning on and off a source of energy (light, RF carrier, etc.). (See Fig 1.) Originally, lanterns with shutters were used. Then, the Morse telegraph used a DC current that was turned on and off to form the dashes and dots of Morse Code. Later, radio waves were used to do the same thing. A key turns a transmitter on and off, generating a continuous wave (CW) signal. Although this technique is not used as widely today, it remains one of the simplest and most efficient means of communication. Only a very simple transmitter, even a very simple oscillator circuit with a single transistor, is needed. The inherently narrow bandwidth occupied by the signal permits the use of a very narrow-band receiver (20 to 100 Hz). This setup enables low-power transmitters to send signals thousands of miles. Reception with a relatively simple receiver is possible.

Radio amateurs do such transmission quite often. This activity is called "QRP operation," where QRP is CW shorthand for reduced or lowered transmitter power. Worldwide contacts have been made with only a milliwatt of power in

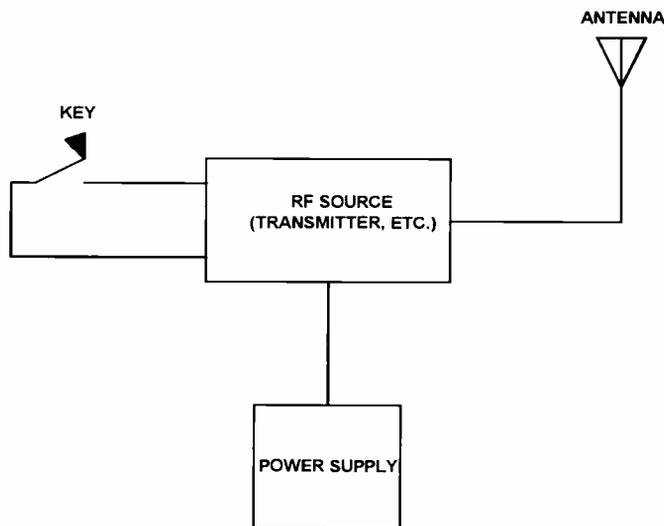
the HF region of the spectrum (2–30 MHz), often enough as to be almost commonplace.

Before we discuss modulation methods, let's look at one factor that limits the potential performance of any given system. This factor is the noise inherent in any physical system.

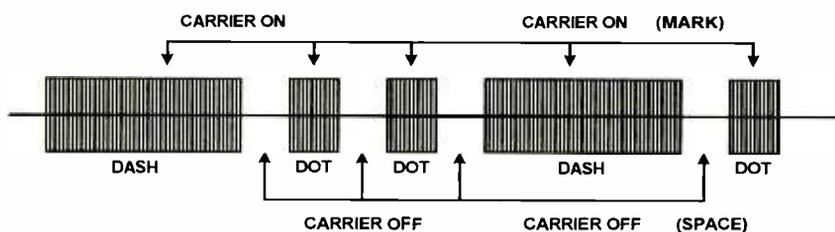
#### Let's Do The Math

The limiting factor on how weak a signal can be and still be received depends on the receiver bandwidth, temperature, and type of modulation. In the following discussion, some high school math is used (algebra and trigonometry). Sorry for the math, but there is really no better way to present this information properly. Mathematics is a fascinating field and the language of science. If you really want to get into electronics or other aspects of engineering or the physical sciences, you need mathematical proficiency to fully understand many theoretical and practical design concepts. If you would rather not follow the math, you will have to take our word for the figures and numbers we use.

The noise power measured in watts in any bandwidth is given by the formula  $\text{Power} = KTB$ . See a physics textbook for the derivation of this equation if you are curious.  $K$  is Boltzmann's constant, which is equal to  $1.38 \times 10^{\text{exp}(-23)}$  joules/degree K;  $T$  is the absolute temperature in degrees Kelvin; and  $B$  is the bandwidth in cycles per second (Hz). One joule is equal to one watt for one second and is a measure of energy. At normal room temperature (taken as 20° C or 68° F),  $T$  is 293° K. Multiplying this out, at room temperature in a 1-Hz



**SIMPLE CW (CONTINUOUS WAVE) TRANSMITTER**



**SIGNAL PRODUCED BY CW TRANSMITTER**

**FIGURE 1  
CONTINUOUS WAVE (CW) TRANSMISSION**

bandwidth we have a noise power of  $4.04 \times 10 \exp(-21)$  watts of power. The watt is inconveniently large for this work, so the milliwatt (.001 watt) is used instead. This noise level is then  $4.04 \times 10 \exp(-18)$  milliwatts. Since in RF systems we are usually dealing with very large variations in power levels, the decibel system is used to avoid inconveniently large or small numbers and ratios. Converting this power level to the more useful measurement of decibels referred to as a milliwatt and remembering that a decibel is a logarithmic ratio of two power levels results in:

$$\text{dB} = 10 \log (P_2/P_1) \text{ for a ratio of } 4.04 \times 10 \exp(-18) \text{ where } P_2/P_1 \text{ is the power ratio}$$

$$\text{dB} = 10 \log (4.04 \times 10 \exp(-18)) = 10 \log 4.04 + (-18) \times 10 \log 10$$

$$\text{dB} = 10 (0.606 - 18) = -173.94 \text{ dBm (very closely equals } -174 \text{ dBm)}$$

Note:

$$\text{dB} = 10 \log (P_2/P_1) \text{ where } P_2 \text{ and } P_1 \text{ are power levels}$$

$$\text{dB} = 20 \log (V_2/V_1) \text{ where } V_2 \text{ and } V_1 \text{ are voltage levels}$$

$$\text{dBm} = \text{decibels with respect to 1 milliwatt reference power level}$$

$$0 \text{ dBm} = 1 \text{ milliwatt} = 0.223 \text{ volts RMS in a 50-ohm system}$$

As an example, the following figures are typically those signal levels one would encounter in operating HF (2-30 MHz) SSB or CW amateur radio equipment. Figures have been rounded off, are for a 50-ohm impedance (the usual situation), and are approximate within a

percent or so. These readings are those that would be seen on a typical short-wave receiver signal-strength ("S") meter:

