

Objectives

- To become familiar with:
 - The different modulation modes used by Amateur Radio operators; and
 - The functional diagrams of several types of transmitters.

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Modulation

- A pure radio signal (a "**carrier**") does not convey any information by itself. It **must be changed** in some manner such that a listener can reverse the modification process and recover the information.
- The **process of imparting information** onto the carrier is called **Modulation**.
- Modulation is achieved by changing the presence, frequency, amplitude or phase of the carrier wave.
- The process of recovering that information is called
 Demodulation or Detection.
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In electronics and telecommunications, **modulation** is the process of varying one or more properties of a periodic waveform, called the *carrier signal*, with a modulating signal that typically contains information to be transmitted. Most radio systems in the 20th century used frequency modulation (FM) or amplitude modulation (AM) for radio broadcast.

A **modulator** is a device that performs modulation.

A **demodulator** (sometimes *detector* or *demod*) is a device that performs demodulation, the inverse of modulation.

A modem (from **mo**dulator–**dem**odulator) can perform both operations.

Telegraphy

- Simplest method of modulating a signal.
- A key is used to turn the carrier on and off in accordance with the Morse Code.
- **Dits** are one time unit in length, **Dahs** are 3 time units in length.
- Commonly known as **Continuous Wave (CW).**

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A continuous wave or continuous waveform (CW) is

an electromagnetic wave of constant amplitude and frequency, typically a sine wave, that for mathematical analysis is considered to be of infinite duration. Continuous wave is also the name given to an early method of radio transmission, in which a sinusoidal carrier wave is switched on and off. Information is carried in the varying duration of the on and off periods of the signal, for example by Morse code in early radio. In early wireless telegraphy radio transmission, CW waves were also known as "undamped waves", to distinguish this method from damped wave signals produced by earlier *spark gap* type transmitters.

Early radio transmitters could not be modulated to transmit speech, and so CW radio telegraphy was the only form of communication available. CW still remains a viable form of radio communication many years after voice transmission was perfected, because simple, robust transmitters can be used, and because its signals are the simplest of the forms of modulation able to penetrate interference. The low bandwidth of the code signal, due in part to low information transmission rate, allows very selective filters to be used in the receiver, which block out much of the radio noise that would otherwise reduce the intelligibility of the signal.

Continuous-wave radio was called radiotelegraphy because like

the telegraph, it worked by means of a simple switch to transmit Morse code. However, instead of controlling the electricity in a cross-country wire, the switch controlled the power sent to a radio transmitter. This mode is still in common use by amateur radio operators.

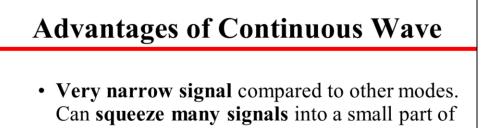
In military communications and amateur radio the terms "CW" and "Morse code" are often used interchangeably, despite the distinctions between the two. Aside from radio signals, Morse code may be sent using direct current in wires, sound, or light, for example. For radio signals, a carrier wave is keyed on and off to represent the dots and dashes of the code elements. The carrier's amplitude and frequency remains constant during each code element. At the receiver, the received signal is mixed with a heterodyne signal from a BFO (beat frequency oscillator) to change the radio frequency impulses to sound. Almost all commercial traffic has now ceased operation using Morse, but it is still used by amateur radio operators. Non-directional beacons (NDB) and VHF omnidirectional radio range (VOR) used in air navigation use Morse to transmit their identifier.

Key clicks

In order to transmit information, the continuous wave must be turned off and on with a telegraph key to produce the different length pulses, "dots" and "dashes", that spell out text messages in Morse code, so a "continuous wave" radiotelegraphy signal consists of pulses of sine waves with a constant amplitude interspersed with gaps of no signal.

In on-off carrier keying, if the carrier wave is turned on or off abruptly, communications theory can show that the bandwidth will be large; if the carrier turns on and off more gradually, the bandwidth will be smaller. The bandwidth of an on-off keyed signal is related to the data transmission rate as: where is the necessary bandwidth in hertz, is the keying rate in signal changes per second (baud rate), and is a constant related to the expected radio propagation conditions; K=1 is difficult for a human ear to decode, K=3 or K=5 is used when fading or multipath propagation is expected.

The spurious noise emitted by a transmitter which abruptly switches a carrier on and off is called *key clicks*. The noise occurs in the part of the signal bandwidth further above and below the carrier than required for normal, less abrupt switching. The solution to the problem for CW is to make the transition between on and off to be more gradual, making the edges of pulses *soft*, appearing more rounded, or to use other modulation methods (e.g. phase modulation). Certain types of power amplifiers used in transmission may aggravate the effect of key clicks.



- the spectrum.
 No "accent" to hinder communications. CW is actually its own language with many abbreviations and "Q" signals.
- CW can "**punch through**" under **difficult conditions** where other modes might have difficulty.

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Advantages of the Morse code

The are several advantages to Morse code or CW even in these days of advanced technology, digital transmissions and the like:

•*Simplicity:* It simplicity gives it a number of advantages. The first has already been mentioned and is its simplicity. The overall equipment required for two way radio communications is relatively straightforward and this makes it ideal for ham radio constructors. Simple ham radio transmitters consisting of just a few electronic components: transistors, resistors, capacitors and possibly a quartz crystal can be built and used to make contacts with ham radio stations all around the globe.

•**Bandwidth:** The rate at which signalling is performed is relatively low and this means that it occupies only a small bandwidth. This gives two advantages. The first is that a large number of stations can occupy a small section of the band, and secondly narrow filters can be used to reduce the level of background noise and interference. Coupled to this the human brain can read Morse signals when they are lower than the surrounding noise level. As a result it is possible to copy a Morse signal at a lower strength than any other form of transmission. •*International appeal:* The use a large number of abbreviations and the formalised formats for ham radio contacts means that Morse or CW can be used by people from around the globe even with a poor command of languages like English that are widespread. Using the standard abbreviations basic contacts can be conducted with a limited knowledge of the language of the other person. This may not be possible using other modes of transmission.

How to Send CW

- Old style key (called a "straight key") is still used by many.
- Many use electronic keyers, which automatically make the Dits and Dahs.
- Some people use **keyboards and computers** to key the transmitter. These methods can also "read" the CW and **display it on the screen.**
- **Mechanical 'bugs**" can make the Dits, but the Dahs are still made manually.
- Leave between 150 to 500 Hz separation between your frequency and a contact in progress. Al Penney VO1NO

How is the Morse code used

The way in which Morse code is transmitted is quite straightforward. For radio applications such as ham radio / amateur radio, all that is needed is a radio frequency signal that can be turned on and off. In view of the fact that generating the signal is relatively straightforward, this means that the transmitters can be made more easily than for some of the other types of transmission that are heard on the short wave bands. This makes Morse an ideal medium for use in ham radio or amateur radio for those people who like constructing their own equipment.

To receive the signal is a little more complicated. If the signal was received on an ordinary domestic radio, all that should be heard are clicks and plops as the signal turns on and off. To generate the characteristic Morse tone, the radio receiver must be equipped with a beat frequency oscillator, BFO, or carrier insertion oscillator, CIO. This generates a signal within the receiver that beats with the incoming signal to generate an audio tone that is associated with a Morse signal, and can also be easily read.

Most radio receivers used in ham radio will have a BFO which can be used for receiving Morse code signals. Today most HF band radios will have a mode switch position specifically for Morse / CW. If not specifically marked for Morse or CW, a position marked SSB is equally effective. Some older radios may have a BFO or CIO which needs to be turned on separately.



Most Morse keys and keyers mainly fall into in one of three categories:

•**Straight Morse key:** The straight Morse key is the traditional form of Morse key that has a lever and uses an up and down motion to make and break the contact and hence make the dots and dashes. It was also the first type of Morse key to be used.

•*Mechanical semi-automatic keyer:* The semi-automatic Morse key was developed to overcome an injury known as telegraph hand, but today we would know as repetitive strain injury, RSI. The key had a paddles and when moved to the left made contact for the dashes, and when moved to the right, it set a vibrating arm in motion to create the dots. Although it took a little time to learn, it reduced the instances of telegraph hand and enabled much faster operation. One of the first keys of this type to be made was the Vibroplex, and these can still be obtained to this day.

•**Electronic keyer:** The electronic keyer is an electronic development of the mechanical key. The basic version has a paddle which creates a series of dashes when moved to the left, and dots when moved to the right. More advanced keyers have two paddles next to each other to provide a "squeeze" function where alternate dots and dashes are produced. These were first known as squeeze keyers but today they are generally known as iambic keyers.

•Although full electronic keyers are available, most modern ham radio transceivers have the keyer electronics contained within the transceiver and only the paddle is required. This simplifies their installation and enables fast two way radio communications using Morse code.

These are the main types of Morse key that are in use. Computer technology is also widely used, ad using the right software, messages can be types in on a keyboard and the resulting Morse code is generated. Some software can also read Morse code, but often this is not as good as the human ear at decoding the message int he presence of interference.

Amplitude Modulation

- Changes the **instantaneous power** in the radio wave **in time with a modulating signal**.
- The strength (amplitude) of the carrier signal is made to vary in accordance with the audio signal.

Amplitude modulation was the first form of modulation used to carry sound and in particular voice communication over radio.

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Although it is nowhere near as efficient as other forms of modulation, it can very occasionally be heard on the amateur radio bands today.

In amateur radio circles the abbreviation, AM, is sometimes jokingly used to stand for Ancient Mode.

Amplitude modulation technology

The concept behind amplitude modulation is very straightforward, the audio or other modulating signal is used to vary the amplitude of the carrier or RF signal.

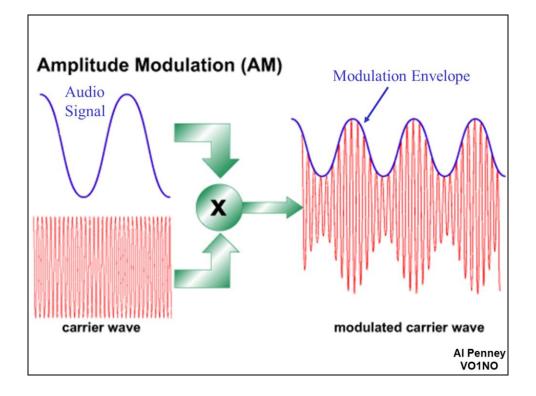
The main drawback of amplitude modulation is the very low efficiency. The carrier is there purely as a reference for demodulation and does not contribute to the transfer of information, resulting in a large wastage of power. Also the fact that the two sidebands are mirror images of each other means that overall signal uses twice as much spectrum as is really needed.

Decline in use of AM for amateur radio

The use of AM within amateur radio was relatively widespread in the early to mid 1960s on the HF bands as single sideband transceivers were still something of a novelty. As SSB became the mode of preference for ham radio operation on the HF bands, the use of AM declined in the late 1960s and early 1970s. It was sometimes used for local contacts on 160 metres and sometimes 80 metres.

One of the issues that was that when the bands were busy with amateur radio stations being received from many areas of the globe, the carriers would cause heterodynes that could make copy very difficult. With single sideband today, interference is still an issue, but the heterodynes caused by the AM carriers could give rise to much higher levels of interference, especially as notch filters were not common on receivers.

On the amateur radio VHF bands, AM was the operating mode of choice in the early and mid 1960s. As frequency modulation, FM because more popular as a result of its superior performance for mobile and portable applications, the use of AM declined. Now it is hardly ever heard on the amateur radio VHF and UHF bands.



In order that a radio signal can carry audio or other information for broadcasting or for two way radio communication, it must be modulated or changed in some way. Although there are a number of ways in which a radio signal may be modulated, one of the easiest is to change its amplitude in line with variations of the sound.

In this way the amplitude of the radio frequency signal varies in line with the instantaneous value of the intensity of the modulation. This means that the radio frequency signal has a representation of the sound wave superimposed in it.

In view of the way the basic signal "carries" the sound or modulation, the radio frequency signal is often termed the "carrier".

From the diagram, it can be seen that the envelope of the signal follows the contours of the modulating signal.

Advantages & disadvantages of amplitude modulation, AM

As with any technology there are advantages and disadvantages to be considered. The summary below gives a highlight of the basic pro's and con's.

Advantages

•It is simple to implement

•it can be demodulated using a circuit consisting of very few components

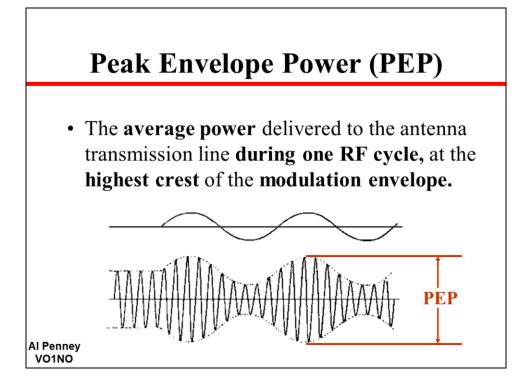
•AM receivers are very cheap as no specialised components are needed.

Disadvantages

•It is not efficient in terms of its power usage

•It is not efficient in terms of its use of bandwidth, requiring a bandwidth equal to twice that of the highest audio frequency

•It is prone to high levels of noise because most noise is amplitude based and obviously AM detectors are sensitive to it.



Peak envelope power (PEP) is the highest envelope power supplied to the antenna transmission line by a transmitter during any full undistorted RF cycle or series of complete <u>radio frequency</u> cycles. PEP is normally considered the occasional or continuously repeating crest of the modulation envelope under normal operating conditions. The United States Federal Communications Commission uses PEP to set maximum power standards for amateur radio transmitters.

AM PEP

Assuming linear, perfectly symmetrical, 100% modulation of a carrier, PEP output of an AM transmitter is four times its carrier PEP; in other words, a typical modern 100-watt amateur transceiver is usually rated for no more than, and often less than, 25 watts carrier output when operating in AM.

PEP vs. Average Power

PEP is equal to steady carrier power, or radiotelegraph dot or dash average power, in a properly-formed CW transmission. PEP is also equal to average power in a steady FM, FSK, or RTTY transmission.

Although average power is the same as PEP for complex modulation forms, such as FSK, the peak envelope power bears no particular ratio or mathematical relationship to longer-term average power in distorted envelopes, such as a CW waveform with power overshoot, or with amplitude modulated waveforms, such as SSB or AM voice transmissions. Typical average power of a SSB voice transmission, for example, is 10-20% of PEP. The percentage of longer term average power to PEP increases with processing, and commonly reaches ~50% with extreme speech processing.

AM Signal Quality

- **Overmodulation** can **seriously distort** an AM signal.
- If overmodulated, the **negative signal peaks** will **cut off the carrier**, resulting in **serious distortion** and the appearance of **spurious frequencies**.
- Ensure Microphone Gain (Mic Gain) and Automatic Level Control (ALC) are set as directed by the manufacturer.
- More is not necessarily better!

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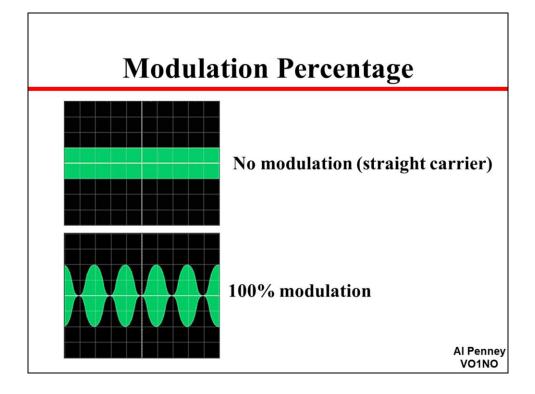
Overmodulation is the condition that prevails

in telecommunication when the instantaneous level of the modulating signal exceeds the value necessary to produce 100% <u>modulation</u> of the carrier. In the sense of this definition, it is almost always considered a fault condition. In layman's terms, the signal is going "off the scale". Overmodulation results in spurious emissions by the modulated carrier, and distortion of the recovered modulating signal. This means that the envelope of the output waveform is distorted.

Although overmodulation is sometimes considered permissible, it should not occur in practice; a distorted waveform envelope will result in a distorted output signal of the receiving medium

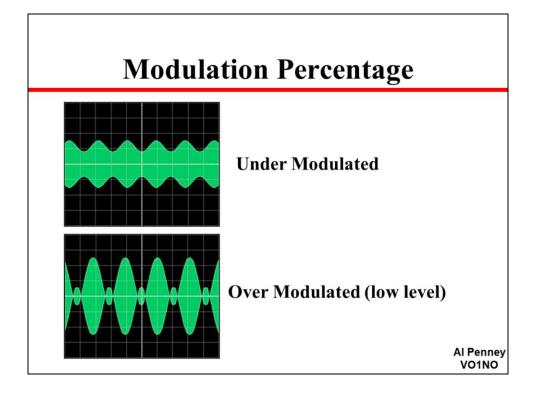
It is possible to vary the level of modulation applied to an amplitude modulated signal. This is an important factor for broadcast and two way radio communications applications.

If little modulation is applied then the audio (assuming it us an audio transmission) will be difficult to hear. However if too much is applied, distortion can result and signals will not be easy to listen to and interference will increase and this could affect users on nearby frequencies or channels. As a result of this it is necessary to have a way of defining the level of modulation applied to an amplitude modulated signal, and monitoring the level.Two figures are used for this, namely the amplitude modulation, AM modulation index, and the modulation depth. Both are related, but they have slightly different meanings.



It is helpful to see some examples of amplitude modulated waveforms with different levels of modulation index.

The most widely seen modulation level is for a signal that has 100% modulation. Under these circumstances the signal level falls to zero and rises to twice the value with no modulation. In this case the voltage rises to a maximum of twice the normal level – this means that the power will be four times that of the quiescent value, i.e. 2^2 the value of the no modulation level.



If less than 100% modulation is applied, then the carrier will not fall to zero, no will it rise to twice the level, but the deviation will be less than this from the quiescent level. The diagram above shows a level of 50% modulation, but the principle holds good for any value between 0 and 100% modulation.

Levels greater than 100% modulation

If the level of modulation is raised up above a modulation index of 1, i.e. more than 100% modulation this causes what is termed overmodulation.

The carrier experiences 180° phase reversals where the carrier level would try to go below the zero point. These phase reversals give rise to additional sidebands resulting from the phase reversals (phase modulation). These sideband caused by the phase reversal extend out, in theory to infinity. This can cause serious interference to other users if not filtered.

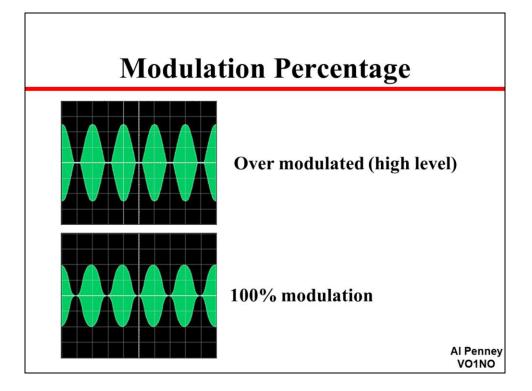
Broadcast stations using amplitude modulation take measures to ensure that the carriers or signals for their transmissions never become over modulated. The transmitters incorporate limiters to prevent more than 100% modulation. They also normally incorporate automatic audio gain controls to keep the audio levels such that near 100% modulation levels are achieved for most of the time. In this way the signal sounds clearer and stronger when demodulated. The audio processor may also clip the audio if it becomes very close to the 100% modulation level. This will ensure that the carrier is not over-modulated.

High and low level AM modulators

AM modulators may be classed as either high or low level dependent upon their level in the overall signal chain.

•*High level modulator:* A high level modulator is defined as one that modulates a high power section of the circuit, typically the final RF amplifier. It has the advantage that linear amplifiers are not required for the RF amplification stages after AM modulation has been applied. The drawback is that high power audio amplifiers are needed. For broadcast transmitters where very high power levels are used, class D or class E amplifiers may be employed for the audio output.

•Low level modulator: A low level AM modulator would be one where the modulation is applied to low power stage of the transmitter, typically in the RF generation stages, or via the digital signal processing areas. The drawback of this approach is that linear amplification is required for the RF stages.



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Mixing Signals Together

- When **two signals** are **mixed together**, the result is **four signals**:
 - The original two signals;
 - The sum of the two original signals; and
 - The **difference** between the **two original signals**.
- Example: A carrier signal at 3.8 MHz is modulated by a 1000 Hz tone (0.001 MHz). What are the resulting signals?

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In electronics, a **mixer**, or **frequency mixer**, is a nonlinear electrical circuit that creates new frequencies from two signals applied to it. In its most common application, two signals are applied to a mixer, and it produces new signals at the sum and difference of the original frequencies. Other frequency components may also be produced in a practical frequency mixer.

Mixers are widely used to shift signals from one frequency range to another, a process known as heterodyning, for convenience in transmission or further signal processing. For example, a key component of a superheterodyne receiver is a mixer used to move received signals to a common intermediate frequency. Frequency mixers are also used to modulate a carrier signal in radio transmitters.

One of the most useful RF or radio frequency processes is that of mixing. Unlike an audio mixer where signals are simply added together, when a radio or RF engineer talks about mixing, he means a whole different process. Here signals are multiplied together and signals an new frequencies are generated.

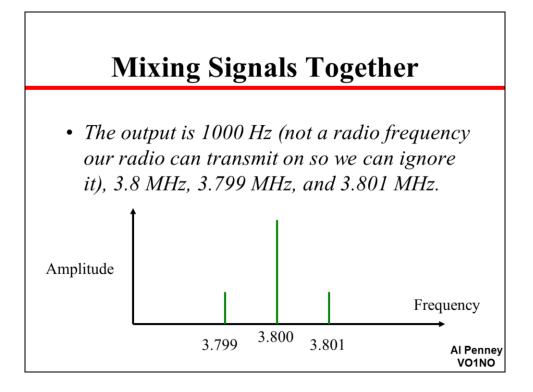
The process of RF or non-linear mixing or multiplication is used in virtually every radio set these days and also in many other circuits

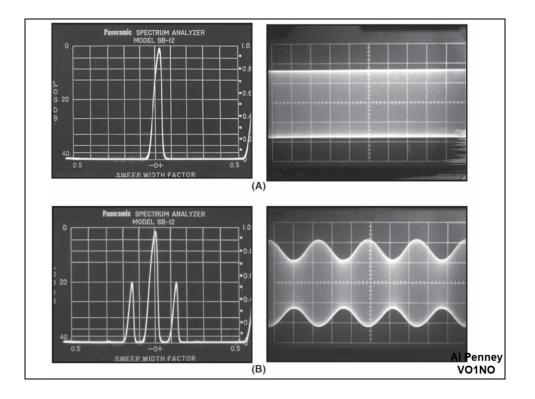
beside. It enables signals to be changed from one frequency to another so that signal processing for example can be undertaken on a low frequency where it is easier to perform, but the signal can be changed to a from a higher frequency where the signal is to be transmitted or received.

What happens when signals are mixed

It is found that if two signals are passed through a non-linear circuit, then additional signals on new frequencies are formed. These appear at frequencies equal to the sum and difference frequencies of the original signals. In other words if signals at frequencies of f1 and f2 enter the mixer, then additional signals at frequencies of (f1+f2) and (f1-f2) will also be seen at the output.

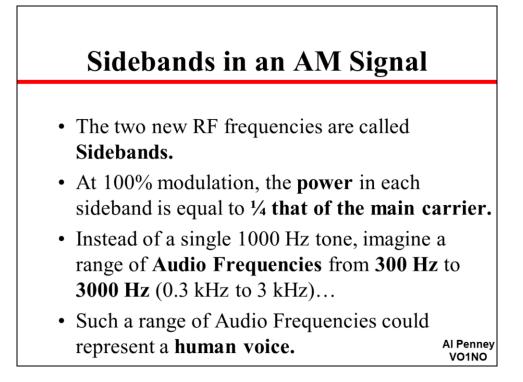
To give an example if the two original signals are at frequencies of 1 MHz and 0.75 MHz, then the two resultant signals will appear at 1.75 MHz and 0.25 MHz.





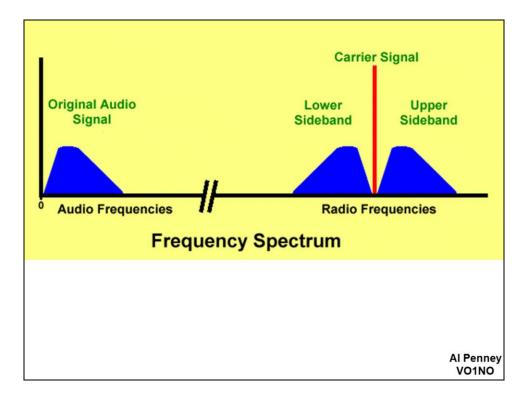
A – straight carrier wave, no modulation. First in the spectrum analyzer image (strength vs frequency), the second is an oscilloscope display (strength vs time).

B – Carrier modulated with a single tone (less than 100% modulation).



In radio communications, a **sideband** is a band of frequencies higher than or lower than the carrier frequency, that are the result of the modulation process. The sidebands carry the information transmitted by the radio signal. The sidebands comprise all the spectral components of the modulated signal except the carrier. The signal components above the carrier frequency constitute the **upper sideband** (**USB**), and those below the carrier frequency constitute the **lower sideband** (**LSB**). All forms of modulation produce sidebands.

Amplitude modulation of a carrier signal normally results in two mirrorimage sidebands. The signal components above the carrier frequency constitute the upper sideband (USB), and those below the carrier frequency constitute the lower sideband (LSB). For example, if a 900 kHz carrier is amplitude modulated by a 1 kHz audio signal, there will be components at 899 kHz and 901 kHz as well as 900 kHz in the generated radio frequency spectrum; so an audio bandwidth of (say) 7 kHz will require a radio spectrum bandwidth of 14 kHz. In conventional AM transmission, as used by *broadcast band* AM stations, the original audio signal can be recovered ("detected") by either synchronous detector circuits or by simple envelope detectors because the carrier and both sidebands are present. This is sometimes called **double** **sideband amplitude modulation** (**DSB-AM**), but not all variants of DSB are compatible with envelope detectors.



Sidebands in an AM Signal

- The two resulting range of frequencies, one above and the other below the original carrier frequency, are called the Upper Sideband (USB) and the Lower Sideband (LSB).
- These two sidebands carry the same information.
- Note that **to hear the original audio** on the receiver's speaker, the **two sidebands must mix** with the **original carrier signal**.
- The resulting **sum and difference signals** will be in the **Audio Frequency range**.

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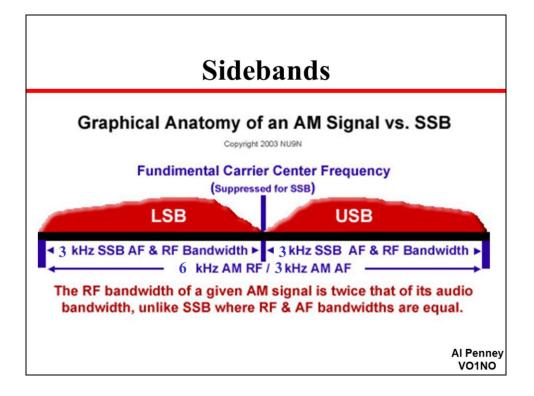
Amplitude modulation sidebands

When a carrier is modulated in any way, further signals are created either side of the steady carrier. These sidebands carry the actual modulation information.

The amplitude modulation sidebands are generated above and below the main carrier. To see how this happens, take the example of a carrier on a frequency of 1 MHz which is modulated by a steady tone of 1 kHz.

The process of modulating a carrier is exactly the same as mixing two signals together, and as a result both sum and difference frequencies are produced. Therefore when a tone of 1 kHz is mixed with a carrier of 1 MHz, a "sum" frequency is produced at 1 MHz + 1 kHz, and a difference frequency is produced at 1 MHz - 1 kHz, i.e. 1 kHz above and below the carrier.

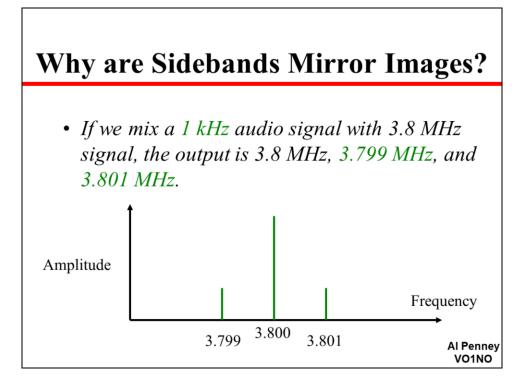
If the steady state tones are replaced with audio like that encountered with speech of music, these comprise many different frequencies and an audio spectrum with frequencies over a band of frequencies is seen. When modulated onto the carrier, these spectra are seen above and below the carrier. In order to hear the sidebands (the actual audio), the sidebands must mix with the carrier to reproduce the audio i.e.: produce the frequency difference between the carrier and the sidebands.

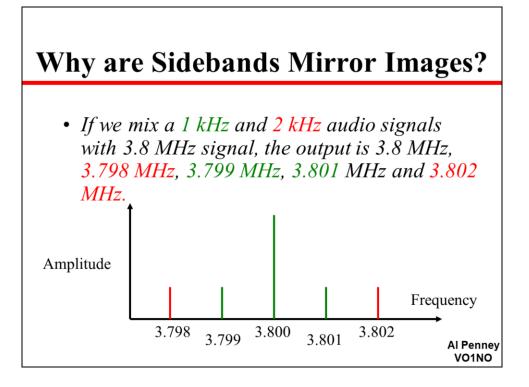


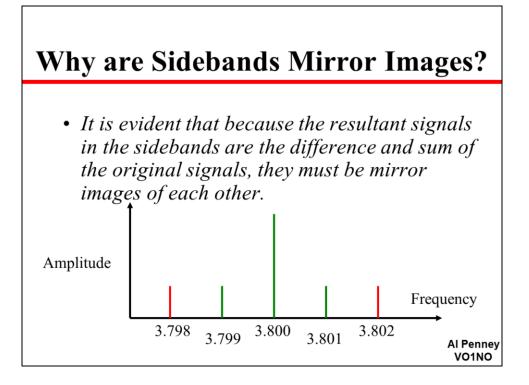
Amplitude modulation, AM bandwidth

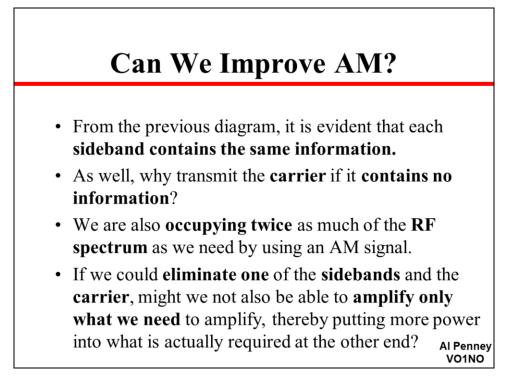
It can be seen that if the top frequency that is modulated onto the carrier is 6 kHz, then the top spectra will extend to 6 kHz above and below the signal. In other words the bandwidth occupied by the AM signal is twice the maximum frequency of the signal that is used to modulated the carrier, i.e. it is twice the bandwidth of the audio signal to be carried.

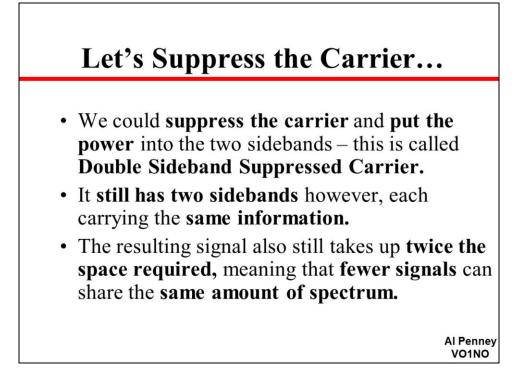
The bandwidth of amplitude modulation can be seen to be twice that of the highest audio signal to be carried. This makes in relatively poor in terms of spectral efficiency, but nevertheless AM is still used for some applications in view of its simplicity, especially in terms of demodulation.







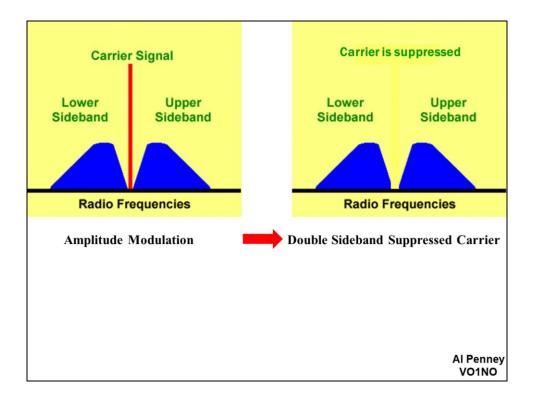




Double-sideband suppressed-carrier transmission (DSB-SC)

is transmission in which frequencies produced by amplitude modulation (AM) are symmetrically spaced above and below the carrier frequency and the carrier level is reduced to the lowest practical level, ideally being completely suppressed.

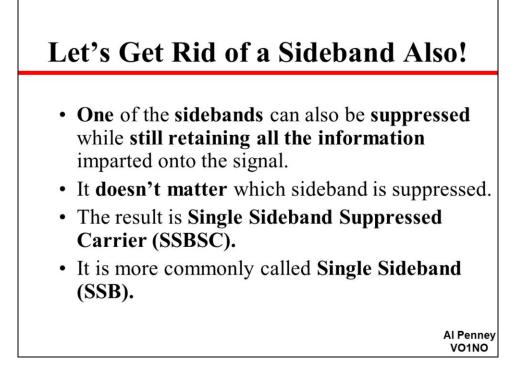
In the DSB-SC modulation, unlike in AM, the wave carrier is not transmitted; thus, much of the power is distributed between the side bands, which implies an increase of the cover in DSB-SC, compared to AM, for the same power use.



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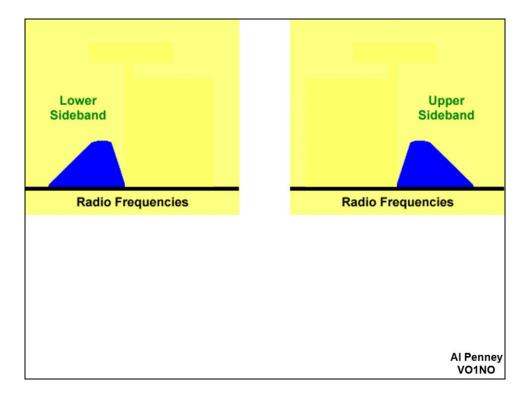


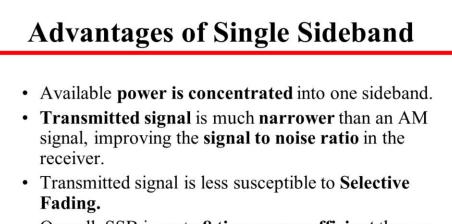
In radio communications, **single-sideband modulation** (**SSB**) or **single-sideband suppressed-carrier modulation** (**SSB-SC**) is a type of modulation used to transmit information, such as an audio signal, by radio waves. A refinement of amplitude modulation, it uses transmitter power and bandwidth more efficiently. Amplitude modulation produces an output signal the bandwidth of which is twice the maximum frequency of the original baseband signal. Singlesideband modulation avoids this bandwidth increase, and the power wasted on a carrier, at the cost of increased device complexity and more difficult tuning at the receiver.

The first U.S. patent for SSB modulation was applied for on December 1, 1915 by John Renshaw Carson. The U.S. Navy experimented with SSB over its radio circuits before World War I. SSB first entered commercial service on January 7, 1927 on the longwave transatlantic public radiotelephone circuit between New York and London. The high power SSB transmitters were located at Rocky Point, New York and Rugby, England. The receivers were in very quiet locations in Houlton, Maine and Cupar Scotland.

SSB was also used over long distance telephone lines, as part of a technique known as frequency-division multiplexing (FDM). FDM was pioneered by telephone companies in the 1930s. This enabled many voice channels to be sent down a single physical circuit, for example in L-carrier. SSB allowed channels to be spaced (usually) just 4,000 Hz apart, while offering a speech bandwidth of nominally 300–3,400 Hz.

Amateur radio operators began serious experimentation with SSB after World War II. The Strategic Air Command established SSB as the radio standard for its aircraft in 1957. It has become a de facto standard for long-distance voice radio transmissions since then.





- Overall, SSB is up to **8 times more efficient** than an AM signal.
- The price to pay is that SSB circuitry is more complex than AM, both in the transmitter and in the receiver.
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Single Sideband Modulation: SSB is a special form of AM that uses only one of the two mirror image AM bands. Complete voice information is contained in either one of these AM sidebands, so a sufficient voice signal may be sent using half the bandwidth of AM, or about 3 kHz. Why is that important? Two primary reasons:

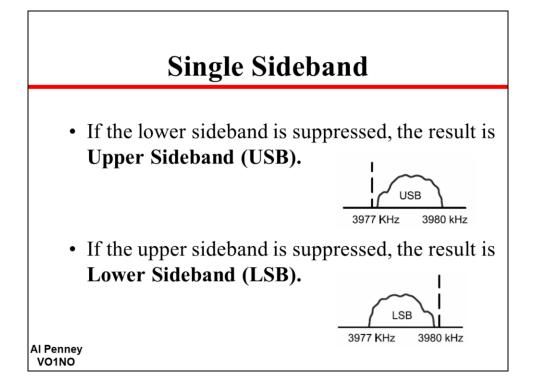
1. When each operator uses less bandwidth more signals will fit into the overall amateur band. In a geographic area where "everybody can hear everybody else," SSB would allow double the number of QSOs on a limited amateur band without interference than would AM, and about five times more than FM would allow.

2. The power of your transmitted signal must be used to generate the band of frequencies for carrying your voice on the air. You have a finite amount of power with which to transmit. When you transmit with a broad bandwidth that power is spread across the production of all those many frequencies, so the power allocated to each frequency is low. But if you have a narrow bandwidth, perhaps only half as wide, the power allocated to the production of each frequency will be doubled. Narrower bandwidth uses your power more efficiently than broad bandwidth, boosting your signal strength.

Notes on SSB

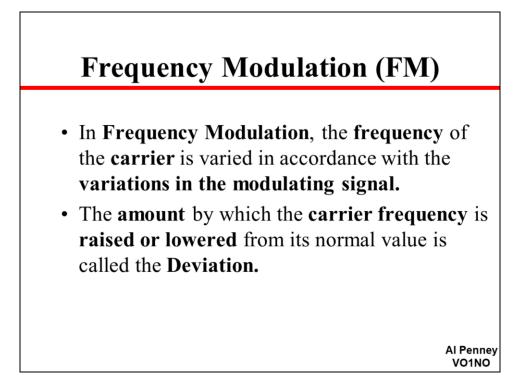
- LSB is used on 160, 80 and 40 Meter bands.
- USB is used on all other bands.
- Amateurs specify the **carrier frequency** when describing their operating frequency, even though no signal is actually transmitted there!
- Although SSB dates to the 1915, Amateur interest didn't start until the 1950s.
- It gained popularity steadily, and by the 1970s was the **standard voice mode** on the **HF bands**.

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Note that both of these signals occupy the actual spectrum, but the carrier frequency (the frequency indicated on the radio's display) differs by 3 kHz.

Note also that you must take your operating mode and signal bandwidth into account to ensure that you don't transmit a signal outside the Amateur bands!