-20 dBm = 22.3 millivolts, "pegs:" S meter, very strong signal

-47 dBm = 1 millivolt (approx.), S9 + 26 dB, strong signal level

-60 dBm = 223 microvolts, S9 + 13 dB, a good signal

-73 dBm = 50 microvolts, an S9 (average) signal

-87 dBm = 10 microvolts, (S7+), a weaker but still decent signal

-107 dBm = 1 microvolt, (S3), weak, SSB marginal, CW is OK

-127 dBm = 0.1 microvolt, very weak, only CW readable

The dBm is independent of the resistance or impedance of the system, but the impedance must be specified for it to have any relation to actual voltages or currents. Since noise voltage is related to power and resistance, and power is  $V^2/R$ , then the noise voltage across a resistance is

$$V_{\text{noise}}^2 = KTB/R$$

and

$$V_{\text{noise}} = \sqrt{KTB}/\sqrt{R}$$

In any generator with a voltage  $V$  and internal resistance  $R$ , the maximum power available to the load occurs when  $R_{\text{load}} = R_{\text{generator}}$ . This is the maximum power transfer theorem. The load power will be  $(V/2)^2/R$  or  $V^2/4R$ . Then the noise voltage will be :

$$V_{\text{noise}} = \sqrt{4KTBR} \text{ where } K = 1.38 \times 10 \exp(-23)$$

$$T = \text{Temp deg K; note: deg K} = \text{deg C} + 273$$

$$B = \text{Bandwidth Hz}$$

$$R = \text{Resistance in ohms}$$

Normally, we use power levels in noise work as it is more convenient. In a system, for example, the noise power

level is inherently -174 dBm in a 1-Hz bandwidth. Considering a 10-kHz bandwidth typically used in an AM broadcast receiver, we could take the ratio of 10 kHz to 1 Hz as 10,000 to 1. This is a 40-dB power ratio (10 log 10,000, or  $10 \times 4$  since the log of 10,000 is 4; therefore a 10,000 to 1 ratio, which is 40 dB). Adding 40 dB to -174 dB gives -134 dBm, or 134 dB below a milliwatt. In a 50-ohm system, 1 milliwatt equals 0.223 volts RMS across 50 ohms. Since:

$$\text{dB} = 10 \log (P2/P1)$$

then

$$\log P2/P1 = \text{dB}/10$$

and

$$P2/P1 = \text{antilog} (\text{dB}/10)$$

Here we divide the dB ratio by 10 and find the inverse log of the result, in this case 13.4. Since we want the voltage ratio—the square root of the power ratio for a given resistance, we can divide the logarithm by two, which gives 6.7. Finding the antilog of this will give the voltage ratio that 134 dB represents

$$\text{antilog} (6.7) = 5.01 \times 10 \exp(6)$$

or a 5.01 million to one ratio.

Thus, -134 dBm =  $0.223/5.01 \times 10 \exp(-6)$ . It comes out to be 0.045 microvolts across 50 ohms—the noise-power level in a perfect receiver with a 10-kHz bandwidth. Theoretically, this is the *minimum detectable signal* (MDS), assuming that the received signal power equals the noise power. (This is only an assumption, as techniques exist for detecting signals below the noise, and the MDS also depends on the signal processing used in the receiver.)

### Dealing With Noise

A good Morse code operator can usually copy a weak signal that is at the receiver noise level. However, receivers are not perfect. Good receivers used for VHF-UHF work may have noise figures of 1 dB, which means that the receiver noise level is 1 dB above ideal. A typical HF receiver has a 10- to 20-dB noise figure; thus, the signal detectable in a 10-kHz bandwidth, in this case, would be 10- to 20-dB higher (a three to ten times voltage ratio).

External and atmospheric noise limits

reception anyway, so noise figures lower than 15 dB or so are of dubious advantage in an HF receiver, especially below 20 MHz. (Strong signal performance is generally more important in the HF region). It would then be ten times 0.045 microvolts, or 0.45 microvolts.

However, for voice work, at least a 6-dB signal-to-noise ratio is needed for barest intelligibility, with 10 dB being more like it. This requirement raises the minimum input signal to the 1- to 1.5-microvolt level for copying a voice signal, such as that from an AM medium-wave or short-wave station. You would probably not listen to this program for a long time, as it would be quite noisy. Another 10- to 20-dB signal level would be needed for comfortable copying, depending on how badly you wanted to listen to it, bringing the signal level up to 5 to 15 microvolts for reasonable reception. The important thing is the signal-to-noise ratio and not just the signal level. In noisy reception areas, stronger signals are needed. For any system, the bandwidth is important in optimizing the quality of the received signal: too wide, we get more noise and poorer signal-to-noise ratio; too narrow, we may lose some of the information in the signal or introduce distortion.

### The Morse CW Signal

In the case of the Morse CW signal, the necessary bandwidth can be estimated by examining the signal. (See Fig 1.) At a speed of 25 words per minute (a fairly rapid, but comfortable speed typical of experienced CW operators), this would be about 125 Morse characters per minute, assuming an average five-letter word. This is roughly one letter and space per 500 milliseconds.

Taking the worst case, the Morse code symbol for the number 5 has five consecutive dots and can be considered as a square wave with five complete cycles in half a second. This is equivalent to a 10-Hz square wave. A square wave consists of frequencies that are mainly fundamental, and the third and fifth harmonics (odd) of the fundamental.

If the square wave is asymmetrical (typical for Morse Code as there are dots, dashes, and spaces), there are second and fourth (even) harmonics, also. Although it is an approximation, a square wave decent enough to be copied as a Morse Code signal consists of harmonics up to at least the fifth. Therefore, a minimum bandwidth of 50

to 100 Hz would be needed in this example, for 25 words-per-minute speed of transmission. This minimum allows for some tuning error and short-term receiver drift. More than this, the signal-to-noise ratio will start to decrease. Less bandwidth will cause loss of the higher harmonics and rounding of the waveforms to where the signal would be difficult to copy, unless the sending speed were reduced.

If speeds of five words per minute were used, bandwidth could be reduced accordingly at the expense of speed of transmission. For this reason, very weak signal CW work is done at slow transmission speeds—to allow narrow bandwidth and an increase in effective receiving sensitivity.

In practice, many receivers for amateur radio CW use 200- to 400-Hz bandwidth, as it allows for more comfortable tuning by the operator, for some receiver drift, and less costly filtering. Even with 400-Hz bandwidth and a 20-dB noise figure, the minimum discernable signal level is around 0.1 microvolts, depending on the operator's skill and hearing acuity.

In most cases, external noise will be the limit anyway. A 0.5-microvolt signal is typically comfortable to copy. Contrast this with the 5- to 15-microvolt figure needed for AM or 2 to 5 microvolts for SSB for marginal copy, and you can readily see the advantages of CW techniques using Morse code or other forms of slow-speed digital modulation in weak signal work. In this era of cheap and powerful computers, the Internet, cell phones, and sophisticated equipment, simplicity still is important.

It is a sobering fact and somewhat amusing to note that the use of plain old (obsolete....?) Morse Code, 1940-era radio technology, with a skilled operator can give reliable and dependable emergency communications when all else is knocked out. Only a simple transmitter, a shortwave receiver, and a length of wire strung up between two trees or other supports are needed to get a station on the air. A 12-volt auto battery will do for power. In emergency situations, communications might be impossible using much more sophisticated equipment, whose operation depends on a vulnerable infrastructure destroyed or rendered inoperable in a natural disaster; or made useless and/or inaccessible during a lockdown, terrorist, or national emergency. Do not count on using the

Internet, the telephone system, or your cell phones at these times.

### Amplitude Modulation

The next form of modulation that evolved was probably *amplitude modulation*, called AM. In this case, the amplitude of the signal is modulated in some way by the waveform of the intelligence to be transmitted. Here, the envelope of the transmitted AM signal is a replica of the modulating AM signal. (See Fig 2.) Usually, the carrier is a sinusoidal waveform, and the modulation is audio or data. The modulating waveform can be represented as a superposition of harmonically related sine-wave components (Fourier's Theorem). The amplitude of the carrier waveform is modulated by the modulation (audio/data), and a mixing action takes place.

The carrier waveform can be represented as:

$$V_c(t) = A \sin \omega_c T$$

where

$$\omega_c = \text{freq. radians/second} = 2\pi \times \text{Frequency in Hz}$$

A = peak amplitude of sinewave in volts

$V_c(t)$  = Instantaneous voltage of carrier

T = time

If a waveform is available, having an amplitude that swings between zero and  $V_m$  volts, described as:

$$V_m(t) = 1 + M \sin \omega_m T$$

where

$V_m(t)$  = total modulating signal

$V_s(t)$  = modulating signal

$\omega_m$  = modulating frequency rad/sec

T = time

M = relative amplitude of modulation

(M is 0 minimum to 1 maximum)

then this signal can be used to modulate a carrier signal.

If these two signals are mixed (multiplied together) in a modulator circuit

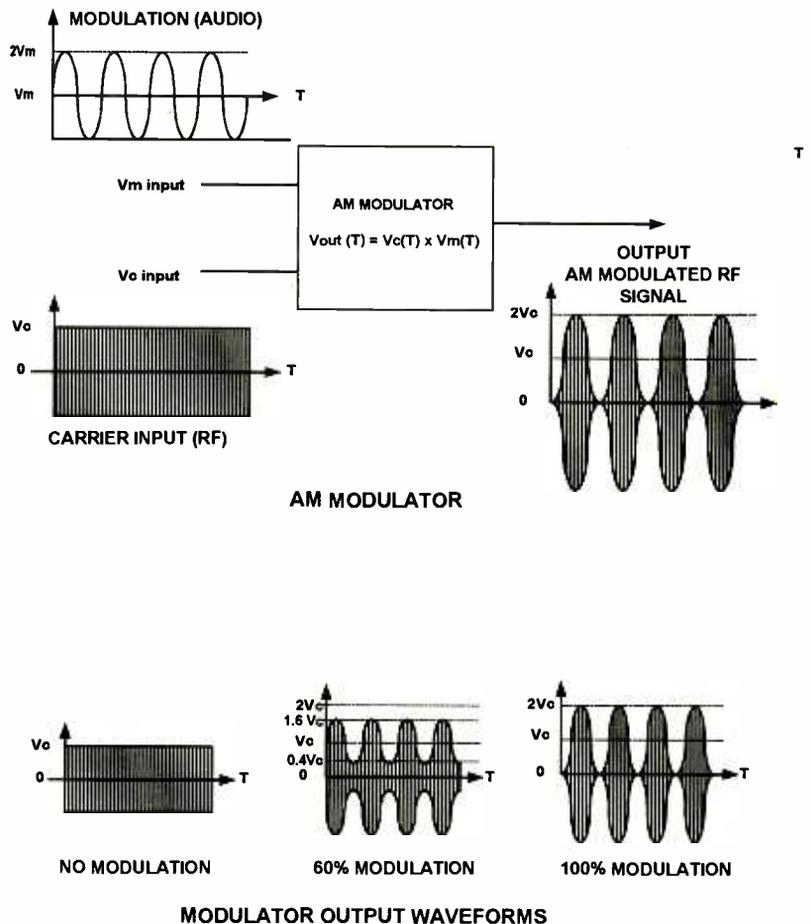


FIGURE 2 AMPLITUDE MODULATION (AM)

that produces an output proportional to the mathematical product of the input signals, the resultant output is an amplitude-modulated signal. We will assume this circuit has a gain of unity for simplicity. Then, multiplying the two signals we get an output signal as follows:

$$V_c(t) \times V_m(t) = A \sin \omega_c T + AM (\sin \omega_c T)(\sin \omega_m T) = \text{resultant signal}$$

A trigonometric identity from your high school trigonometry book says that the product of two sines of two angles is as follows:

$$\sin X \sin Y = \frac{1}{2} \cos (X-Y) + \frac{1}{2} \cos (X+Y)$$

For simplicity, assume  $A = M = 1$ . (This will result in a 1-volt carrier with a 1-volt peak modulating signal.) Substituting, in the trigonometric identity,  $X = \omega_c$  and  $Y = \omega_m$ ,  $A = B = 1$

$$V_c(t) \times V_m(t) = \sin \omega_c T + \frac{1}{2} \cos (\omega_c - \omega_m) T + \frac{1}{2} \cos (\omega_c + \omega_m) T$$

Now, we have three components in the resulting signal:

1)  $\sin \omega_c T$ , which is a unit level sinewave signal at the carrier frequency.