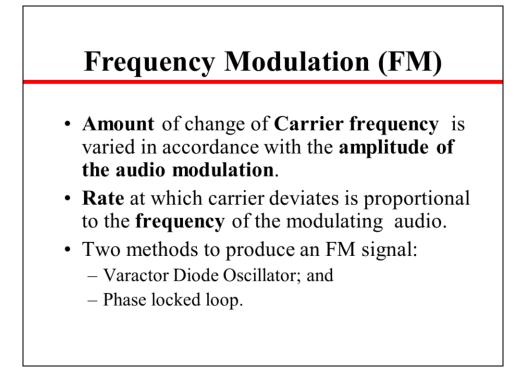


Frequency modulation (**FM**) is the encoding of information in a carrier wave by varying the instantaneous frequency of the wave. The term and technology is used in both telecommunications and signal processing.

In analog frequency modulation, such as FM radio broadcasting of an audio signal representing voice or music, the instantaneous frequency deviation, the difference between the frequency of the carrier and its center frequency, is proportional to the modulating signal.

Digital data can be encoded and transmitted via FM by shifting the carrier's frequency among a predefined set of frequencies representing digits – for example one frequency can represent a binary 1 and a second can represent binary 0. This modulation technique is known as frequency-shift keying (FSK). FSK is widely used in modems such as fax modems, and can also be used to send Morse code Radioteletype also uses FSK.

Frequency modulation is widely used for FM radio broadcasting. It is also used in telemetry, radar, seismic prospecting, and monitoring newborns for seizures via EEG, two-way radio systems, sound synthesis, magnetic tape-recording systems and some video-transmission systems. In radio transmission, an advantage of frequency modulation is that it has a larger signal-to-noise ratio and therefore rejects radio frequency interference better than an equal power amplitude modulation (AM) signal. For this reason, most music is broadcast over FM radio.



While changing the amplitude of a radio signal is the most obvious method to modulate it, it is by no means the only way. It is also possible to change the frequency of a signal to give frequency modulation or FM. Frequency modulation is widely used on frequencies above 30 MHz, and it is particularly well known for its use for VHF FM broadcasting.

Although it may not be quite as straightforward as amplitude modulation, nevertheless frequency modulation, FM, offers some distinct advantages. It is able to provide near interference free reception, and it was for this reason that it was adopted for the VHF sound broadcasts. These transmissions could offer high fidelity audio, and for this reason, frequency modulation is far more popular than the older transmissions on the long, medium and short wave bands.

In addition to its widespread use for high quality audio broadcasts, FM is also used for a variety of two way radio communication systems. Whether for fixed or mobile radio communication systems, or for use in portable applications, FM is widely used at VHF and above.

What is frequency modulation, FM?

To generate a frequency modulated signal, the frequency of the radio carrier is changed in line with the amplitude of the incoming audio signal.

When the audio signal is modulated onto the radio frequency carrier, the new radio frequency signal moves up and down in frequency. The amount by which the signal moves up and down is important. It is known as the deviation and is normally quoted as the number of kilohertz deviation. As an example the signal may have a deviation of plus and minus 3 kHz, i.e. \pm 3 kHz. In this case the carrier is made to move up and down by 3 kHz.

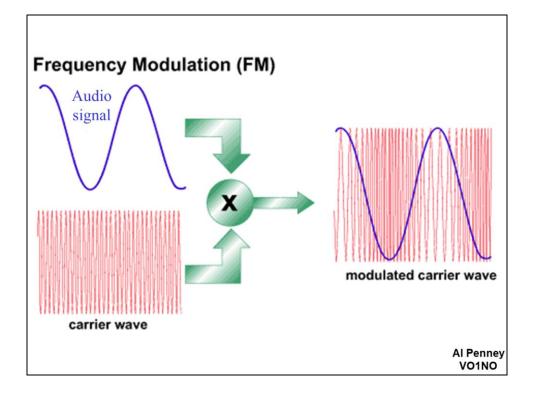
Broadcast stations in the VHF portion of the frequency spectrum between 88.5 and 108 MHz use large values of deviation, typically ±75 kHz. This is known as wide-band FM (WBFM). These signals are capable of supporting high quality transmissions, but occupy a large amount of bandwidth. Usually 200 kHz is allowed for each wide-band FM transmission. For communications purposes less bandwidth is used. Narrow band FM (NBFM) often uses deviation figures of around ±3 kHz.

It is narrow band FM that is typically used for two-way radio communication applications. Having a narrower band it is not able to provide the high quality of the wideband transmissions, but this is not needed for applications such as mobile radio communication.

There is a variety of different methods that can be used to generate frequency modulated signals.

•*Varactor diode oscillator:* This method simply requires the use of a varactor diode placed within the tuned circuit of an oscillator circuit. It is even possible to use a varactor diode within a crystal oscillator circuit. Typically when crystal oscillators a re used the signal needs to be multiplied in frequency, and only narrow band FM is attainable.

•*Phase locked loop:* Phase locked loops provide an excellent method of generating frequency modulation. It is often necessary to manage the constraints within the loop carefully but once done it provides and excellent solution.



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Frequency modulation is widely used for FM radio broadcasting. It is also used in telemetry, radar, seismic prospecting, and monitoring newborns for seizures via EEG, two-way radio systems, sound synthesis, magnetic tape-recording systems and some video-transmission systems. In radio transmission, an advantage of frequency modulation is that it has a larger signal-to-noise ratio and therefore rejects radio frequency interference better than an equal power amplitude modulation (AM) signal. For this reason, most music is broadcast over FM radio.

Frequency modulation and phase modulation are the two complementary principal methods of angle modulation; phase modulation is often used as an intermediate step to achieve frequency modulation. These methods contrast with amplitude modulation, in which the amplitude of the carrier wave varies, while the frequency and phase remain constant.

Frequency modulation

While AM is the simplest form of modulation to envisage, it is also possible to vary the frequency of the signal to give frequency modulation (FM). It can be seen from the Figure above that the frequency of the signal varies as the voltage of the modulating signal changes.

The amount by which the signal frequency varies is very important. This is known as the deviation, and is normally quoted in kilohertz. As an example, the signal may have a deviation of ± 3 kHz. In this case, the carrier is made to move up and down by 3 kHz.

FM is used for a number of reasons. One particular advantage is its resilience to signal-level variations and general interference. The modulation is carried only as variations in frequency, and this means that any signal-level variations will not affect the audio output provided that the signal is of a sufficient level. As a result, this makes FM ideal for mobile or portable applications where signal levels vary considerably. The other advantage of FM is its resilience to noise and interference when deviations much greater than the highest modulating frequency are used. It is for this reason that FM is used for highquality broadcast transmissions where deviations of ± 75 kHz are typically used to provide a high level of interference rejection. In view of these advantages, FM was chosen for use in the first-generation analogue mobile phone systems.

Pros and Cons of FM



- Very **resilient to noise** and interference;
- Can use non-linear power amplifiers.
- Cons:
 - Not as spectrally efficient as some other modes; and
 - Requires a more complicated demodulator;

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Frequency modulation advantages & disadvantages

As with any form of modulation there are several advantages and disadvantages to its use. These need to be considered before making any decision or choice about its use:

Advantages of frequency modulation, FM:

•**Resilience to noise:** One particular advantage of frequency modulation is its resilience to signal level variations. The modulation is carried only as variations in frequency. This means that any signal level variations will not affect the audio output, provided that the signal does not fall to a level where the receiver cannot cope. As a result this makes FM ideal for mobile radio communication applications including more general two-way radio communication or portable applications where signal levels are likely to vary considerably. The other advantage of FM is its resilience to noise and interference. It is for this reason that FM is used for high quality broadcast transmissions.

•Easy to apply modulation at a low power stage of the transmitter: Another advantage of frequency modulation is associated with the transmitters. It is possible to apply the modulation to a low power stage of the transmitter, and it is not

necessary to use a linear form of amplification to increase the power level of the signal to its final value.

•*It is possible to use efficient RF amplifiers with frequency modulated signals:* It is possible to use non-linear RF amplifiers to amplify FM signals in a transmitter and these are more efficient than the linear ones required for signals with any amplitude variations (e.g. AM and SSB). This means that for a given power output, less battery power is required and this makes the use of FM more viable for portable two-way radio applications.

Disadvantages of frequency modulation, FM:

•FM has poorer spectral efficiency than some other modulation formats: Some phase modulation and quadrature amplitude modulation formats have a higher spectral efficiency for data transmission than frequency shift keying, a form of frequency modulation. As a result, most data transmission system use PSK and QAM.

•**Requires more complicated demodulator:** One of the minor disadvantages of frequency modulation is that the demodulator is a little more complicated, and hence slightly more expensive than the very simple diode detectors used for AM. However this is much less of an issue these days because many radio integrated circuits incorporate a built in frequency demodulator .

•Some other modes have higher data spectral efficiency: Some phase modulation and quadrature amplitude modulation formats have a higher spectral efficiency for data transmission that frequency shift keying, a form of frequency modulation. As a result, most data transmission system use PSK and QAM.

•*Sidebands extend to infinity either side:* The sidebands for an FM transmission theoretically extend out to infinity. They are normally significant for wideband frequency modulation transmissions, although small for narrow band FM. To limit the bandwidth of the transmission, filters are often used, and these introduce some distortion of the signal. Normally this is not too much of an issue although care has to be taken to include these filters for wideband FM and to ensure they are properly designed.

Notes on Frequency Modulation

- The maximum deviation of an FM communications system must be defined in advance – this is the 100% modulation point.
- Under-deviation results in weak, "thin" audio.
- Over-deviation will cause the audio to be distorted, and will splatter onto adjacent frequencies. It may also break up. To correct this, hold the microphone further away.
- FM is known for clear, high fidelity audio and immunity to static, as well as Capture Effect.

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Noise reduction

FM provides improved signal-to-noise ratio (SNR), as compared for example with AM. Compared with an optimum AM scheme, FM typically has poorer SNR below a certain signal level called the noise threshold, but above a higher level – the full improvement or full quieting threshold – the SNR is much improved over AM. The improvement depends on modulation level and deviation. For typical voice communications channels, improvements are typically 5–15 dB. FM broadcasting using wider deviation can achieve even greater improvements. Additional techniques, such as pre-emphasis of higher audio frequencies with corresponding de-emphasis in the receiver, are generally used to improve overall SNR in FM circuits. Since FM signals have constant amplitude, FM receivers normally have limiters that remove AM noise, further improving SNR.

How FM was introduced

In the early days of radio, static was a major issue and the way everyone tried to reduce the effects of static was to reduce the bandwidth - in this way less noise was picked up by the receiver.

An American engineer named Edwin Armstrong was investigating

this issue and whether frequency modulation, rather than amplitude modulation might provide an advantage.

Around 1928, Armstrong started to develop the concept of using FM, and rather than reducing the bandwidth, he increased it.

Many did not go along with Armstrong's ideas for a variety of reasons. He approached RCA, and although they were impressed, they was focussing upon television and did not want to divert any resource onto a new form of broadcasting.

After many difficulties along the way, Armstrong launched his own radio station in 1939 to demonstrate the effectiveness of FM. To accommodate this and other stations following on the FCC allocated a band of frequencies between 42 and 50 MHz. Others soon followed, but after the war, the FCC in the USA, changed the allocated frequency band to the one we know today between 88 and 108 MHz. Although there was some initial pain because a few hundred thousand radios had been sold, the band was accepted globally and it is the VHF FM band we know today.

With FM established as a medium for high quality broadcasting, it quickly developed.

In addition to this a form of narrow band FM became popular for VHF and UHF mobile communications. The nature of FM meant that signal strength variations did not affect the operation nearly as much as if it had been an AM signal.

Capture Effect

- Phenomena in FM receiver where only the stronger of 2 signals is demodulated.
- The capture effect is defined as the **complete suppression of the weaker signal** at the receiver limiter (if it has one) where the weaker signal is not amplified, but attenuated.
- If signals are nearly identical in strength the receiver may alternate from one to the other.
- This is called "Picket Fencing".
- Aviation uses AM to avoid capture effect.

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In a radio receiver, the **capture effect**, or **FM capture effect**, is a phenomenon associated with FM reception in which only the stronger of two signals at, or near, the same <u>frequency</u> or channel will be demodulated.

The capture effect is defined as the complete suppression of the weaker signal at the receiver's limiter (if present) where the weaker signal is not amplified, but attenuated. When both signals are nearly equal in strength, or are fading independently, the receiver may switch from one to the other and exhibit picket fencing.

The capture effect can occur at the signal limiter, or in the demodulation stage, for circuits that do not require a signal limiter. Some types of radio receiver circuits have a stronger capture effect than others. The measurement of how well a receiver can reject a second signal on the same frequency is called the capture ratio for a specific receiver. It is measured as the lowest ratio of the power of two signals that will result in the suppression of the smaller signal.

Amplitude modulation, or AM radio, transmission is not subject to this effect. This is one reason that the aviation industry, and others, have chosen to use AM for communications rather than FM, allowing multiple signals transmitted on the same channel to be heard

Amateur Radio FM

- Hams use Narrow Band FM (NBFM), which has a maximum deviation of 5 kHz. The maximum modulating frequency should be 3 KHz. Commercial stations use 75 kHz.
- Total bandwidth required is 15 kHz, so FM is not allowed on HF bands EXCEPT 10M.
- Most FM is found on the 2M and 70cm bands.

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Bandwidth: The question is asking about *bandwidth*. The bandwidth of an RF signals is the range of frequencies used to carry information. It is a range of radio frequencies transmitted or received for which the power is not zero. An RF signal typically utilizes a substantial range of radio frequencies to carry information such as a set of audio frequency signals representing an operator's voice.

When you push-to-talk and speak into the microphone your FM transmitter emits a range of several thousand hertz of different frequencies and not just that singular frequency value to which you have tuned the transceiver. That displayed frequency value is only a reference value called the *carrier frequency*, and with FM the emitted signals will vary in frequency both higher and lower than that carrier frequency value by several thousand hertz. The full range of the frequencies emitted, as determined by the highest frequency value minus the lowest frequency value, is the bandwidth.

For example, suppose you are tuned to the 2-meter FM phone band carrier value 146.520 MHz and you make a call, stating your call sign. Suppose as a result that the FM transmitter emits signals representing the modulated audio of your voice from 146.526 MHz down to 146.514 MHz. The bandwidth of the signal is 146.526 – 146.514 = 0.012 MHz, or 12 kHz.

VHF Repeater FM Phone Signal: The question specifies a "VHF repeater FM phone signal." The VHF portion indicates operations on the 6-meter, 2-meter, or 1.25-meter bands. Beyond the fact that FM phone signals are commonly used on the three VHF amateur bands this information is mostly irrelevant. The fact that it is a repeater signal is also irrelevant. The fact that it is *phone mode* is very relevant! Let's see why.

With FM the amplitude of the modulating signal determines the magnitude of the frequency deviations from the carrier frequency. For phone modes the modulating signal is the audio signal generated by the microphone, and the amplitude of these audio signals represents their power. The power of the audio, and hence the amplitude of the audio signal, is determined by the combined effects of the loudness of your voice into the microphone and the microphone's audio amplifier circuit.

The upshot of these effects is that as you speak louder into the microphone and increase the amplitude of the modulating audio signals, the FM frequency deviations from the carrier value increase. As the frequency deviations increase your transmitter emits a broader range of frequencies. That is, your FM signal has greater bandwidth. So, scream into your microphone and your bandwidth gets very wide (within some limits), or whisper into the microphone and your signal is of much narrower bandwidth. Remain completely silent and your bandwidth drops to almost nothing. If your screaming results in the modulating circuit exceeding normal FM bandwidth limits, you are over-deviating and your signal may be distorted or cause interference to adjacent phone channels in the band. However, most modern transceivers have nice RF *limiter circuits* that help to avoid FM overdeviation in transmissions, but that do not eliminate distorted audio resulting from overdriving the microphone amplifier.

FM Bandwidth: From the discussion above you can see that the bandwidth of an FM phone signal will vary from moment to moment depending on the loudness of the operator's voice. This is controlled by speaking distance to the microphone and the loudness of the voice. The amplification setting of the microphone will also impact the resulting amplitude of the audio signals fed into the FM modulator circuit, and most radios provide for operator adjustment of the amplifier to help obtain good modulation for a variety of voice characteristics. Simply asking your

fellow hams on the air how your FM audio sounds is the best way to judge your own need to increase or decrease your voice volume or speaking distance to the microphone.

The typical amateur radio FM signal bandwidth varies from about 10 kHz to 15 kHz as a result of the characteristics of FM transmitter engineering for amateur radio use. A good estimate of an FM signal's bandwidth can be obtained using Carson's Rule. Carson's Rule is a simple calculation using the transmitter's engineered peak frequency deviation value and the highest modulating frequency (highest audio frequency in the phone mode case).

The bandwidth consumed by an FM phone signal becomes wider as the modulating audio amplitude becomes greater, as happens when the sound of the operator's voice is loud.

Phase Modulation

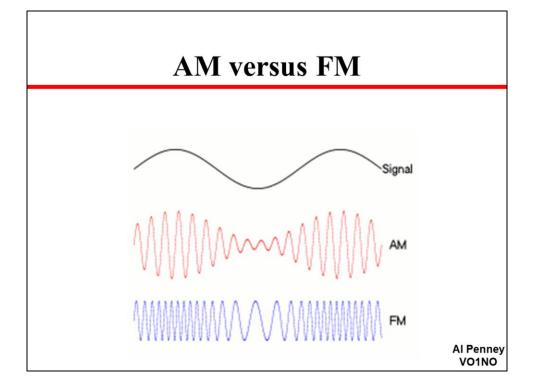
- **Phase Modulation** is similar to FM, but instead of changing the frequency of the carrier, the **phase is changed** instead.
- Phase Modulation is generated by a reactance modulator connected to an RF power amplifier!
- This is the only question I can find about Phase Modulation on the question bank!

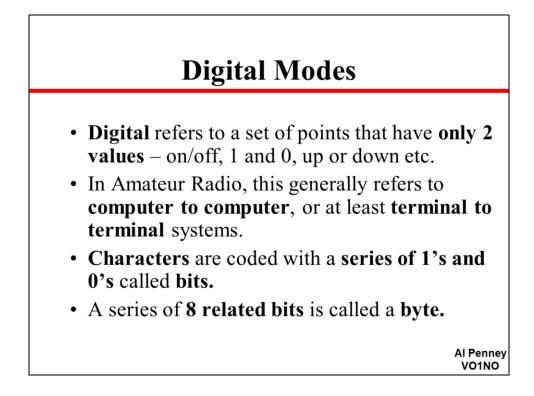
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Phase modulation (**PM**) is a modulation pattern for conditioning communication signals for transmission. It encodes a message signal as variations in the instantaneous phase of a carrier wave. Phase modulation is one of the two principal forms of angle modulation, together with frequency modulation.

The phase of a carrier signal is modulated to follow the changing signal level (amplitude) of the message signal. The peak amplitude and the frequency of the carrier signal are maintained constant, but as the amplitude of the message signal changes, the phase of the carrier changes correspondingly.

Phase modulation is widely used for transmitting radio waves and is an integral part of many digital transmission coding schemes that underlie a wide range of technologies like Wi-Fi, GSM and satellite television.





The **byte** is a unit of digital information that most commonly consists of eight bits. Historically, the byte was the number of bits used to encode a single character of text in a computer and for this reason it is the smallest addressable unit of memory in many computer architectures.

Digital Coding

- A byte could have up to 256 different values (2⁸ = 2 x 2 x 2 x 2 x 2 x 2 x 2 x 2).
- The rate of signalling is called the **baud**, which indicates the **number of signal changes per second**.
- One signal change could carry more than one bit of information, so a signal at 2400 baud could reflect a data transmission rate of 9600 bits per second for example.

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In telecommunication and electronics, **baud** is a common measure of symbol rate, which is one of the components that determine the speed of communication over a data channel.