2)  $\frac{1}{2} \cos (\omega_c - \omega_m) T$ , which is a half-unit level cosinusoidal signal at a frequency equal to the difference between the carrier frequency and the modulating signal frequency. This is called the lower sideband.

3)  $\frac{1}{2} \cos (\omega_c + \omega_m) T$ , which is a half-unit level cosinusoidal signal at a frequency equal to the sum of the carrier frequency and the modulating signal frequency. This is called the upper sideband.

The three signals produced are the carrier, the lower sideband, and the upper sideband. A cosinusoidal waveform is just a sine waveform shifted in phase by 90 degrees, so at  $T = 0$  it is maximum, falling to zero at  $\omega T = 90$

(Continued on page 36)

## ALL ABOUT

(continued from page 20)

degrees. A sinewave starts at zero at  $T=0$  and has a maximum at 90 degrees. Note that the two sidebands are one half that of the carrier in amplitude and are different in frequency from the carrier by the modulating frequency.

There is also a 90-degree phase shift. The ratio of the modulating signal to its peak value at full modulation is called the modulation index and is denoted by the letter  $M$ .  $M$  has a value between zero (no modulation) and 1 (maximum modulation). If  $M$  exceeds 1, this is called overmodulation and results in distortion.

The important thing to see is that the total signal bandwidth needed to pass these three components is twice the modulation frequency. It does not depend on the value of  $M$ . Therefore, for a standard AM broadcast signal with a maximum modulating frequency of 5 kHz, a 10-kHz bandwidth is required in the receiver. Also note, since the carrier term is simply a constant amplitude sine wave, it carries no intelligence and its amplitude is constant. Now, comes the big kicker!

### Sidebands

Note that the amplitude of each sideband is only half of that of the carrier, even when  $M = 1$ . Therefore, the power in each sideband when  $M=1$  is only one quarter that of the carrier. Since there are two sidebands, there is a total sideband energy of only half that of the carrier. Since these sidebands are identical, differing only in frequency by twice the modulating frequency, they both carry the same information and are redundant from an information viewpoint. The

sidebands contain only one third the total signal power generated by the transmitter, but they carry all the information. Really, only one is needed, the other being redundant.

The modulating system must supply this sideband energy, half the power of the carrier signal if  $M = 1$ . The modulating power needed is equal to one half  $M$  squared. A 1000-watt AM carrier, for example, needs 500 watts of audio to fully modulate it. Well, then why not generate the AM signal at low level and amplify it? Not very efficient. Since the total peak amplitude of the signal is twice that of the carrier, a peak power of 4000 watts is present in a 1000-watt AM signal. Therefore, a power amplifier used for AM must be capable of delivering four times the carrier power on modulation peaks. The 4000-watt amplifier is delivering only a 1000-watt carrier and seldom operates at full power except on modulation peaks.

The overall efficiency is then low. Just as in real life, you do not get something for nothing. The alternative to a 500-watt modulator and a 1000-watt RF amplifier in this case is a low-level audio amplifier and a 4000-watt RF amplifier running inefficiently. Not that this is so bad, because at high-power levels it has the advantage of eliminating the expensive and heavy 500-watt modulation transformer needed to couple the audio energy to the transmitter-power amplifier. No matter how you look at it or do it, AM is a rip-off from an efficiency standpoint. However, it is simple to do, has fairly good audio fidelity, and still has better weak-signal performance over certain other modulation methods. It is easily received with a simple low-cost receiver and is not critical as to receiver

mistuning. AM is still used worldwide for short-, medium-, and long-wave broadcasting, and for air-to-ground VHF voice communications.

It was realized in the early days of radio that since only one of the sidebands is needed, why bother to transmit the carrier and the other sideband? The carrier doesn't "carry" anything, as both the sidebands are RF and can be radiated by an antenna. Getting rid of the carrier and one sideband gets rid of five sixths of the radiated power with no loss of information. So the transmitter power can be effectively increased by a factor of six, since all the energy can be placed in the transmitted sideband.

Furthermore, the receiver bandwidth can be reduced by a factor of two. This gives a total of 8-dB transmitter gain and 3-dB receiver sensitivity, or 11-dB improvement in signal-to-noise ratio. The likelihood of interference to or from other signals is also reduced by using half the bandwidth; and channel capacity of a frequency band can be doubled, since each signal needs only half the bandwidth of an AM signal. This modified form of AM modulation is called single sideband, or SSB. We will discuss this subject in the next part of this article. Tune in next month. P

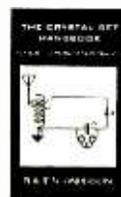
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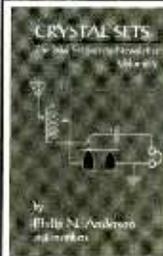
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## Modulation Methods, Part 2: SSB

Last month, we discussed CW and AM modulation and introduced the subject of Single SideBand (SSB). This month, we'll look at this type of modulation in depth.

SSB is a type of AM without the carrier and with only one sideband. DSB or Double SideBand is AM with the carrier suppressed, but with both upper and lower sidebands. DSB is compatible with SSB receivers, the receiver merely rejects the unwanted or redundant sideband. The use of both sidebands to carry two separate channels of information is called ISB, or Independent SideBand. ISB was somewhat popular with hams in the early 1960s as AM was gradually yielding to SSB, since a DSB transmitter was and is relatively simple to build.

DSB is seldom used today. However, it was a cheap way back then to gradually phase over to SSB; SSB receivers could handle it, and the unwelcome carrier signal was absent. We will not discuss DSB any further as it is considered obsolete as a voice-transmission method in HF communications work. It is still used in FM stereo transmission for the 38-kHz audio channel difference (L-R) subcarrier. This topic will be covered in a later column.