It is the unit for symbol rate or modulation rate in *symbols per* second or pulses per second. It is the number of distinct symbol changes (signaling events) made to the transmission medium per second in a digitally modulated signal or a bd rate line code.

Baud is related to gross bit rate, which can be expressed in bits per second. If there are precisely two symbols in the system (typically 0 and 1), then baud and bit per second (bit/s) are equivalent.

Digital versus Analog

- Noise (static) is less of a problem because information has only 2 levels.
- Characters can be coded with additional information called parity bits the number of "1" bits in a character is always odd or even as specified by the coding system.
- Receive system **confirms the parity** of the byte, and will know if there is a discrepancy.
- Sometimes the byte can be **corrected**, or a **retransmission** can be requested.

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Analog and digital signals are different types which are mainly used to carry the data from one apparatus to another. Analog <u>signals are continuous</u> wave signals that change with time period whereas digital is a discrete signal is a nature. The main difference between <u>analog and digital</u> signals is, analog signals are represented with the sine waves whereas digital signals are represented with square waves. Let us discuss some dissimilarity of analog & digital signals. The best example of an analog and digital is electrons because it deals with analog as well as digital signals, input & outputs. In some way, an **electronics project** mainly interacts by the real analog world whereas digital signals are similar to different electronic languages. As some of the other languages can only recognize as well as speak one of the two. This article discusses an overview of both analog as well as digital signals, and comparison between them.

What is Analog and Digital Signal?

An analog signal is one type of continuous time-varying signals, and these are classified into composite and simple signals. A simple type of analog signal is nothing but a sine wave, and that can't be decomposed, whereas a composite type analog signal can be decomposed into numerous sine waves. An analog signal can be defined by using amplitude, time period otherwise frequency, & phase. Amplitude streaks the highest height of the signal, frequency streaks the rate at which an analog signal is varying, and phase streaks the signal position with respect to time nothing. An analog signal is not resistant toward the noise, therefore; it faces distortion as well as reduces the transmission quality. The analog signal value range cannot be fixed.

Digital signals are more resistant toward the noise; therefore, it barely faces some distortion. These waves are simple in transmitting as well as more dependable while contrasted to analog waves. Digital signals include a limited variety of values which lies among 0-to-1.

Analog signal is a continuous signal in which one time-varying quantity represents another time-based variable. These kind of signals works with physical values and natural phenomena such as earthquake, frequency, volcano, speed of wind, weight, lighting, etc.

What is a Digital Signal?

A digital signal is a signal that is used to represent data as a sequence of separate values at any point in time. It can only take on one of a fixed number of values. This type of signal represents a real number within a constant range of values.

Characteristics OF Analog Signal

Here, are essential characteristics of Analog Signal

- •These type of electronic signals are time-varying
- •Minimum and maximum values which is either positive or negative.
- •It can be either periodic or non-periodic.
- •Analog Signal works on continuous data.
- •The accuracy of the analog signal is not high when compared to the digital signal.
- •It helps you to measure natural or physical values.

•Analog signal output form is like Curve, Line, or Graph, so it may not be meaningful to all.

Characteristics of Digital Signals

Here, are essential characteristics of Digital signals

•Digital signal are continuous signals

•This type of electronic l signals can be processed and transmitted better compared to analog signal.

•Digital signals are versatile, so it is widely used.

•The accuracy of the digital signal is better than that of the analog signal.

Advantages of Analog Signals

Here, are pros/benefits of Analog Signals

•Easier in processing

•Best suited for audio and video transmission.

•It has a low cost and is portable.

•It has a much higher density so that it can present more refined information.

•Not necessary to buy a new graphics board.

•Uses less bandwidth than digital sounds

•Provide more accurate representation of a sound

•It is the natural form of a sound.

Advantages of Digital Signals

Here, are pros/advantages of Digital Signals:

•Digital data can be easily compressed.

•Any information in the digital form can be encrypted.

•Equipment that uses digital signals is more common and less expensive.

•Digital signal makes running instruments free from observation errors like parallax and approximation errors.

•A lot of editing tools are available

•You can edit the sound without altering the original copy

•Easy to transmit the data over networks

Disadvantages of Analog Signals

Here are cons/drawback of Analog Signals:

•Analog tends to have a lower quality signal than digital.

•The cables are sensitive to external influences.

•The cost of the Analog wire is high and not easily portable.

•Low availability of models with digital interfaces.

•Recording analog sound on tape is quite expensive if the tape is damaged

•It offers limitations in editing

- •Tape is becoming hard to find
- •It is quite difficult to synchronize analog sound
- •Quality is easily lost

•Data can become corrupted

•Plenty of recording devices and formats which can become confusing to store a digital signal

•Digital sounds can cut an analog sound wave which means that you can't get a perfect reproduction of a sound

•Offers poor multi-user interfaces

Disadvantage of Digital Signals

•Sampling may cause loss of information.

•A/D and D/A demands mixed-signal hardware

•Processor speed is limited

•Develop quantization and round-off errors

•It requires greater bandwidth

•Systems and processing is more complex.

KEY DIFFERENCES:

•An analog signal is a continuous signal whereas Digital signals are time separated signals.

•Analog signal is denoted by sine waves while It is denoted by square waves

•Analog signal uses a continuous range of values that help you to represent information on the other hand digital signal uses discrete 0 and 1 to represent information.

•The analog signal bandwidth is low while the bandwidth of the digital signal is high.

•Analog instruments give considerable observational errors whereas Digital instruments never cause any kind of observational errors.

•Analog hardware never offers flexible implementation, but Digital hardware offers flexibility in implementation.

•Analog signals are suited for audio and video transmission while Digital signals are suited for Computing and digital electronics.

Parity Bits

A **parity bit**, or **check bit**, is a bit added to a string of binary code. Parity bits are used as the simplest form of error detecting code. Parity bits are generally applied to the smallest units of a communication protocol, typically 8-bit octets (bytes), although they can also be applied separately to an entire message string of bits.

The parity bit ensures that the total number of 1-bits in the string is even or odd. Accordingly, there are two variants of parity bits: **even parity bit** and **odd parity bit**. In the case of even parity, for a given set of bits, the occurrences of bits whose value is 1 are counted. If that count is odd, the parity bit value is set to 1, making the total count of occurrences of 1s in the whole set (including the parity bit) an even number. If the count of 1s in a given set of bits is already even, the parity bit's value is 0. In the case of odd parity, the coding is reversed. For a given set of bits, if the count of 1s in the whole set (including the parity bit value is set to 1 making the total count of 1s in the whole set (including the parity bit value is set to 1 making the total count of 1s in the value of 1 is even, the parity bit value is set to 1 making the total count of 1s in the value of 1 is odd, the count is already odd so the parity bit's value is 0.

Radio Teletype

- The first digital mode used by Amateurs.
- Bits known as **mark** and **space**, mapped to two different frequencies, usually **170 Hz apart**.
- This frequency difference is called the shift.
- Transmitted at 60 words per minute.

Al Penney VO1NO

Radioteletype (**RTTY**) is a telecommunications system consisting originally of two or more electromechanical teleprinters in different locations connected by radio rather than a wired link. These machines were superseded by personal computers (PCs) running software to emulate teleprinters. Radioteletype evolved from earlier landline teleprinter operations that began in the mid-1800s. The US Navy Department successfully tested printing telegraphy between an airplane and ground radio station in 1922. Later that year, the Radio Corporation of America successfully tested printing telegraphy via their Chatham, Massachusetts, radio station to the R.M.S. Majestic. Commercial RTTY systems were in active service between San Francisco and Honolulu as early as April 1932 and between San Francisco and New York City by 1934. The US military used radioteletype in the 1930s and expanded this usage during World War II. From the 1980s, teleprinters were replaced by computers running teleprinter emulation software.

The term radioteletype is used to describe both the original radioteletype system, sometimes described as "Baudot", as well as the entire family of systems connecting two or more teleprinters or PCs using software to emulate teleprinters, over radio, regardless of alphabet, link system or modulation.

In some applications, notably military and government, radioteletype is known by the acronym RATT (Radio Automatic Teletype).

Technical description

A radioteletype station consists of three distinct parts: the Teletype or teleprinter, the modem and the radio.

The Teletype or teleprinter is an electromechanical or electronic device. The word *Teletype* was a trademark of the Teletype Corporation, so the terms "TTY", "RTTY", "RATT" and "teleprinter" are usually used to describe a generic device without reference to a particular manufacturer.

Electromechanical teleprinters were heavy, complex and noisy, and have been replaced with electronic units. The teleprinter includes a keyboard, which is the main means of entering text, and a printer or visual display unit (VDU). An alternative input device is a perforated tape reader and, more recently, computer storage media (such as floppy disks). Alternative output devices are tape perforators and computer storage media.

The line output of a teleprinter can be at either digital logic levels (+5 V signifies a logical "1" or *mark* and 0 V signifies a logical "0" or *space*) or line levels (-80 V signifies a "1" and +80 V a "0"). When no traffic is passed, the line idles at the "mark" state.

Baudot Code

- Uses 5 bit groups, so only 32 unique characters are possible (2⁵).
- In order to include the alphabet, numbers and special characters, the **code set is used twice**, with an "**upper**" and "**lower**" character used to switch between the two sets.
- Baud rate is 45.5

Al Penney VO1NO

The **Baudot code** [bodo] is an early character encoding for telegraphy invented by Émile Baudot in the 1870s. It was the predecessor to the International Telegraph Alphabet No. 2 (ITA2), the most common teleprinter code in use until the advent of ASCII. Each character in the alphabet is represented by a series of five bits, sent over a communication channel such as a telegraph wire or a radio signal. The symbol rate measurement is known as baud, and is derived from the same name.

When a key of the teleprinter keyboard is pressed, a 5-bit character is generated. The teleprinter converts it to serial format and transmits a sequence of a *start bit* (a logical 0 or space), then one after the other the 5 data bits, finishing with a *stop bit* (a logical 1 or mark, lasting 1, 1.5 or 2 bits). When a sequence of start bit, 5 data bits and stop bit arrives at the input of the teleprinter, it is converted to a 5-bit word and passed to the printer or VDU. With electromechanical teleprinters, these functions required complicated electromechanical devices, but they are easily implemented with standard digital electronics using shift registers. Special integrated circuits have been developed for this function, for example the Intersil 6402 and 6403. These are stand-alone UART devices, similar to computer serial port peripherals.

The 5 data bits allow for only 32 different codes, which cannot accommodate the 26 letters, 10 figures, space, a few punctuation marks and the required control codes, such as carriage return, new line, bell, etc. To overcome this limitation, the teleprinter has two *states*,

the *unshifted* or *letters* state and the *shifted* or *numbers* or *figures* state. The change from one state to the other takes place when the special control codes *LETTERS* and *FIGURES* are sent from the keyboard or received from the line. In the *letters* state the teleprinter prints the letters and space while in the shifted state it prints the numerals and punctuation marks. Teleprinters for languages using other alphabets also use an additional *third shift* state, in which they print letters in the alternative alphabet.

The modem is sometimes called the terminal unit and is an electronic device which is connected between the teleprinter and the radio transceiver. The transmitting part of the modem converts the digital signal transmitted by the teleprinter or tape reader to one or the other of a pair of audio frequency tones, traditionally 2295/2125 Hz (US) or 2125/1955 Hz (Europe). One of the tones corresponds to the *mark* condition and the other to the *space* condition. These audio tones, then, modulate an SSB transmitter to produce the final audio-frequency shift keying (AFSK) radio frequency signal. Some transmitters are capable of direct frequency-shift keying (FSK) as they can directly accept the digital signal and change their transmitting frequency according to the *mark* or *space* input state. In this case the transmitting part of the modem is bypassed.

On reception, the FSK signal is converted to the original tones by mixing the FSK signal with a local oscillator called the BFO or *beat frequency oscillator*. These tones are fed to the demodulator part of the modem, which processes them through a series of filters and detectors to recreate the original digital signal. The FSK signals are audible on a communications radio receiver equipped with a BFO, and have a distinctive "beedle-eeedle-eeelle-eee" sound, usually starting and ending on one of the two tones ("idle on mark").

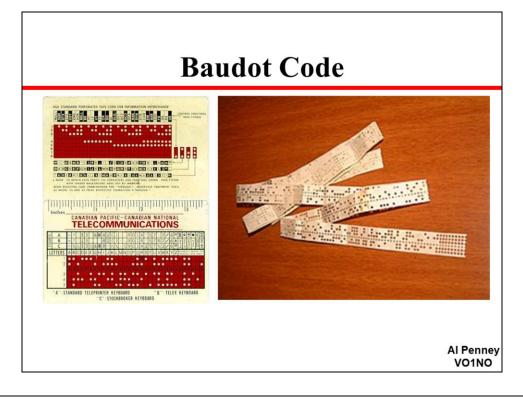
The transmission speed is a characteristic of the teleprinter while the shift (the difference between the tones representing mark and space) is a characteristic of the modem. These two parameters are therefore independent, provided they have satisfied the minimum shift size for a given transmission speed. Electronic teleprinters can readily operate in a variety of speeds, but mechanical teleprinters require the change of gears in order to operate at different speeds.

Today, both functions can be performed with modern computers equipped with digital signal processors or sound cards. The sound card performs the functions of the modem and the CPU performs the processing of the digital

bits. This approach is very common in amateur radio, using specialized computer programs like fldigi, MMTTY or MixW.

Before the computer mass storage era, most RTTY stations stored text on paper tape using paper tape punchers and readers. The operator would type the message on the TTY keyboard and punch the code onto the tape. The tape could then be transmitted at a steady, high rate, without typing errors. A tape could be reused, and in some cases - especially for use with ASCII on NC Machines - might be made of plastic or even very thin metal material in order to be reused many times.

The most common test signal is a series of "RYRYRY" characters, as these form an alternating tone pattern exercising all bits and are easily recognized. Pangrams are also transmitted on RTTY circuits as test messages, the most common one being "The quick brown fox jumps over the lazy dog", and in French circuits, "Voyez le brick géant que j'examine près du wharf"



Punched tape or **perforated paper tape** is a form of data storage that consists of a long strip of paper in which holes are punched. Now effectively obsolete, it was widely used during much of the 20th century for teleprinter communication, for input to computers of the 1950s and 1960s, and later as a storage medium for minicomputers and CNC machine tools.

Punched tape was used as a way of storing messages for teletypewriters. Operators typed in the message to the paper tape, and then sent the message at the maximum line speed from the tape. This permitted the operator to prepare the message "off-line" at the operator's best typing speed, and permitted the operator to correct any error prior to transmission. An experienced operator could prepare a message at 135 words per minute (WPM) or more for short periods.

The line typically operated at 75WPM, but it operated continuously. By preparing the tape "off-line" and then sending the message with a tape reader, the line could operate continuously rather than depending on continuous "on-line" typing by a single operator. Typically, a single 75WPM line supported three or more teletype operators working offline. Tapes punched at the receiving end could be used to relay messages to another station. Large store and forward networks were developed

using these techniques.

Model 19 Teletype Set



Standard Model: 19 Military Models: TT-7/FG, TT-8/FG Design Relatives: M15-KSR, M14-TD, M14-ROTR? Manufactured: 1942-1950? Units Produced: ?? Units Remaining: ?? (estimated) Dimensions (inches): 38L x 24W x 27H Weight (pounds): ??

Keyboard: 3-row with spring-cushioned green keycaps
Code: 5-level baudot (ustty or ita2) at 60 wpm (45.5 baud)
Interface: 60-mA current loop (@120VDC typical)
Full- or Half-Duplex; 60/20-mA selectable on later models
Motors: 115-VAC Synchronous, or Governed
Options: Line relay; Polar-line relay; Motor-control relay;
Line-Break/Keyboard-Lockout; RFI-Suppression; Remote bell;
Auto-Carriage-Return/Line-Feed; Tabulator; Sprocket-feed paper drive.

RTTY Transmission

- Transmission is a **shifted carrier**, where the **carrier rests on the mark frequency**, and is **shifted 170 Hz** to the **space** frequency.
- To minimize interference with adjacent stations, try to stay 250 to 500 Hz away.
- Use LSB for RTTY, regardless of the band.

Al Penney VO1NO

RTTY basics

RTTY uses a form of transmission known as frequency shift keying. The code representing the letters consists of a series of bits represented by high and low voltages. In turn these are represented on the radio signal by a shift between two frequencies, one frequency signifying a mark or high voltage and another frequency representing a space of a low voltage.

On the HF bands the carrier for the RTTY signal is shifted between two frequencies, and this gives rise to differing audio tones when a beat frequency oscillator or BFO is used. At VHF and above a frequency modulated signal is generally used for RTTY and this is modulated by an audio tone that changes.

RTTY Transmission

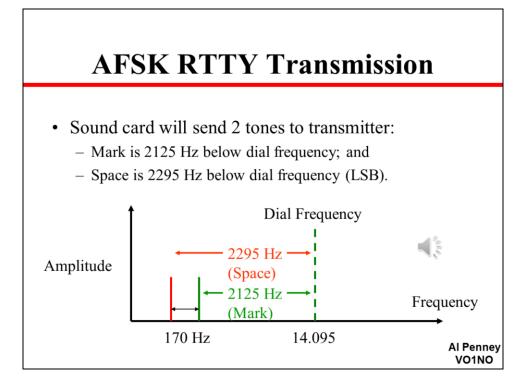
- Mark frequency is 2125 Hz, Space is 2295 Hz (away from the frequency displayed by the radio's readout).
- Some systems will key the transmitter directly, alternating directly between the mark and space. This is called FSK (Frequency Shift Keying).
- Most Hams use a **computer soundcard** connected to the **microphone input** of the radio.
- Two tones, corresponding to mark and space, are sent to the mic input when the radio is keyed. This is called Audio Frequency Shift Keying (AFSK), but is indistinguishable from FSK.

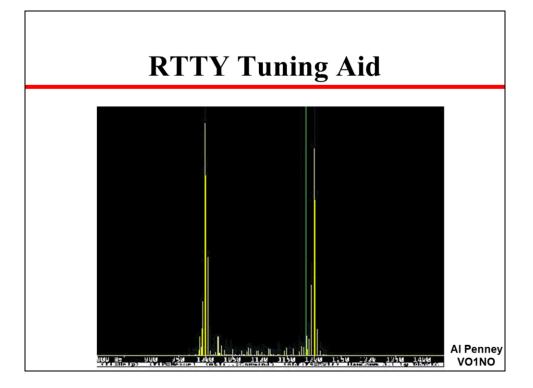
Al Penney VO1NO

Teletype on the Air

To send teletype signals on the air, the transmitter generates a continuous carrier that is shifted slightly between two different frequencies that correspond to the mark or space states. This technique is known as "frequency shift keying" or FSK. Nowadays FSK transmission is normally achieved by feeding an audio tone generator (or, more usually, a computer sound card) into an SSB or FM transmitter. This technique is known as AFSK (audio frequency shift keying).

On the receiving end, RTTY stations originally used decoders known as "terminal units". The terminal unit would decode the incoming audio from the receiver and convert the two-tone signal to a series of pulses that were then sent to the teleprinter. Nowadays the decoding can be done by a computer sound card, so there is no need for a separate terminal unit.





Packet Radio

- **Computer to computer mode** that once was the most popular digital mode, particularly on 2M.
- Data is bundled into packets of information.
- A Terminal Node Controller (TNC) is used to connect the radio with the computer.
- VHF Packet is sent at 1200 baud.



- HF Packet is sent at 300 baud.
- Uses 8-bit ASCII code (American Standard Code for Information Interchange).

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Packet radio was widely used on the amateur bands, particularly on frequencies above 30 MHz where it was well established and forming one of the mainstays of communications within the amateur community. The basic system allows modern day computer technology to be used to enable error free communication combined with many useful facilities, making it a mode that is compatible with many computer style applications, and allows people to combine the hobby of amateur radio with computer technology.

Fortunately it is not difficult to convert an existing amateur radio station to be able to transmit packet radio. Typically most VHF / UHF FM equipment is capable of transmitting amateur radio packet signals. Even many small handheld radios provide an excellent means of communicating via this means. As many people already have computers, the investment required is often minimal.

Packet radio basics

As the name implies this mode of transmission splits the data to be sent up into a series of packets which can be sent one at a time. As messages are usually much longer than the amount of data which can be sent in one packet, it takes several packets to complete the message.

One of the advantages of amateur packet radio is that one channel can be used by several amateur radio stations at the same time. This means that when sending data any station has to wait until the channel is clear. Once the frequency is free the first packet can be sent, and the receiving station will return an acknowledgement to say that all the data has been received correctly. If this acknowledgement is not received the transmitting station waits for the frequency to clear and re-sends the data. This process is repeated until the data has been correctly received. Once the first packet has been transferred, the second, and subsequent ones are all transmitted in the same way.

As the receiving radio station checks for errors and the transmitter repeats the data until it has been correctly received the system is very resilient and gives very high levels of accuracy. The other advantage is that the approach of waiting until the frequency is clear before transmitting allows many stations to use the same frequency, providing an efficient utilisation of the available spectrum. Nevertheless traffic is often high and as a result several channels may be allocated for amateur packet radio on a given band.

Details for amateur packet radio transmissions

Like other data modes or digimodes, packet radio uses frequency shift keying. A transmission speed of 1200 baud with tone frequencies of 2200 Hz for the space and 1200 Hz for the mark condition have been adopted for VHF. On HF where conditions are a little more difficult a speed of 300 baud with a 200 Hz shift is generally employed.

The format for each data packet is accurately defined so that the receiver can decipher the incoming data. Data is sent using ASCII (American Standard Code for Information Interchange) and each packet has five different elements or sections. There are flags at the beginning and end of each packet, an address, control information, a frame check sequence, and the data itself.

The flag at the beginning of the packet is used to allow the receiving decoder to synchronise to the incoming data. This is followed by a station address. This is used to define the callsign of the station to whom the data is being sent. Also included is the source or sending station callsign, and

the callsigns of any repeaters or digipeaters which are to be used to relay the message. This means that any other station using the frequency will be able to ignore the data and only receive the signals intended for it.

The element within the data packet is the control byte. This is used to signal acknowledgements and requests to repeat transmissions. This is followed by the data itself. The length of this can be up to 256 bytes. Once it is complete the frame check sequence or FCS is sent. This is a check-sum whose value is calculated from the data. It is used by the receiving station to check that all the data has been correctly received. Only when the receiver is able to generate an identical code to match the received one is an acknowledgement sent.

The final part of the packet radio transmission is the terminating flag. This recognised by the receiver as the end of the message and enables it to check the data and send its acknowledgement.

Amateur packet radio features

Packet radio is able to utilise a number of features which were not present in previous types of data communication. One of the most widely used is the ability for other stations to relay messages, so that much greater distances can be covered. Stations which relay messages in this way are called digital repeaters or digipeaters for short.

Packet radio transmissions take place on a single frequency. This means that digipeaters have to receive and transmit on a single frequency. For them to be able to relay the messages, the message must first be received in full, stored and then transmitted. Once the final station in the chain has received the message the acknowledgement is sent back along the chain to the first station. This is known as an end to end acknowledgement. Only then is the next packet sent. This means that when a message is sent over a long path using several stations as repeaters, the message can take a very long time to get through, especially if any packets have to be repeated.

One powerful facility which amateur packet radio offers is the ability to read data from a mailbox. Sometimes called a bulletin board system (BBS), it enables messages to be sent to a particular mailbox and left for collection by a particular station. In many respects it is like a radio e-mail system.

A message is sent to the local mailbox. Once received it is stored and then it is passed on via a network of mailbox stations until the required destination mailbox is reached. The message is stored at this mailbox until it is read by the recipient station. The advantage of using the mailbox system is that it is not necessary to know the route required to be taken by the message. This is worked out by the system, as it has a knowledge of the stations and works out a suitable route. Data is generally sent at periods of low activity, often at night, and this means that messages can take a few days to arrive. However as many links exist between countries it is possible to send messages around the world.

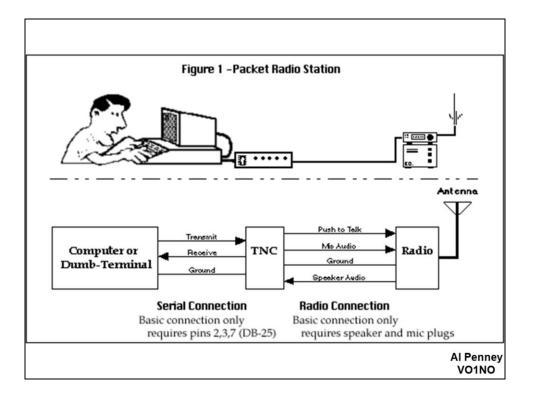
In addition to the basic mail facility, many items of general interest are stored and can be accessed by any station.

Although much of the initial experimental and practical work regarding packet radio was carried out by radio amateurs many commercial packet systems are now in use around the world. They are particularly useful where a large number of users have to send small amounts of data at intervals which would not demand a separate frequency for each separate user. For example packet radio is ideal for monitoring systems where each outpost has to be polled or accessed at intervals, or where it periodically reports a status or other information to the main station.

ASCII abbreviated from American Standard Code for Information

Interchange, is a <u>character encoding</u> standard for electronic communication. ASCII codes represent text in computers, telecommunications equipment, and other devices. Most modern character-encoding schemes are based on ASCII, although they support many additional characters.

Originally based on the English alphabet, ASCII encodes 128 specified characters into seven-bit integers. Ninety-five of the encoded characters are printable: these include the digits 0 to 9, lowercase letters a to z, uppercase letters A to Z, and punctuation symbols. In addition, the original ASCII specification included 33 non-printing control codes which originated with Teletype machines; most of these are now obsolete, although a few are still commonly used, such as the carriage return, line feed and tab codes.

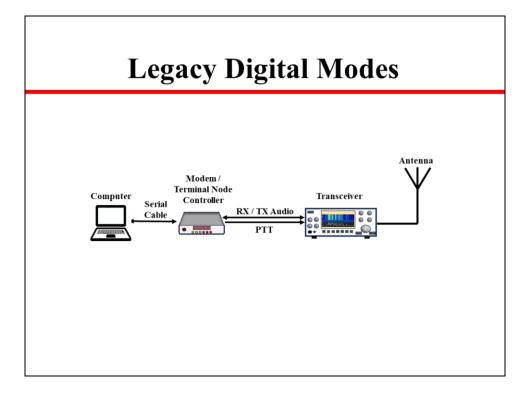


WHAT IS A TNC?

A "Terminal Node Controller" (TNC) is similar to the modem you use when connecting to the internet. One difference is; The TNC is used to interface our terminal or computer into the "RF" or radio (wireless) medium. There is one other, *very* significant difference; Inside the TNC we have added some internal firmware called a "PAD." The pad or "Packet assembler/disassembler" captures incoming and out-going data and assembles it into "packets" of data that can be sent to and from a data radio or transceiver.

In addition to the data stream conversion to and from packets, the PAD also enables the Push-To-Talk (PTT) circuits of the radio transceiver. When you press the enter key of your computer keyboard, the typed in data is sent out over the air to the target station or a nearby "store-and-forward" device known as a "node."

Incoming (received) data from the transceiver is also converted within the PAD, from Packets of data into a stream of usable data and sent to the TNC/modem. Here the data stream is sent to the serial comport of the computer for display on the screen, or manipulated by a resident terminal program into on-screen text, pictures, or save-to-disk processing.



Packet Radio

- Stations are linked by the **Connected** mode, ensuring that packets are re-transmitted if not received properly.
- Each packet requires an "Acknowledgement".
- You can see what traffic is being sent without being connected by using the "Monitor" mode.
- Repeaters called **Digipeaters** are used to **receive**, **store and re-transmit packets marked for re-transmission** in order to extend the range, sometimes into a **network of digipeaters**.
- The protocol used to transmit Packet data is called
 AX.25.
 Al Penney
 VO1NO

"Digipeater" is short for "Digital Repeater"; a repeater for packet data rather than voice. Unlike the standard voice repeater that receives on one frequency and retransmits what it hears simultaneously on another frequency, the usual digipeater is a single frequency device. It receives a packet of data, stores it in internal memory and then a moment later retransmits it on the SAME frequency.

Automatic Packet Reporting System

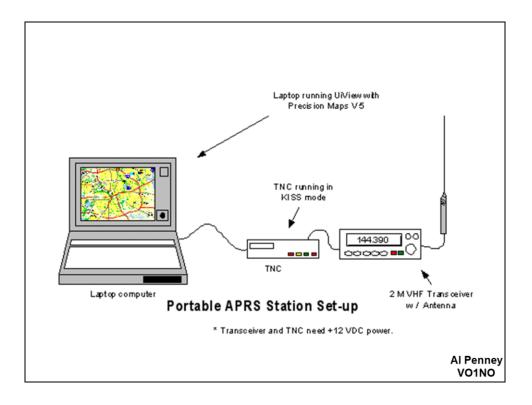
- An Amateur Radio-based system for **real time tactical digital communications** of information of immediate value in the local area.
- Now the primary use of Packet Radio.
- Displays **position**, weather info, announcements etc. in an unconnected broadcast manner.
- Retransmitted using **digipeaters** and the **Internet**.
- Maps are an integral part of the system.

Al Penney VO1NO

Automatic Packet Reporting System (APRS) is an amateur radiobased system for real time digital communications of information of immediate value in the local area. Data can include object Global Positioning System (GPS) coordinates, weather station telemetry, text messages, announcements, queries, and other telemetry. APRS data can be displayed on a map, which can show stations, objects, tracks of moving objects, weather stations, search and rescue data, and direction finding data.

APRS data is typically transmitted on a single shared frequency (depending on country) to be repeated locally by area relay stations (digipeaters) for widespread local consumption. In addition, all such data are typically ingested into the APRS Internet System (APRS-IS) via an Internet-connected receiver (IGate) and distributed globally for ubiquitous and immediate access. Data shared via radio or Internet are collected by all users and can be combined with external map data to build a shared live view.

APRS has been developed since the late 1980s by Bob Bruninga, call sign WB4APR, currently a senior research engineer at the United States Naval Academy. He still maintains the main APRS Web site. The initialism "APRS" was derived from his call sign.

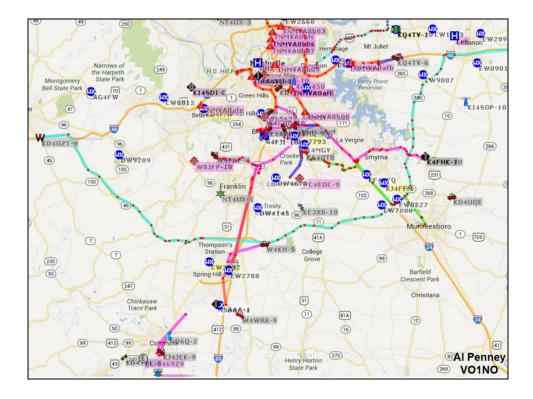




Portable APRS gateway station we use for tracking high altitude balloon launches by the Annapolis Royal Space Agency.



Portable APRS gateway station we use for tracking high altitude balloon launches by the Annapolis Royal Space Agency. The small box is the TNC,



APRS map of Nashville TN. www.aprs.fi

AMTOR

- Amateur Teleprinting Over Radio.
- Rarely used today.
- For the test: Mode A uses Automatic Repeat Request (ARQ) protocol and is normally used for one-on-one communications after contact has been established.

Al Penney VO1NO

AMTOR (Amateur Teleprinting Over Radio) is a type of telecommunications system that consists of two or more electromechanical teleprinters in different locations that send and receive messages to one another. AMTOR is a specialized form of RTTY protocol. The term is an acronym for Amateur Teleprinting Over Radio^[11] and is derived from ITU-R recommendation 476-1 and is known commercially as SITOR (Simplex Telex Over radio) developed primarily for maritime use in the 1970s.^[21] AMTOR was developed in 1978 by Peter Martinez, G3PLX, with the first contact taking place in September 1978 with G3YYD on the 2m Amateur band. It was developed on homemade Motorola 6800-based microcomputers in assembler code. It was used extensively by <u>amateur radio</u> operators in the 1980s and 1990s but has now fallen out of use as improved PC-based data modes are now used and teleprinters became out of fashion.

AMTOR improves on RTTY by incorporating error detection or error correction techniques. The protocol remains relatively uncomplicated and AMTOR performs well even in poor and noisy HF conditions. AMTOR operates in one of two modes: an error detection mode and an automatic repeat request (ARQ) mode.

The AMTOR protocol utilizes a 7-bit code for each character, with each code-word having four mark and three space bits. If the received code

does not match a four-to-three (4:3) ratio, the receiver assumes an error has occurred. In error detection mode, the code word will be dropped; in automatic repeat request mode, the receiver requests that the original data be resent. AMTOR also supports FEC in which simple bit-errors can be corrected.

AMTOR utilizes FSK, with a frequency shift of 170 Hz, and a symbol rate of 100 Baud.

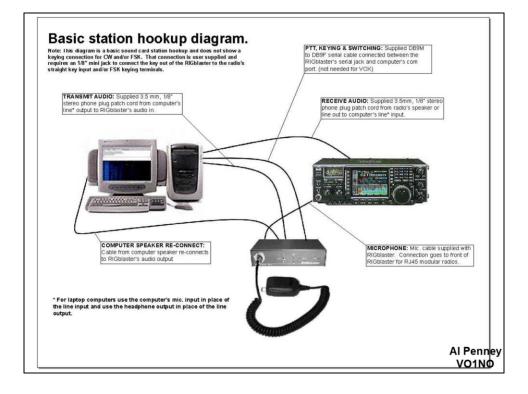
AMTOR is rarely used today, as other protocols such as PSK31 are becoming favoured by amateur operators for real-time text communications

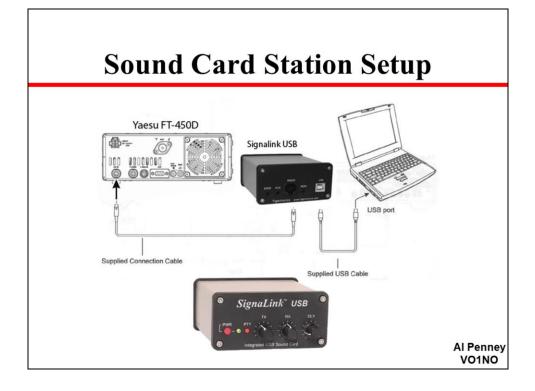
Sound Card Modes

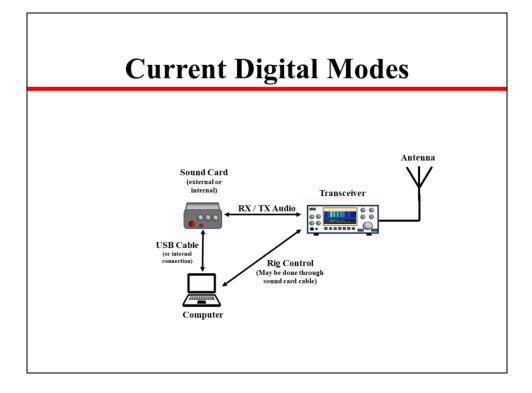
- The introduction of **simple interfaces** between computers and radios has **revolutionized** Amateur Radio digital modes.
- As **new modes** are developed, they can be **downloaded free** of charge.

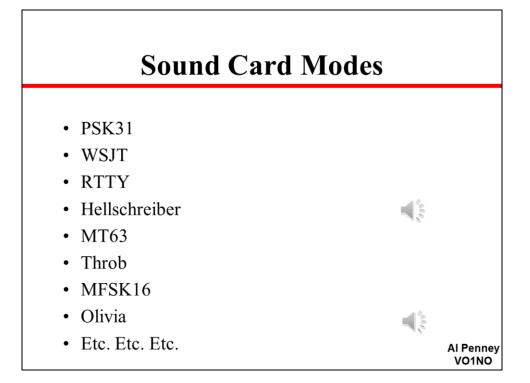
Al Penney VO1NO

http://www.ws1sm.com/Digital-Modes.html









OLIVIA

<u>OLIVIA</u> is a new digital MFSK mode created in 2005 by Pawel Jalocha (SP9VRC), who also developed MT63. OLIVIA seems highly resistant to fading and noise, and incorporates Forward Error Correction (FEC), based on Walsh functions. As with other modes, OLIVIA has several variants, each having a different bandwidth (from 500Hz to 2 kHz) and different tones. The mode can combine 4-256 tones (2ⁿ), with a 250, 500, 1000, or 2000 Hz bandwidth. The prevailing standard setting is 32 tones and 1000 Hz with 31.25 baud. This allows for ±125 Hz of mis-tuning.

Where can I find OLIVIA signals?

40 METERS 7038.5 20 METERS 14104.5, 14105.5, 14106.5, 14107.5, 14108.5 (Calling Frequency) 17 METERS 18102.5, 18103.5, 18104.5 15 METERS 21129.5

A new variation, <u>CONTESTIA</u>, is a digital MFSK mode derived from OLVIA by Nick Fedoseev (UT2UZ). One of the best programs that features both of these new modes is <u>FLDIGI</u>.

WSPR

WSPR, (pronounced "whisper"), stands for *Weak Signal Propagation Reporter*. It is an open source software tool that uses the transmission mode MEPT-JT, written by Joe Taylor, (K1JT), of Princeton, New Jersey. WSPR is designed for sending and receiving low-power transmissions to test potential propagation paths on the MF and HF bands.

According to amateur-radio-wiki.net, in MEPT transmissions, the "radio becomes a beacon that transmits for just under 2 minutes." Normal transmissions carry a station's call sign, maidenhead grid locator, and transmitter power in dBm. Modulation is by narrow-band FSK. The program is said to be able to detect and decode signals with a signal-to-noise ratio as low as -28 dB in a 2500 Hz bandwidth. A nice feature about WSPR is that "spots can be automatically downloaded to a central database: <u>WSPRnet.org</u> and results can be shown on a large map."

JT9

JT9, developed by Joe Taylor K1JT, is intended for MF and HF use, and was introduced in an experimental version of the WSJT software, known as <u>WSJT-X</u>. It uses the same logical encoding as JT65, but modulates to a 9-FSK signal. With 1-minute transmission intervals, JT9 occupies less than 16 Hz bandwidth. JT9 also has versions designed for longer transmission intervals of 2 minutes, 5 minutes, 10 minutes or 30 minutes. These extended versions take increasingly less bandwidth and permit reception of even weaker signals.

FEC in JT9 uses the same strong convolutional code as JT4: constraint length K=32, rate r=1/2, and a zero tail, leading to an encoded message length of $(72+31) \times 2 = 206$ information-carrying bits. Modulation is nine-tone frequency-shift keying, 9-FSK at 12000.0/6912 = 1.736 baud. Eight tones are used for data, one for synchronization. Eight data tones means that three data bits are conveyed by each transmitted information symbol. Sixteen symbol intervals are devoted to synchronization, so a transmission requires a total of 206 / 3 + 16 = 85 (rounded up) channel symbols.

The sync symbols are those numbered 1, 2, 5, 10, 16, 23, 33, 35, 51, 52, 55, 60, 66, 73, 83, and 85 in the transmitted sequence. Tone spacing of the 9-FSK modulation for JT9A is equal to the keying rate, 1.736 Hz. The total occupied bandwidth is $9 \times 1.736 = 15.6$ Hz.

- Modulation: 9-FSK
- •Bandwidth: 15.6 hz

- •TX duration: 49 secs.
- •Baud: 1.736
- Min SNR:-27db based on 2500hz bandwidth noise

If you ran across this on the radio and were not familiar with the mode you might think it is a birdie in your radio due to the small bandwidth.