SSB was known in the early days of radio, but circuit techniques and hardware to generate it did not become readily available until after WW II. There was a transatlantic telephone circuit operating on about 55 kHz in the long-wave band during the 1920s, which used SSB transmission. Amateur radio operators (hams) who liked to experiment explored SSB after WW II, while AM was still "king."

### The Shift To SSB

However, the gradual shift to SSB started during the late 50s. During the early 60s, reasonably priced manufac-

tured SSB equipment became available to amateurs, and a gradual changeover to SSB took place. By 1970, AM was mainly used on the 28-MHz and VHF amateur bands, and it was called "Ancient Modulation." Even on the VHF bands, FM (Frequency Modulation) took over during the 70s, and, by 1980, AM was pretty scarce. AM activity can still be found near the 3.9- and 29-MHz frequencies in the 75-meter and 10-meter ham bands, with some local AM work also at 50.4 MHz in the 6-meter band.

AM has made a small comeback since the late 80s, since SSB equipment using LSI chips and microprocessors has become smaller, sophisticated, and too complex and forbidding for home experimentation. Old vacuum-tube AM equipment has enjoyed somewhat of a revival, as it lends itself to amateur experimenting. It's ideal for those interested in restoring and operating old-time vintage equipment.

The military has also long since converted to SSB for its HF communications work. With the exception of international broadcasting, HF voice communications is practically all SSB. International shortwave broadcasting is also going this route as well as toward digital radio. However, AM is still the simplest and cheapest from a reception standpoint, and it is almost universally used for broadcasting and air-to-ground VHF-UHF voice communications. Nevertheless, the use of SSB allows superior weak-signal reception and less transmitter power for the same results.

### VSB And Selective Fading

Most SSB exciters first generate a DSB signal, which is then processed into SSB. An AM signal with one sideband partially suppressed is called VSB or vestigial sideband. This signal is widely

used in television transmission to reduce bandwidth while still allowing AM detection schemes to be used. An SSB signal can be transmitted with a carrier to reduce occupied bandwidth, and this is called CSSB or compatible SSB. It has little advantage over AM other than the reduction in bandwidth and selective fading effects.

Selective fading is a phenomenon in radio transmission where the fading of a signal at the receiver is very frequency selective, usually due to radio-wave cancellation effects caused by phase differences from multipath transmission and ionospheric effects. It acts as a sharp notch filter, which continuously and randomly varies in center frequency. The filtering effect randomly nulls out one sideband, then the carrier, then the other sideband, and then might reverse direction. This randomly moving "notch filter" causes the fading and intermittent audio distortion heard on received AM signals. These disturbances can be easily heard on distant AM broadcast stations during the nighttime hours when multipath effects from sky-wave and groundwave signals cause them to occur. SSB is less susceptible to this as there is no carrier and no other sideband to deal with. Therefore, SSB transmission usually only exhibits rising and falling signal levels, with little extra distortion as compared to AM.

### SSB Generation

Refer to Fig. 1 for the following discussion of how SSB is generated. Audio information at the transmitter input is first fed into an amplifier and possibly a speech compressor or clipper. This step serves to increase average modulation level. **Note:** A word of caution here. Unlike an AM signal, in which the envelope has the same waveform as the modulating waveform, the envelope wave-

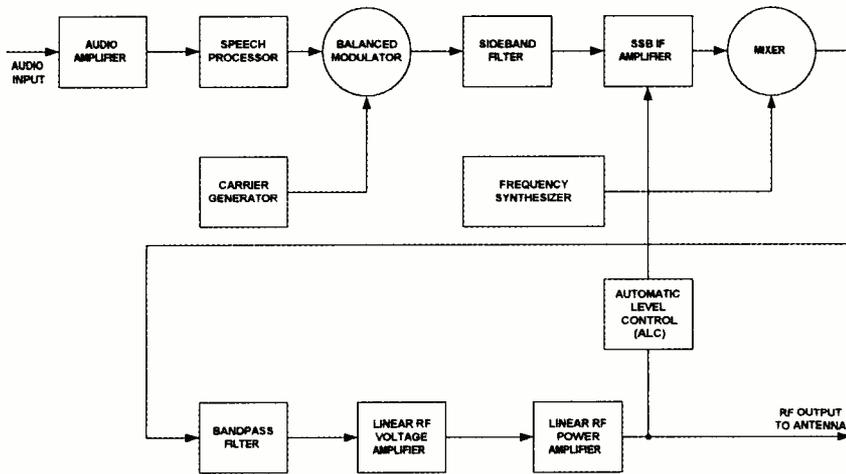


FIG. 1. BLOCK DIAGRAM SSB TRANSMITTER USING FILTER METHOD

form of an SSB signal has no direct simple relationship to the modulating signal (see Fig. 2). Using clipping of peaks can introduce undesirable effects and actually degrade the signal. Compression, on the other hand, largely preserves the waveshape of the modulating signal, mainly affecting its amplitude, and can be effective in boosting the average modulation level. The lesson here is to avoid the all too commonly heard over-clipped and overcompressed signals that are strong but nearly unreadable. The idea that "if enough is enough, then more is better, and too much is just right" does not apply in this case.

Next, the audio should be bandlimited to eliminate products outside the intended bandwidth. Typically, this bandwidth will be 200 to 3500 Hz for speech, although 2500 Hz is sometimes used as an upper limit. The audio is then fed to a balanced modulator that is also driven with an RF carrier at the SSB-generation frequency, sometimes called the transmitting IF frequency.