Phase Shift Keying 31 (PSK 31)

- Was the most popular HF digital mode until FT8.
- It combines the advantages of a simple variable length text code with a narrow bandwidth phase-shift keying (PSK) signal using DSP techniques.
- Uses a simple interface between the radio and computer sound card.
- Excellent low power capabilities.

Al Penney VO1NO

PSK31 or "Phase Shift Keying, 31 Baud", also **BPSK31** and **QPSK31**, is a popular computer-sound card-generated radioteletype mode, used primarily by amateur radio operators to conduct real-time keyboard-tokeyboard chat, most often using frequencies in the high frequency amateur radio bands (near-shortwave). PSK31 is distinguished from other digital modes in that it is specifically tuned to have a data rate close to typing speed, and has an extremely narrow bandwidth, allowing many conversations in the same bandwidth as a single voice channel. This narrow bandwidth makes better use of the RF energy in a very narrow space thus allowing relatively low-power equipment (5 watts) to communicate globally using the same skywave propagation used by shortwave radio stations.

Peter Martinez (G3PLX), who is responsible for adapting the commercial SITOR mode as AMTOR for the amateur radio bands, is credited with inventing <u>PSK31</u>. Over the last decade the mode has been among the most popular of all HF digital modes, especially for keyboard-to-keyboard operating, but that almost wasn't the case. In 2001, Steve Ford, author of the *ARRL HF Digital Handbook*, wrote that "for a few years, PSK31 languished in obscurity because special DSP hardware was required to use it. But in 1999, Martinez designed a version of PSK31 that needed nothing more than a common sound

card" (Ford). To make it even better, the software was made available for free. Shortly thereafter, other programs like DigiPan and WinPSK, made operating PSK31 easier than ever.

PSK31 combines the advantages of simple variable length text code with a narrow bandwidth phase-shift-keying (PSK) signal using DSP techniques. The standard BPSK mode offers unconnected live keyboard-to-keyboard chat without Forward Error Correction, at a 31 baud rate. Most ASCII characters are supported, with faster sending speeds when using lower case letters. A second version having four (quad) phase shifts, called QPSK is also available.

The chief advantage of using PSK31 is its narrow bandwidth but this isn't achieved naturally. With digital phase modulation, the phase changes abruptly, and without additional measures, wide sidebands would be created.

M. Greenman, offers a good explaination of how this is counteracted, saying "To prevent this, these modes also include 100% raised-cosine amplitude modulation (ASK) at the symbol rate, which reduces the power to zero at the phase range. Because of this, the signal bandwidth is relatively narrow" (Greenman).

PSK31 is not an error free digital mode, although improved versions such as the PSKR varieties have been developed to improve on its lack of robustness under adverse conditions. PSKR uses a similar design as the MFSK modes, with a convolutional encoder and interleaver, however this comes at the expense of data speed (which is sometimes divided in half when compared to standard BPSK). The PSKR modes are designed for use with data transfer applications such as pskmail, flarq, or other automated-repeat-request applications.

According to Greenman, "the BPSK modes work well on a quiet, single-hop path, but give poor performance in most other conditions" (Greenman). BPSK31 is the default calling mode, although PSK63F (which does use a form of Forward Error Correction) is also well suited for keyboard-to-keyboard chat. The slower QPSK modes seem most affected by ionospheric doppler phase changes, although Differential PSK helps to maintain sync and reduce the effects of doppler, by allowing the receiver to measure phase difference from symbol to symbol.

A couple of excellent software programs for PSK31 are <u>FLDIGI</u> and <u>DigiPan</u>.

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WSJT Modes

- Suite of digital modes for HF, VHF, UHF used for weak signal communications and esoteric propagation modes.
- Originally developed by Joe Taylor, K1JT.
- Different modes for different propagation e.g:

Al Penney VO1NO

- FSK441 for Meteor Scatter;
- JT65 for EME; and
- FT8 for weak and fading paths.

WSJT is a computer program used for weak-signal radio communication between amateur radio operators. The program was initially written by Joe Taylor, K1JT, but is now open source and is developed by a small team. The digital signal processing techniques in WSJT make it substantially easier for amateur radio operators to employ esoteric propagation modes, such as high speed meteor scatter and moonbounce.

Communication modes provided

The software carries a general emphasis on weak-signal operation and advanced DSP techniques; however, the communication modes rely upon different ionospheric propagation modes and may be used on many different bands.

WSJT's communication modes can be divided into fast and slow modes. While fast modes send character-by-character without error correction, the slow modes aim to optimize for minimal QRO (highpower) use. As of WSJT10, supported fast modes are JTMS, FSK441, ISCAT, and JT6M, and the slow modes are JT65 and JT4. WSJT-X 1.8 additionally implements the "slow" JT9, FT8, and QRA64. Some modes have derived submodes with larger tone spacing. Two other modes, WSPR and Echo are included for measuring propagation and testing moon bounce echo.

FSK441

FSK441, introduced in 2001 as the first communications mode included with WSJT, is designed to support communication using streaks of radioreflecting ions created in the ionosphere by the trails of meteors entering the Earth's atmosphere. The bursts of signal created by such trails are commonly referred to as "pings", due to their characteristic sound. Such pings may be as short as a tenth of a second and carry enough information to complete at least one stage of a contact. FSK441 employs multi-frequency shift keying using four tones, at a data rate of 441 baud. Because of the choice of character codes in the protocol, it is self-synchronizing and does not require an explicit synchronization tone. FSK441 is generally used on the 2-meter and 70-centimeter amateur bands. Contacts may be made at almost any time (that is, a meteor shower is not required to be in progress) at distances of up to 1400 miles (2250 km).

When transmitted messages include at least one space, the FSK441 decoding algorithm uses that space character as a syncword for zero-overhead synchronization.

JT6M

JT6M, introduced in late 2002, is intended for meteor scatter and other ionospheric scattering of signals, and is especially optimized for the 6-meter band. The mode also employs multiple frequency-shift keying, but at 44 tones. One of the tones is a synchronization tone, leaving 43 tones to carry data (one tone per character in the character set, which includes alphanumerics and some punctuation). The symbol rate is 21.53 baud; the actual data rate as encoded for transmit is 14.4 characters per second. The mode is known for sounding "a bit like piccolo music".

JT65

JT65, developed and released in late 2003, is intended for extremely weak but slowly varying signals, such as those found on troposcatter or Earth-Moon-Earth (EME, or "moonbounce") paths. It can decode signals many decibels below the noise floor in a 2500 Hz band (note that SNR in a 2500 Hz band is approximately 28 dB lower than SNR in a 4 Hz band, which is closer to the channel bandwidth of an individual JT65 tone), and can often allow amateurs to successfully exchange contact information without signals being audible to the human ear. Like the other modes, multiple-frequency shift keying is employed; unlike the other modes, messages are transmitted as atomic units after being compressed and then encoded with a process

known as forward error correction (or "FEC"). The FEC adds redundancy to the data, such that all of a message may be successfully recovered even if some bits are not received by the receiver. (The particular code used for JT65 is Reed-Solomon.) Because of this FEC process, messages are either decoded correctly or not decoded at all, with very high probability. After messages are encoded, they are transmitted using MFSK with 65 tones.

Operators have also begun using the JT65 mode for contacts on the HF bands, often using QRP (very low transmit power); while the mode was not originally intended for such use, its popularity has resulted in several new features being added to WSJT in order to facilitate HF operation.

JT9

JT9, intended for MF and HF use, was introduced in an experimental version of WSJT, known as **WSJT-X**. It uses the same logical encoding as JT65, but modulates to a 9-FSK signal. With 1-minute transmission intervals, JT9 occupies less than 16 Hz bandwidth. JT9 also has versions designed for longer transmission intervals of 2 minutes, 5 minutes, 10 minutes or 30 minutes. These extended versions take increasingly less bandwidth and permit reception of even weaker signals.

FT8

Joe Taylor, K1JT, announced on June 29, 2017 the availability of a new mode in the WSJT-X software, FT8. FT8 stands for "Franke-Taylor design, 8-FSK modulation" and was created by Joe Taylor, K1JT and Steve Franke, K9AN. It is described as being designed for "multi-hop Es where signals may be weak and fading, openings may be short, and you want fast completion of reliable, confirmable QSO's".

According to Taylor, the important characteristics of FT8 are -

•T/R sequence length: 15 s

•Message length: 75 bits + 12-bit CRC

•FEC code: (174,87)LDPC

•Modulation: 8-FSK, keying rate = tone spacing = 6.25 Hz

•Waveform: Continuous phase, constant envelope

•Occupied bandwidth: 50 Hz

•Synchronization: three 7x7 Costas arrays (start, middle, end of transmission)

•Transmission duration: 79*1920/12000 = 12.64 s

•Decoding threshold: -20 dB (perhaps -24 dB with a priori decoding, TBD)

•Operational behavior: similar to HF usage of JT9, JT65

•Multi-decoder: finds and decodes all FT8 signals in passband

•Auto-sequencing after manual start of QSO

Compared to the so-called "slow modes" (JT9, JT65, QRA64), FT8 is a few decibels less sensitive, but allows completion of QSOs four times faster. Bandwidth is greater than JT9, but about one-quarter of JT65A and less than one-half of QRA64. Compared with the "fast modes" (JT9E-H), FT8 is significantly more sensitive, has much narrower bandwidth, uses the vertical waterfall, and offers multi-decoding over the full displayed passband. Features not yet implemented include signal subtraction, two-pass decoding, and use of *a priori* (already known) information as it accumulates during a QSO."

FT4

In 2019, Taylor, et al, introduced FT4, an experimental protocol which is similar to FT8 but has a shorter T/R sequence length for faster contest exchanges.

Band Activity						Frequency		
UTC di	B DT Freq	Message		UTC	dB DT Freq	Message		
185930 -10		WB8JUI WAMAD	^		14 0.1 1352 ~			-
185930 -6		N9YBK KOOY 73 KB6NU WA9THI RRR			-3 -0.8 1352 ~ 10 0.3 1204 ~			_
190000 9 190000 2		CQ WA6GXQ FM06	-		Tx 1204 ~	WA9THI KB6		_
190000 -9		WB8JUI WAMAD		185930	6 0.3 1204 ~		the second s	
190000 1	0.1 1545 ~	HB9LBC WA2CXA FN22			Tx 1204 ~			
190030 7	0.3 1204 ~	KB6NU WA9THI 73		190000	9 0.3 1204 ~	and the second s	and a second second	
190030 (190030 8		CQ WA6GXQ FM06 HB9LBC WA2CXA FN22	~	190015 190030	Tx 1204 ~ 7 0.3 1204 ~	WA9THI KB61 KB6NU WA9TH		
			-			1		
Log QSO	Stop	Monitor Erase		Decode	Enable Tx	Halt Tx	Tune	Menu:
[Dx Call	DX Grid Tx 1204 Hz 🜩	Tx ←	·Rx	Generate Sto WA9THI KB6NU EN82			
-80	WA9THI	EM69 Rx 1204 Hz 🜩	Rx +	Tx	WA9THI KB6NU +09) Tx 2	
-60	Az: 222	400 lm		Tx Freq	WA9THI KB6NU R+09		Tx 3	
-40	Lookup	Add Report 9 🗘			WA9THI KB6NU RRR	0	Tx 4	
-20	2017 N	Vov 23	Call 1st		WA9THI KB6NU 73	v (Tx 5	
57 dB	19:0	NA VHE Contest		CQ KB6NU EN82			Tx 6	
Receivin	c FT	Last Tx: WA9THI KB6NU	73				2/1	5 WD:6m
Receive	y Fi	Cast 1X: WA9THI KDONU	13				2/1	wD:6m

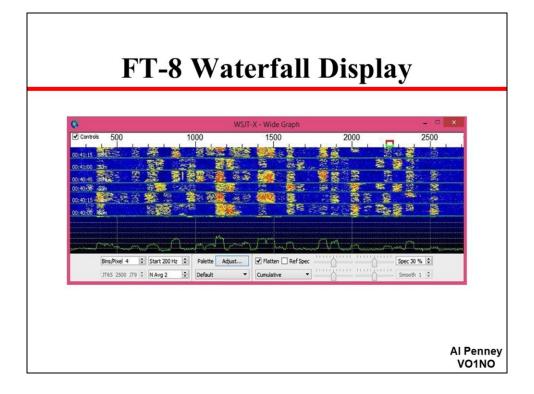
Joe Taylor, K1JT, announced on June 29, 2017 the availability of a new mode in the <u>WSJT-X</u> software, called FT8. FT8 stands for "Franke-Taylor design, 8-FSK modulation" and was created by Joe Taylor, K1JT and Steve Franke, K9AN. It is described as being designed for "multi-hop Es where signals may be weak and fading, openings may be short, and you want fast completion of reliable, confirmable QSO's".

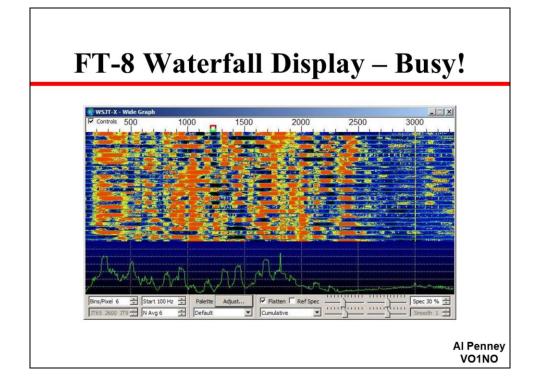
According to Taylor, the important characteristics of FT8 are:

- •T/R sequence length: 15 secs.
- •Message length: 75 bits + 12-bit CRC
- •FEC code: LDPC (174,87)
- Modulation: 8-FSK, keying rate = tone spacing = 5.86 Hz
- •Waveform: Continuous phase, constant envelope
- •Occupied bandwidth: 47 Hz
- •Synchronization: three 7x7 Costas arrays (start, middle, end of TX)
- •Transmission duration: 79*2048/12000 = 13.48 secs.
- Decoding threshold: -20 dB (perhaps -24 dB with a priori decoding, TBD)
- •Operational behavior: similar to HF usage of JT9, JT65

- Multi-decoder: finds and decodes all FT8 signals in passband
- •Auto-sequencing after manual start of QSO

Compared to the so called slow modes (JT9, JT65, QRA64), FT8 is a few dB less sensitive but allows completion of QSOs four times faster. Bandwidth is greater than JT9, but about 1/4 of JT65A and less than 1/2 QRA64. Compared with the fast modes (JT9E-H), FT8 is significantly more sensitive, has much smaller bandwidth, uses the vertical waterfall, and offers multi-decoding over the full displayed passband. Features not yet implemented include signal subtraction, two-pass decoding, and use of a priori (already known) information as it accumulates during a QSO.





Slow Scan TV (SSTV)

- Unlike commercial TV which requires up to 6 MHz of bandwidth, **SSTV** transmits pictures using the **same bandwidth as an SSB voice signal** (2.7 kHz).
- The cost is the rate at which pictures are transmitted – it takes 8 seconds/frame for the fastest mode, and up to 72 seconds/frame for more detailed, colour pictures.

Al Penney VO1NO

Slow Scan television (**SSTV**) is a picture transmission method used mainly by amateur radio operators, to transmit and receive static pictures via radio in monochrome or color.

A literal term for SSTV is narrowband television.

Analog broadcast television requires at least 6 MHz wide channels, because it transmits 25 or 30 picture frames per second (in the NTSC, PAL or SECAM color systems), but SSTV usually only takes up to a maximum of 3 kHz of bandwidth. It is a much slower method of still picture transmission, usually taking from about eight seconds to a couple of minutes, depending on the mode used, to transmit one image frame.

Since SSTV systems operate on voice frequencies, amateurs use it on shortwave (also known as HF by amateur radio operators), VHF and UHF radio.

Slow Scan Television (SSTV)

Slow Scan Television is a method of transmitting medium and high resolution images over the air using standard audio modulation. Most modern SSTV activities are computer based, and are relatively easy to set up. All that is required to join the fun is a radio, a sound card interface, and a PC with SSTV software.

The concept of Slow Scan Television dates back to 1957, when Copthorne MacDonald began experimenting with an electrostatic monitor and a vidicon tube. The FCC legalized SSTV for amateur radio use in 1968, but because of the specialized equipment needed, few took it up. At the time, operating the mode required a scanner or camera, a modem (to create audio tones), and a cathode ray tube with long persistence phosphors, (that would keep an image visible for ten seconds or longer). Many SSTV operators used tubes found in surplus radar equipment.

A Slow Scan Television transmission consists of horizontal lines, scanned left to right, with color components sent seperately one line after another. The color encoding and order of transmission can vary between modes. To create an image, an audio tone is frequency modulated and transmitted (usually as a single sideband signal). The audio signal has a frequency of 1200 Hz for a frame pulse, 1500 Hz for black, and up to 2300 Hz for peak white. Other signals, called synchronization pulses are represented by a frequency lower than the one representing the black level. These are said to be blacker than black, and therefore cannot be seen on the screen.

In the beginning, SSTV images were produced by an 8-second black and white transmission format, but it didn't take long for experimenters to get tired with black and white. Soon, they devised clever ways to send color images using the same equipment. The frame - sequential method involved sending the same picture three times, with a red, green, or blue colored filter placed in front of the camera lens.

The receiving station would take three long-exposure photographs of the screen, also placing red, green and blue filters in front of the film camera's lens at the correct time. John Langner (WB2OSZ) writes in the ARRL Handbook, that this method, "had some drawbacks, as any noise on the band could ruin the image registration (overlay of the frames), and spoil the picture."

Images could then be saved and simultaneously displayed on an ordinary color TV. (Langner). The Line-Sequential method remedied this by scanning each line three times, allowing pictures to be received in full color in real time. Early line-sequential modes, such as Wrasse SC-1, used a horizontal sync pulse for each of the color components, but a drawback to this method, according to Langner, "is that if the receiving end gets out of step, it won't know which scan represents which color" (Langner). Robot Research, in an attempt to solve these issues, moved away from the traditional RGB color model with their 1200C modes, using Luminance and Chrominance signals instead. With this method, the first part of each 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, (which is used to convey the color information of the picture). 1200C is efficient, allowing a 120 line image to be sent in about 12 seconds compared to the usual 24, but picture quality suffers, especially on images with sharp, high-contrast edges. One of the most important advantages of 1200C is its compatibility with older black and white equipment (Langner).

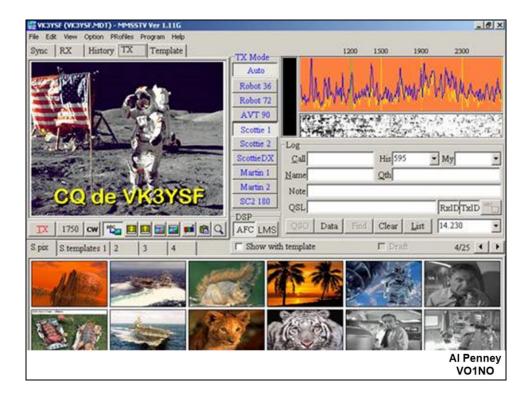
The Martin (M1) and Scottie (S1) modes, which are two of the most popular today, have returned to RGB encoding. Both use a single horizontal sync pulse for each set of RGB scans, and differ only in the timing.

Another innovation, introduced by Robot Research, is Vertical-Interval-Signalling, which is a way of encoding the transmission mode into the vertical sync signal. VIS is composed of a start bit, 7 data bits, an even parity bit, and a stop bit, each 30 ms long. To this day, every new transmission mode has adopted Robot's scheme and has assigned codes to their particular mode. With each mode having a unique VIS code, this allows software programs to automatically select the correct mode when set to automatic receive.

SSTV picture quality can vary widely depending on the mode, and the receiver's ability to detect synchronization pulses. There are a wide variety of standards for picture size. Typically, pictures are 128 lines long and take about eight seconds to send and resolution is around 320 X 240.