In many instances, this transmitting frequency is the same as the receiver IF frequency, which is often done in transceiver systems—using the same circuitry for modulation and demodulation. The output of the balanced modulator (actually a mixer) is a double-sideband suppressed carrier signal, since the carrier is cancelled out. In the absence of a modulating signal, the output is ideally zero. In practical balanced modulators, about 30- to 40-dB suppression of the carrier is obtained. There is usually some provision for optimizing carrier suppression in most circuits, although with modern solid-state diode, doubly balanced mixer assemblies inherent suppression is good

enough and no adjustment is necessary. Next, the output of the mixer or modulator is fed to a sharp cutoff filter. This filter may be made up of L-C elements (in the 10-50-kHz range), or mechanical resonators (455 or 500 kHz), or, most often, made from quartz crystals. Crystal

filters are available at many popular frequencies as off-the-shelf assemblies, such as 1.65, 3.0, 5, 9, 10.7, and 21.4 MHz—common SSB IF frequencies that are stock crystal filters. Many other frequencies are also used. The 5- to 9-MHz range seems most popular, as crystal filters for this range are easily made. The filter should have a bandwidth (for speech) of about 2.1 to 3 kHz, should have a center frequency about 1.5 kHz above or below the carrier frequency, and should have a 20- to 30-dB rejection at the carrier frequency. The filter should cut off sharply on the carrier side and should have 40 dB or better rejection of the unwanted sideband. Crystals are used in these filters because we need the very high Q values to physically realize this kind of rejection and bandwidth. The filter is generally one of the more expensive components in an SSB system.

### Creating A Signal

An SSB generator of this type can

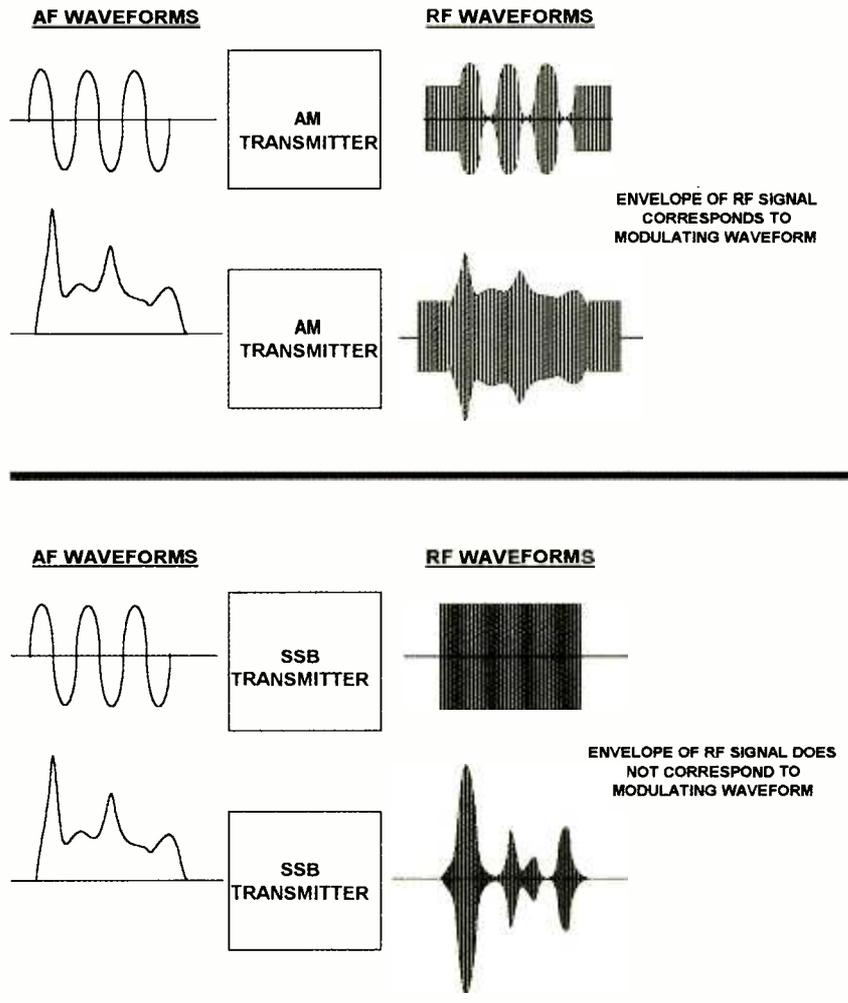


FIG. 2. AM and SSB TRANSMITTER WAVEFORMS

generate an SSB signal of either lower or upper sideband. This option is a function of the filter-response characteristics. If capability to generate a signal of either sideband is needed, there are several approaches.

First, two separate filters can be used with a switching arrangement to select the desired sideband. Alternately, a filter with a symmetrical response curve that has a very sharp cutoff on each side can be used, and the carrier oscillator can be shifted to either side of the filter.

A scheme that was popular some years ago used a filter at 9.000 MHz that had a symmetrical response plus and minus 1.2 kHz from each side of center. This filter gave a bandwidth of 2.4 kHz for the signal, and two separate crystals were provided in the carrier oscillator—one at 8998.5 for USB (*Upper SideBand*) generation, and another at 9001.5 for *Lower SideBand* (LSB) generation. This scheme had the disadvantage of having a 1.5-kHz nominal error in the 9.000-MHz nominal frequency, but it was corrected by shifting the LO (*Local Oscillator*) signal plus or minus 1.5 kHz to compensate, making the final output frequency correct. Today transmitters control the frequency synthesizer via software in the microcontroller programming, so the entire operation is transparent and automatic.

Another method using only one filter involves using a mixer. One old scheme was to generate the SSB signal at 455 kHz using a mechanical filter as the SSB filter, as follows: The output from the SSB generator is 455 kHz USB. Next, the 455-kHz signal is mixed with the fourth harmonic of the carrier, 4 X 455 kHz or 1820 kHz. That gives a USB signal of 2275 kHz, which is the IF frequency in this system. If an LSB signal is desired, the sixth harmonic of 455 kHz at 2730 kHz is used. The 455-kHz USB signal when mixed with 2730 kHz results in a 2275-kHz SSB signal as before, but now this is an LSB signal. Here we are taking the difference rather than the sum.

In sum mixing, the output is the sum of the IF signal and the LO signal. If the IF increases in frequency, so does the sum of the two signals. In difference mixing, when the IF signal increases in frequency, the resulting sum of the IF and LO will decrease in frequency. This relationship results in inversion of the SSB signal about the carrier frequency (in this case 2275 kHz). This system has

the disadvantage of needing a mixer and extra stages to generate the X4 and X6 signals to mix with the generated SSB signal and corresponding switching arrangements. The extra cost and complexity must be weighed against the cost of an extra filter.

In amateur radio HF transceivers, a commonly used technique is the use of one symmetrical filter with corresponding offsetting the LO (as mentioned before). In this case, the software in the synthesizer costs nothing once written and debugged and takes no physical room. The sharp symmetrical filter is cheaper than two separate filters, as well, and transceiver design is simplified as the same conditions apply to both receive and transmit.

### Filter Output

The output of the filter is an SSB signal at the IF frequency. This signal is then mixed with a very stable and pure local oscillator signal from a very stable VFO (*Variable-Frequency Oscillator*) or frequency synthesizer. This is done in a high-level, very linear mixer to produce the desired SSB output frequency. A filter system then removes unwanted mixer products; and the resulting SSB signal is then amplified to the final transmitter power output level, which may be a few watts to many kilowatts. A very linear amplifier must be used to prevent the generation of intermodulation-distortion products that will appear as unwanted components and interference on the transmitted signal.

Linear amplifiers may be vacuum-tube or solid-state. For very high power levels (about 500 watts or more), vacuum-tube technology is still the technology of choice. Most transmitters and transceivers in the 100-watt class use solid-state, bi-polar, or power FET devices. Higher power solid-state amplifiers above a few hundred watts generally need large and heavy heatsinks, RF power combiners, and several large expensive transistors, together with a high-current, low-voltage supply. It is difficult to get these large quantities of heat out of relatively small chip areas while keeping chip temperatures reasonable.

Often, sophisticated protection circuitry is needed to keep the transistors safe against load faults and power spikes. Vacuum tubes do not have this problem, and only a simple cooling fan is required in most cases. There are a few solid state

500-1000 watt amplifiers sold by SSB radio manufacturers. However, vacuum-tube amplifiers are usually smaller; can be just as, or more, efficient than solid-state; and are more reliable, with much better immunity to load faults such as high VSWR (*Voltage Standing-Wave Ratio*) due to broken, mismatched or shorted antennas. A vacuum tube can usually stand a severe fault for a few seconds, while transistors can fail in microseconds. For this reason, high-power applications are often better implemented with vacuum tubes.

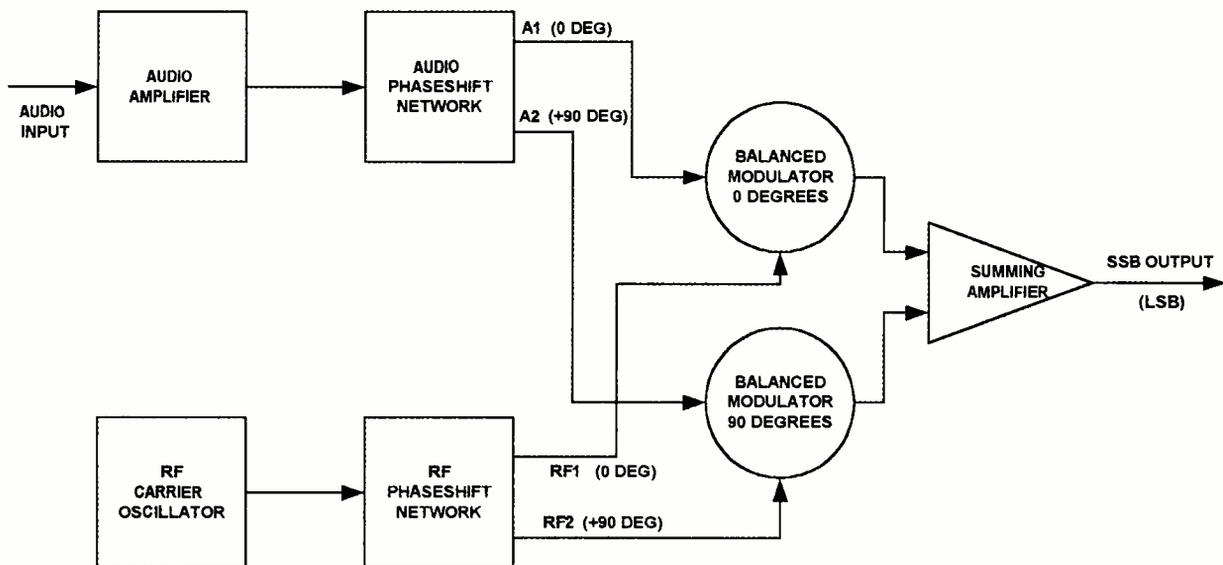
Large, expensive, and heavy 60-Hz transformer-type high-voltage supplies from the old days can now be replaced with much smaller and lighter highly efficient switching-type solid-state supplies, but tubes still are better suited for the RF circuitry. The vacuum tube still is king here, and may always be, for high power levels. However, for low power (200 watts or less) and portable transmitter use, solid-state is undeniably the best approach.

### The Phasing Method

Another approach to SSB generation is called the phasing method. In this approach, a clever phase-cancellation technique is used. This method eliminates the need for a sharp SSB filter and is potentially lower in cost. (See Fig. 3.) First, the audio signal after processing and bandlimiting (very important in this approach) is split into two components, equal in amplitude but exactly 90 degrees apart in phase. This splitting is the difficult part, as a network is needed that provides a 90-degree phaseshift within plus or minus 1 or 1.5 degrees over the entire audio range of 300 to 3000 Hz.

There are classes of R-C networks that have this property, generally involving precision components. In practice, each audio component is fed to a separate network. While the individual network phaseshifts vary over the audio frequency range, the difference between their outputs stays within a degree of 90 degrees, with constant amplitude. The synthesis of these networks is beyond the scope of this article. A typical network is shown in Fig. 3. It is rather simple, but requires precision components. The degree of unwanted sideband suppression depends on it.