Where can I find SSTV signals?

80 METERS3.73540 METERS7.04020 METERS14.23015 METERS21.34010 METERS28.680



MMSSTV software

Fast Scan TV

- More frequently called ATV (Amateur TV).
- Uses the same **NTSC format** that regular (non-HD) TV uses.
- Because of the **bandwidth requirements**, it is **limited to the 70cm band and higher**.
- ATV Repeaters can be found in the 903 MHz and 1.2 GHz bands.
- Video signal is AM, while audio is FM.

Al Penney VO1NO

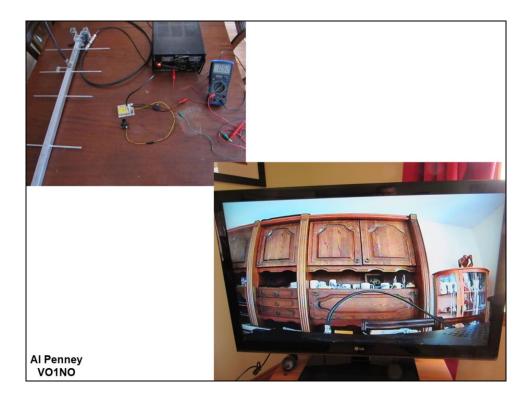
Amateur television (ATV) is the transmission of broadcast quality video and audio over the wide range of frequencies of radio waves allocated for radio amateur (Ham) use. ATV is used for noncommercial experimentation, pleasure, and public service events. Ham TV stations were on the air in many cities before commercial television stations came on the air. Various transmission standards are used, these include the broadcast transmission standards of NTSC in North America and Japan, and PAL or SECAM elsewhere, utilizing the full refresh rates of those standards. ATV includes the study of building of such transmitters and receivers, and the study of radio propagation of signals travelling between transmitting and receiving stations.

ATV is an extension of amateur radio. It is also called HAM TV or fastscan TV (FSTV), as opposed to slow-scan television (SSTV).

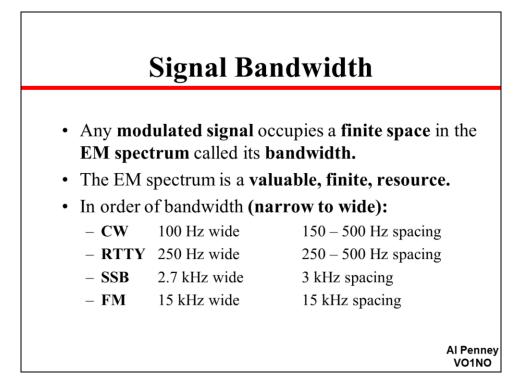
The 70-centimeter band (420-450 MHz) is the most commonly used ham band for ATV. Signals transmitted on this band usually propagate longer distances than on higher frequency bands, for a given transmitter power and antenna gain. The band falls between broadcast TV channels 13 and 14, which are 210–216 MHz and 470–476 MHz respectively. Propagation is similar to the lowest UHF TV Broadcast channels.

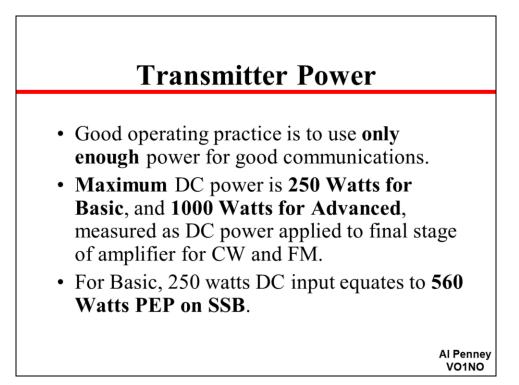
Additionally, this band can be easily received by simply tuning any cable-ready analog television or cable-box to the cable TV channels below and connecting an outdoor TV antenna. Amateur TV signals are much weaker than broadcast TV, so a preamplifier is often used to improve reception.

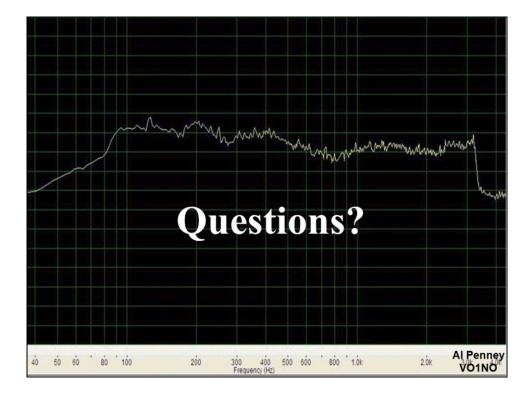




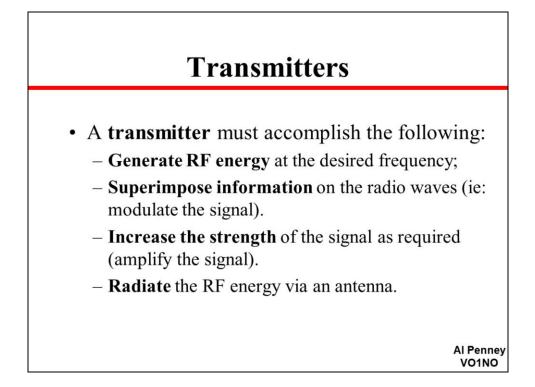
ATV system used for high altitude balloon tests. The little silver thing is the transmitter.











In electronics and telecommunications a **transmitter** or **radio transmitter** is an electronic device which produces radio waves with an antenna. The transmitter itself generates a radio frequency <u>alternating curren</u>t, which is applied to the antenna. When excited by this alternating current, the antenna radiates radio waves.

Transmitters are necessary component parts of all electronic devices that communicate by radio, such

as radio and television broadcasting stations, cell phones, walkietalkies, wireless computer networks, Bluetooth enabled devices, garage door openers, two-way radios in aircraft, ships, spacecraft, radar sets and navigational beacons. The term *transmitter* is usually limited to equipment that generates radio waves for communication purposes; or radiolocation, such as radar and navigational transmitters. Generators of radio waves for heating or industrial purposes, such as microwave ovens or diathermy equipment, are not usually called transmitters, even though they often have similar circuits.

The term is popularly used more specifically to refer to a broadcast transmitter, a transmitter used in broadcasting, as in *FM radio transmitter* or <u>television transmitter</u>. This usage typically includes both the transmitter proper, the antenna, and often the building it is housed in.

A transmitter can be a separate piece of electronic equipment, or an electrical circuit within another electronic device. A transmitter and a receiver combined in one unit is called a transceiver. The term transmitter is often abbreviated "XMTR" or "TX" in technical documents. The purpose of most transmitters is radio communication of information over a distance. The information is provided to the transmitter in the form of an electronic signal, such as an audio (sound) signal from a microphone, a video (TV) signal from a video camera, or in wireless networking devices, a digital signal from a computer. The transmitter combines the information signal to be carried with the radio frequency signal which generates the radio waves, which is called the carrier signal. This process is called *modulation*. The information can be added to the carrier in several different ways, in different types of transmitters. In an amplitude modulation (AM) transmitter, the information is added to the radio signal by varying its amplitude. In a frequency modulation (FM) transmitter, it is added by varying the radio signal's frequency slightly. Many other types of modulation are also used.

The radio signal from the transmitter is applied to the antenna, which radiates the energy as radio waves. The antenna may be enclosed inside the case or attached to the outside of the transmitter, as in portable devices such as cell phones, walkie-talkies, and garage door openers. In more powerful transmitters, the antenna may be located on top of a building or on a separate tower, and connected to the transmitter by a feed line, that is a transmission line.

Components

A practical radio transmitter mainly consists of the following parts:

•In high power transmitters, a power supply circuit to transform the input electrical power to the higher voltages needed to produce the required power output.

•An electronic oscillator circuit to generate the radio frequency signal. This usually generates a sine wave of constant amplitude called the carrier wave, because it serves to "carry" the information through space. In most modern transmitters, this is a crystal oscillator in which the frequency is precisely controlled by the vibrations of a quartz crystal. The frequency of the carrier wave is considered the frequency of the transmitter.

•A modulator circuit to add the information to be transmitted to the carrier wave produced by the oscillator. This is done by varying some aspect of the carrier wave. The information is provided to the transmitter as an electronic signal

called the modulation signal. The modulation signal may be an audio signal, which represents sound, a video signal which represents moving images, or for data in the form of a binary digital signal which represents a sequence of bits, a bitstream. Different types of transmitters use different modulation methods to transmit information:

•In an AM (amplitude modulation) transmitter the amplitude (strength) of the carrier wave is varied in proportion to the modulation signal.

•In an FM (frequency modulation) transmitter the frequency of the carrier is varied by the modulation signal.

•In an FSK (frequency-shift keying) transmitter, which transmits digital data, the frequency of the carrier is shifted between two frequencies which represent the two binary digits, 0 and 1.

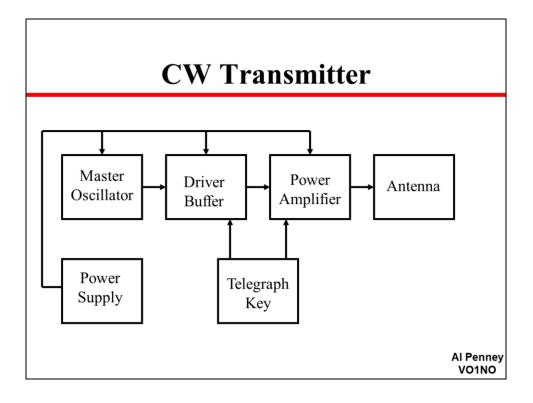
•OFDM (orthogonal frequency division multiplexing) is a family of complicated digital modulation methods very widely used in high bandwidth systems such as WiFi networks, cellphones, digital television broadcasting, and digital audio broadcasting (DAB) to transmit digital data using a minimum of radio spectrum bandwidth. OFDM has higher spectral efficiency and more resistance to fading than AM or FM. In OFDM multiple radio carrier waves closely spaced in frequency are transmitted within the radio channel, with each carrier modulated with bits from the incoming bitstream so multiple bits are being sent simultaneously, in parallel. At the receiver the carriers are demodulated and the bits are combined in the proper order into one bitstream.

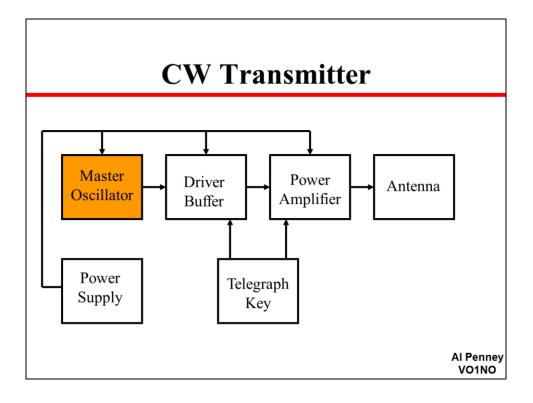
Many other types of modulation are also used. In large transmitters the oscillator and modulator together are often referred to as the *exciter*.

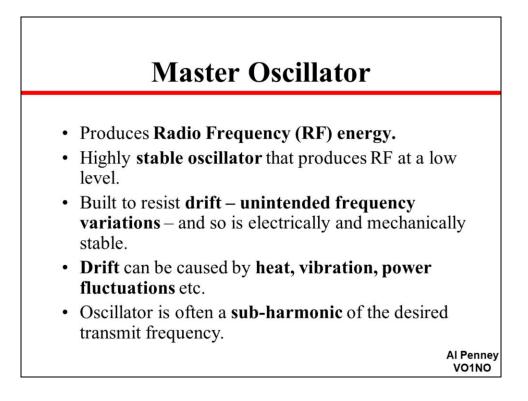
•A radio frequency (RF) amplifier to increase the power of the signal, to increase the range of the radio waves.

•An impedance matching (antenna tuner) circuit to match the impedance of the transmitter to the impedance of the antenna (or the transmission line to the antenna), to transfer power efficiently to the antenna. If these impedances are not equal, it causes a condition called standing waves, in which the power is reflected back from the antenna toward the transmitter, wasting power and sometimes overheating the transmitter.

In higher frequency transmitters, in the UHF and microwave range, free running oscillators are unstable at the output frequency. Older designs used an oscillator at a lower frequency, which was multiplied by frequency multipliers to get a signal at the desired frequency. Modern designs more commonly use an oscillator at the operating frequency which is stabilized by phase locking to a very stable lower frequency reference, usually a crystal oscillator.

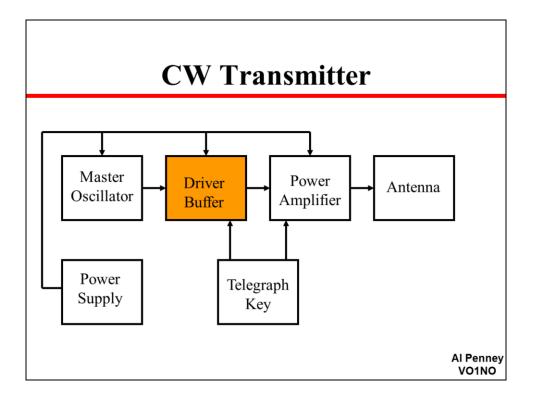


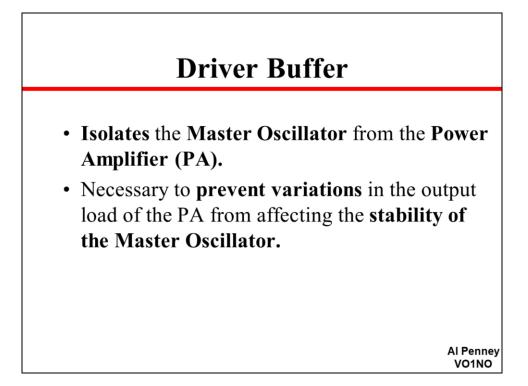




The carrier originates in the **master oscillator** stage where it is generated as a constant-amplitude, constant-frequency sine wave. The carrier is not of sufficient amplitude and must be amplified in one or more stages before it attains the high power required by the antenna.

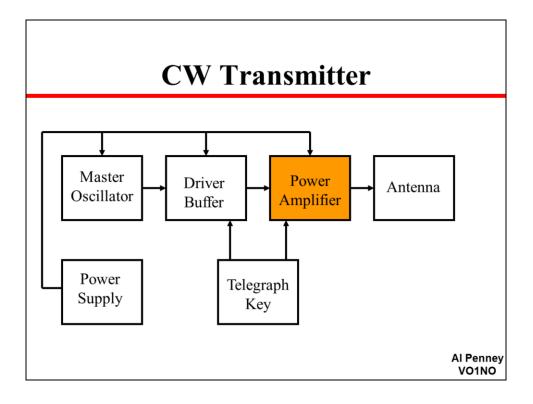
The oscillator generates the rf carrier at a preset frequency and maintains it within close tolerances. The oscillator may be a self-excited type, such as an electron-coupled oscillator, or a quartz crystal type,which uses a crystal cut to vibrate at a certain frequency when electrically excited. In both types, voltageand current delivered by the oscillator are weak. The oscillator outputs must be amplified many times tobe radiated any distance.

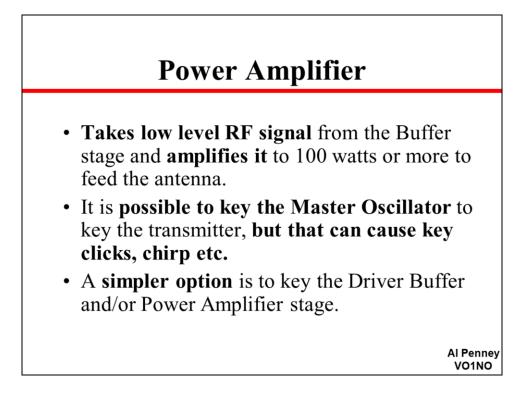




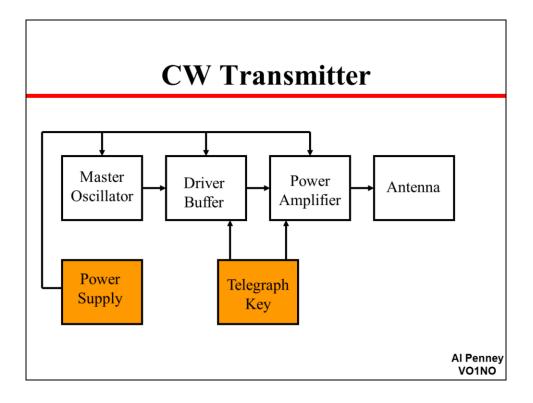
A **buffer amplifier** (sometimes simply called a **buffer**) is one that provides electrical impedance transformation from one circuit to another, with the aim of preventing the signal source from being affected by whatever currents (or voltages, for a current buffer) that the load may be produced with. The signal is 'buffered from' load currents.

The buffer serves two purposes. One is to isolate the oscillator from the amplifier stages. Without a buffer, changes in the amplifier caused by keying or variations in source voltage would vary the load of the oscillator and cause it to change frequency.





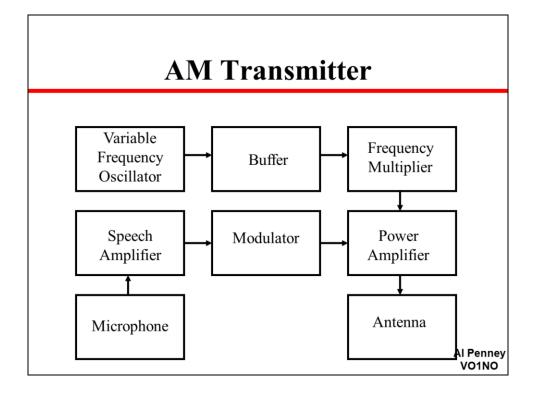
A radio frequency power amplifier (RF power amplifier) is a type of electronic amplifier that converts a low-power radiofrequency signal into a higher power signal. Typically, RF power amplifiers drive the antenna of a transmitter. Design goals often include gain, power output, bandwidth, power efficiency, linearity (low signal compression at rated output), input and output impedance matching, and heat dissipation.

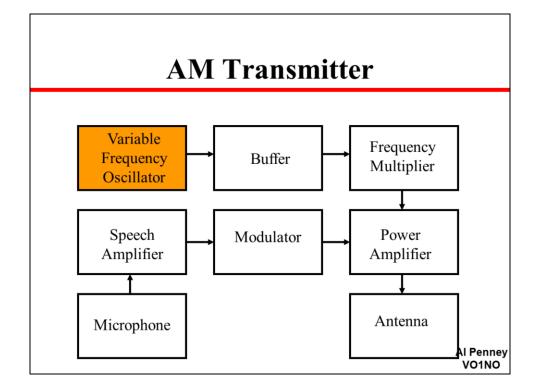


Key and Power Supply

- The **Key** is essentially an **on/off switch** that controls when the **RF energy** is applied to the antenna.
- The **Power Supply** provides the **voltages required** by the transmitter. For solid state equipment this is usually **13.8 VDC**, though some radios need 24 VDC for the Power FETs that make up the PA. Tube gear needs **6.3 VAC** for the **filaments** and various **high DC voltages** for the **plates** and **screens**.