Next, the two 90-degrees-apart audio channels are fed to identical balanced modulators or doubly balanced mixers



IF:  $A1 = \sin WmT$   
 $A2 = \cos WmT$   
 and  
 $RF1 = \sin WcT$   
 $RF2 = \cos WcT$

$$\sin A \sin B = 1/2 \cos(A-B) - 1/2 \cos(A+B)$$

$$\cos A \cos B = 1/2 \cos(A+B) - 1/2 \cos(A-B)$$

THEN:

$$\text{SSB OUTPUT} = 1/2 \cos(WcT - WmT) - 1/2 \cos(WcT + WmT) + 1/2 \cos(WcT + WmT) + 1/2 \cos(WcT - WmT)$$

SUMMING TERMS AND SUBSTITUTING:

$$\text{SSB OUTPUT} = 1/2 (\text{LSB}) - 1/2 (\text{USB}) + 1/2 (\text{USB}) + 1/2 (\text{LSB}) = \text{LSB ONLY}$$

FIG. 3. PHASING METHOD OF SSB GENERATION

that are driven with two carriers identical in frequency but also exactly 90 degrees apart in phase. This step is easily done since the carrier frequency is generally fixed. A network consisting of R-C or L-C circuits can provide this 90-degree phaseshift, or a divide-by-two frequency divider can be used. Two JK flip-flops driven by two identical clock-signal square waves 180 degrees apart will produce two outputs at half the input frequency and 90 degrees apart in phase.

The outputs of the two mixers are then combined. It can be shown that the output will consist of only one sideband (see Fig. 3), since the double sideband signals from each mixer will have phase relationships such that one of the sideband components will have opposite phase with respect to the other and the other will be in phase. Sideband selection occurs simply by reversing the phase of either one audio or one carrier channel. In practice, the audio channel method is normally used.

While a good method, the phasing method requires accurate component matching, narrow tolerances, and accurate setup. Nevertheless, it has been successful in amateur radio equipment, mainly when separate transmitters and receivers were used in the past. Today, transceivers are the main components, and the phasing method is not found as a filter is still needed any way for the receiver section. In the future, digital-signal-processing will undoubtedly be the common method, eliminating or simplifying the filter required. While other methods exist, most SSB generation will be done for a while using the filter method. SSB crystal filters have come down somewhat in price due to manufacturing and design improvements, as well as increasing market demand, keeping the filter method as the most popular.

### Signal Reception

Reception of SSB signals generally follows the reverse of the generation

process. A look at the spectrum of a voice SSB signal will show that it is simply the input audio input spectrum shifted up into the RF region. For example, consider a 10,000-MHz voice-frequency SSB signal. If the USB mode is used, the transmitter will produce a signal having frequency components of 10.0003 to 10.0030 MHz, or simply 300 to 3000 Hz (0.3 to 3 kHz), shifted arithmetically higher in frequency by 10 MHz.

To receive this signal, we must simply shift it back down to the audio region. For LSB, the transmitted spectrum is also inverted, the higher voice frequency components producing lower transmitter frequency components. A simple mixer (in this application commonly called a product detector, same mixer circuit, different name) can be used for this function, and, indeed, a receiver can be built in which an antenna is connected to a mixer that is fed with an LO. If the LO is exactly the same frequency as the suppressed carrier of the input SSB signal from the antenna, the product

detector output will be the original audio that modulated the SSB transmitter. This type of detector when used with an antenna and a suitable audio amplifier will make up a receiver commonly called a direct conversion receiver.

Useful for SSB and CW (Morse code) reception, this scheme is popular for low-cost, ham-radio receiver construction and eliminates much RF circuitry. The LO must be stable and have good noise characteristics; and a low-noise audio amplifier is necessary, but sensitivities around a microvolt can be obtained. The bandwidth is that of the audio amplifier. Disadvantages of this receiver are lack of sideband selection; poor RF selectivity; lack of AM reception capability due to LO beating with the AM carrier; and susceptibility to RF overload, as generally no AGC (Automatic Gain Control) is used. However this receiver provides a lot of performance with very little circuitry and is superior to and easier to use than a regenerative receiver for SSB and CW reception.

### Frequency

The carrier must be reinserted at the detector within a few hertz of the original carrier. Otherwise, the frequency of the audio output will be shifted from the original by an amount equal to this difference. For speech, 50 Hz is acceptable, but for quality 10 Hz is desirable; for music or where frequencies are critical, 1 Hz would be better.

To assist in this process, a pilot carrier may be transmitted. This carrier is a residual sample of the original carrier sent at a known level, i.e. -30 or -40 dB down so that it is not very noticeable. A phase-locked-loop at the receiver locks

on to this pilot carrier, ensuring accurate tuning. Modern SSB equipment used by amateur radio operators can easily hold frequency within 10 Hz, so this is not often done.

If the reinserted carrier is way off, the SSB signal will sound like gibberish—often called the "Donald Duck" sound. If the carrier is way off (a few kHz) and is placed on the opposite side of the signal, the recovered audio may actually be spectrally inverted. Now, the original low-speech frequencies (300-400 Hz) are at the high end of the audio band (near 3000 Hz), and the original audio components at the 3000-Hz end of the audio spectrum are now shifted down to the 300-Hz region. This effect is called "inverted speech," and this concept is used elsewhere to scramble an audio signal for privacy or security purposes. In practice, this scrambling is done with special circuitry. An article by the authors of this column appeared on page 37 of the December 1993 issue of **Electronics Now**, exploring using digital techniques to do such scrambling.

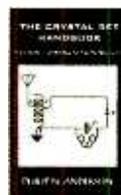
Other than the requirements for an accurate and stable LO frequency and a product detector, an SSB receiver is generally a standard super-heterodyne receiver. It has high RF performance in areas of dynamic range, noise floor, and stability, as well as special AGC circuitry, since there is no such carrier for AGC reference, as exists in an AM receiver. An SSB receiver usually has a separate envelope or synchronous detector for AM reception anyway and has switchable AGC for each reception mode.

In a transceiver system, often the same circuitry as above, SSB generation and detection runs "backwards" from

the receiver system. This type is called bilateral circuitry and will not be discussed here. Interested readers can refer to books such as the *ARRL Radio Amateurs Handbook* or the *RSGB Handbook* for details.

This discussion of SSB techniques has necessarily been brief. Entire books have been written on SSB, but it is impossible to cover the whole topic in a short article. The next part of this discussion will discuss frequency modulation methods and techniques. **P**

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