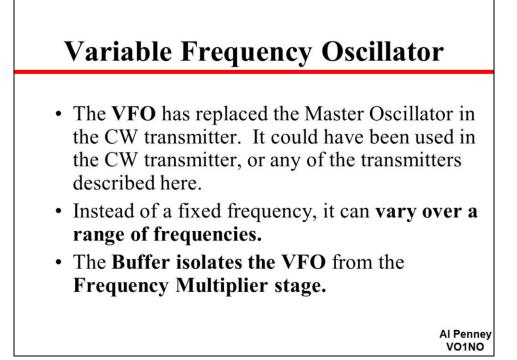
Al Penney VO1NO





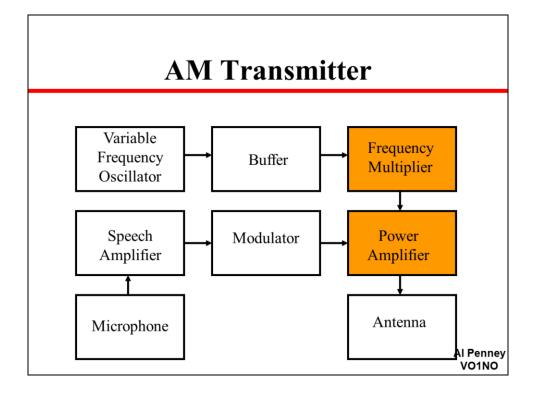
A variable frequency oscillator (VFO) in electronics is

an oscillator whose frequency can be tuned (i.e., varied) over some range. It is a necessary component in any tunable radio receiver or transmitter that works by the superheterodyne principle, and controls the frequency to which the apparatus is tuned.



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an oscillator whose frequency can be tuned (i.e., varied) over some range. It is a necessary component in any tunable radio receiver or transmitter that works by the superheterodyne principle, and controls the frequency to which the apparatus is tuned.



Frequency Multiplier and PA

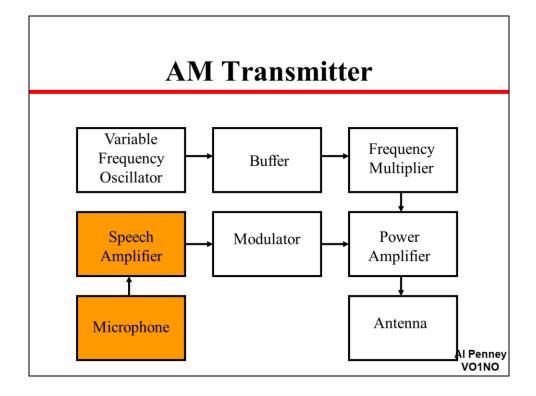
- The **Frequency Multiplier** multiplies the frequency generated by the VFO to bring it to the desired frequency.
- The **PA amplifies** the RF signal delivered by the Frequency Multiplier, but has other duties as well in an AM transmitter.

Al Penney VO1NO

In electronics, a **frequency multiplier** is an electronic circuit that generates an output signal whose output frequency is a harmonic (multiple) of its input frequency. Frequency multipliers consist of a nonlinear circuit that distorts the input signal and consequently generates harmonics of the input signal. A subsequent bandpass filter selects the desired harmonic frequency and removes the unwanted fundamental and other harmonics from the output.

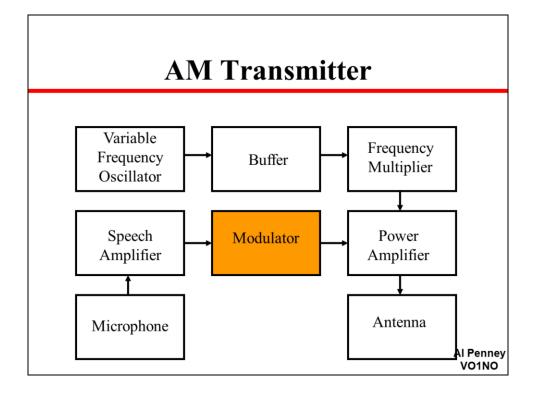
Frequency multipliers are often used in frequency synthesizers and communications circuits. It can be more economical to develop a lower frequency signal with lower power and less expensive devices, and then use a frequency multiplier chain to generate an output frequency in the microwave or millimeter wave range. Some modulation schemes, such as frequency modulation, survive the nonlinear distortion without ill effect (but schemes such as amplitude

modulation do not).



Microphone and Speech Amplifier

- The **Microphone** produces a **low output voltage**, generally in the range of a **few tens of millivolts**.
- It is **amplified** by the **Speech Amplifier** to the level **required by the modulator**.
- In general, the audio for all Amateur AM, SSB and FM transmitters is processed so that the modulated power is contained in the most useful region, typically 300 to 3000 Hz. (Note that there may be a typo in your book.)



Modulator

- The Modulator changes the amplitude of the RF signal to vary in accordance with the speech characteristics.
- The output of the **Frequency Multiplier** stage and the **Modulator** are **combined in the PA** to create the **final signal**, which is delivered to the **antenna** and **radiated**.

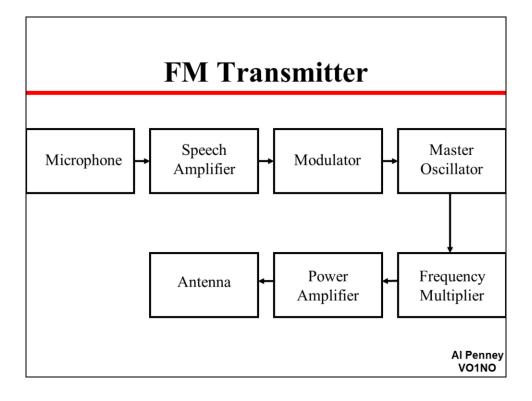
Al Penney VO1NO

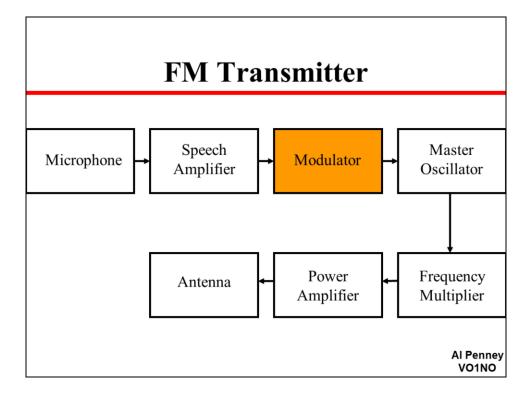
High and low level AM modulators

AM modulators may be classed as either high or low level dependent upon their level in the overall signal chain.

•*High level modulator:* A high level modulator is defined as one that modulates a high power section of the circuit, typically the final RF amplifier. It has the advantage that linear amplifiers are not required for the RF amplification stages after AM modulation has been applied. The drawback is that high power audio amplifiers are needed. For broadcast transmitters where very high power levels are used, class D or class E amplifiers may be employed for the audio output.

•Low level modulator: A low level AM modulator would be one where the modulation is applied to low power stage of the transmitter, typically in the RF generation stages, or via the digital signal processing areas. The drawback of this approach is that linear amplification is required for the RF stages.





FM Modulator

- Sometimes called a **Reactance Modulator**, the **output from the Modulator** is applied to the **input of the Master Oscillator** to **vary its frequency**.
- The **amount of frequency variation** is generally **small**, so the Master Oscillator is generally operated at a fraction (say 1/8th) of the desired frequency.

Al Penney VO1NO

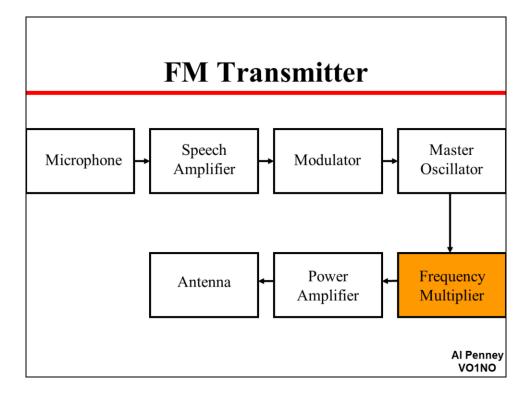
Modulation

FM signals can be generated using either direct or indirect frequency modulation:

•Direct FM modulation can be achieved by directly feeding the message into the input of a voltage-controlled oscillator.

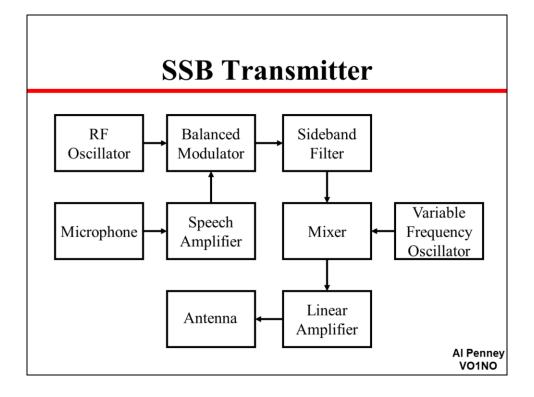
•For indirect FM modulation, the message signal is integrated to generate a phase-modulated signal. This is used to modulate a crystalcontrolled oscillator, and the result is passed through a frequency multiplier to produce an FM signal. In this modulation, narrowband FM is generated leading to wideband FM later and hence the modulation is known as indirect FM modulation.

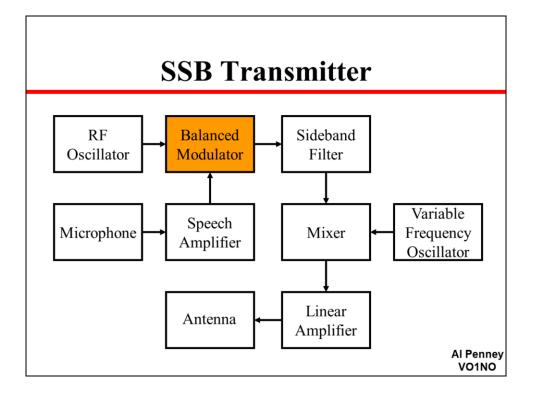
A reactance modulator changes the frequency of the tank circuit of the oscillator by changing its reactance.

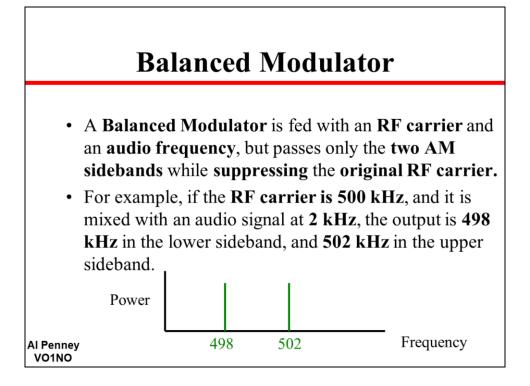


Frequency Multiplier

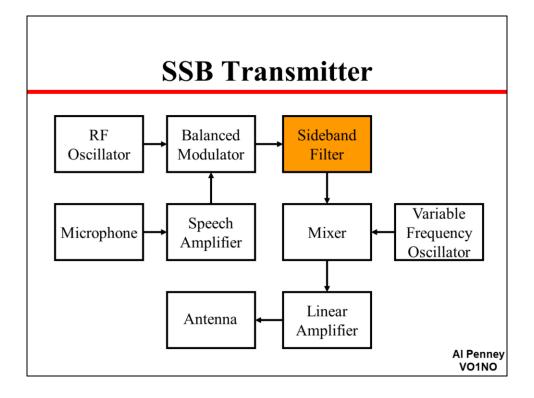
- The **output** from the **Master Oscillator** is **doubled and tripled** as required to **produce RF** in the **desired range**.
- This stage also acts as a **buffer** for the **Master Oscillator**.

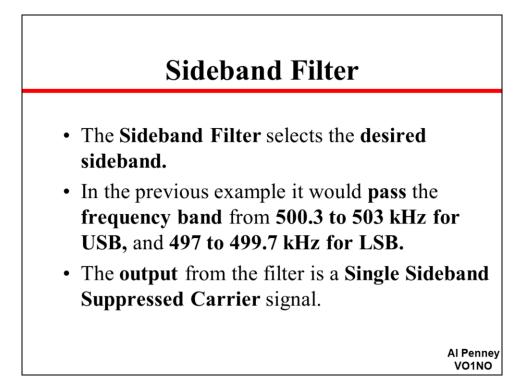




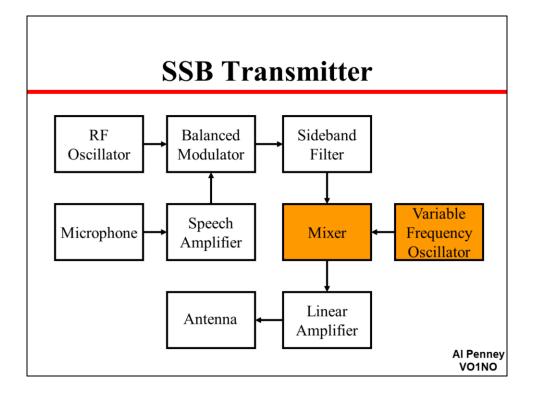


A balanced modulator mixes the audio signal and the radio frequency carrier, but suppresses the carrier, leaving only the sidebands. The output from the balanced modulator is a double sideband suppressed carrier signal and it contains all the information that the AM signal has, but without the carrier.





One method of producing an SSB signal is to remove one of the sidebands via filtering, leaving only either the **upper sideband** (**USB**), the sideband with the higher frequency, or less commonly the **lower sideband** (**LSB**), the sideband with the lower frequency. Most often, the carrier is reduced or removed entirely (suppressed), being referred to in full as **single sideband suppressed carrier** (**SSBSC**). Assuming both sidebands are symmetric, which is the case for a normal AM signal, no information is lost in the process. Since the final RF amplification is now concentrated in a single sideband, the effective power output is greater than in normal AM (the carrier and redundant sideband account for well over half of the power output of an AM transmitter). Though SSB uses substantially less bandwidth and power, it cannot be demodulated by a simple envelope detector like standard AM.



VFO and Mixer

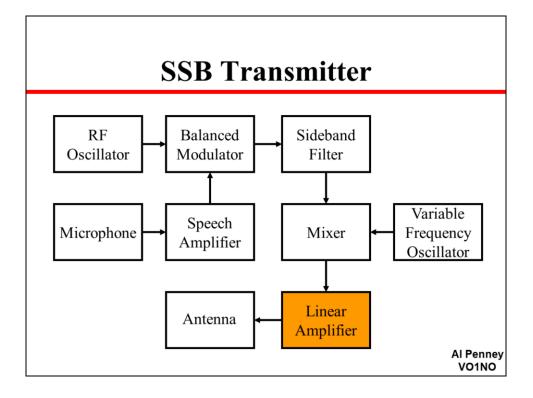
- The Mixer will mix the inputs from the Sideband Filter and the VFO to generate the desired transmit frequency.
- The Mixer will generate the sum and difference of the two frequencies, and the desired signal is selected while the other is filtered out.

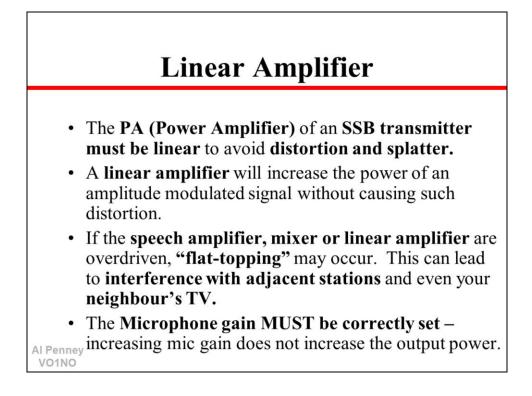
Mixing to Generate Transmit Signal

• If the **output** of the Sideband Filter stage is centered on **500 kHz (0.5 MHz)**, and we wish to operate in the 40M band (7.0 to 7.3 MHz), then we will need a **VFO** operating over the range of **6.5 to 6.8 MHz**.

6.5 MHz + 0.5 MHz = 7.0 MHz 6.8 MHz + 0.5 MHz = 7.3 MHz

• Note that **frequency multiplication cannot** be used to generate the desired transmit frequency as **passing an SSB signal** through a **frequency multiplier** would cause **severe distortion**.

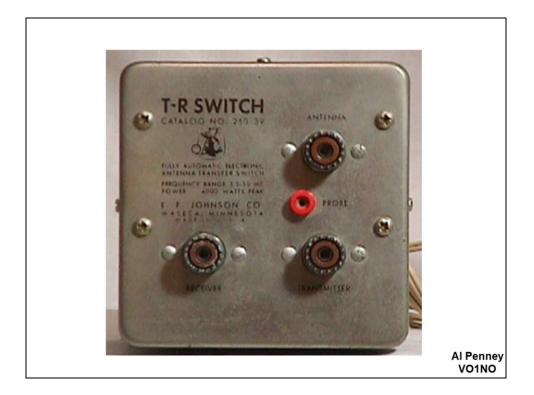




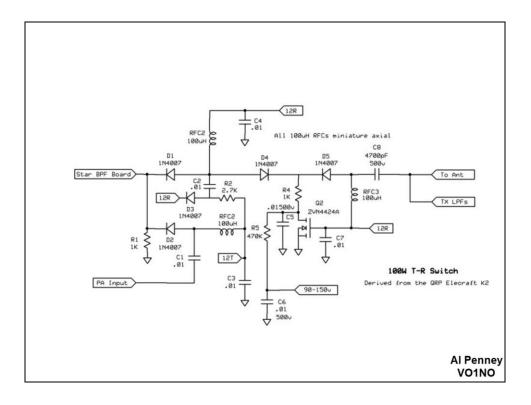
A **linear amplifier** is an electronic circuit whose output is proportional to its input, but capable of delivering more power into a load. The term usually refers to a type of radio-frequency (RF) power amplifier, some of which have output power measured in kilowatts, and are used in amateur radio.

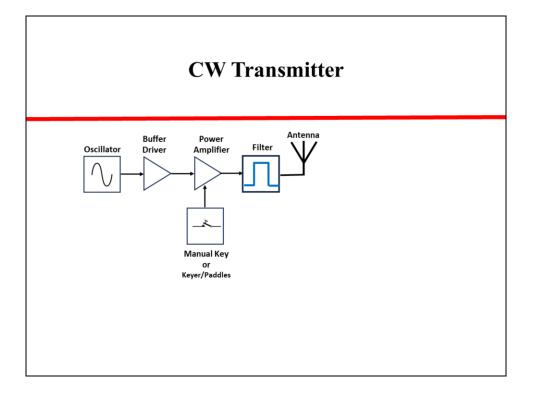
Transmit / Receive Switch

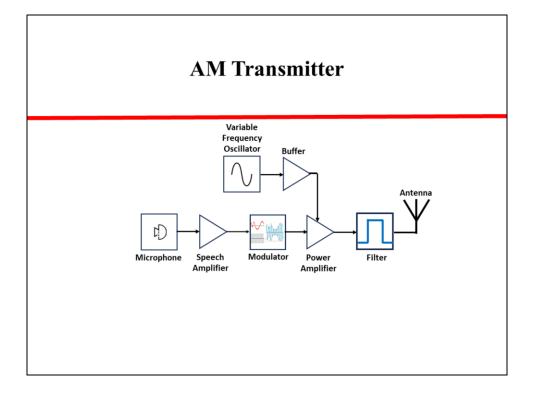
- The antenna is usually common to both the transmitter and receiver, so we need a method to keep the high power from the transmitter out of the sensitive input of the receiver.
- T/R Switch can be either mechanical (relay) or electronic (switching diodes).
- It switches the antenna between the TXmtr and RXer, and may also ground the RX input on transmitting (Mute), and turn the TXmtr off when receiving.
- Usually built-into modern transceivers.

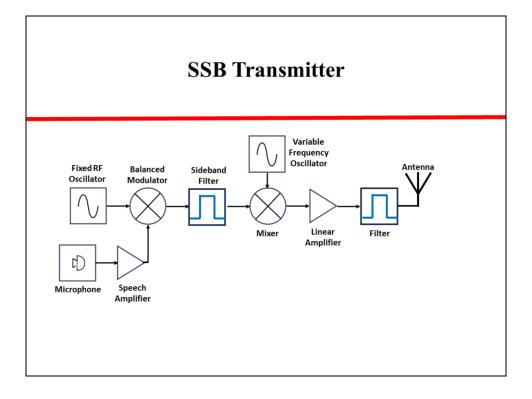


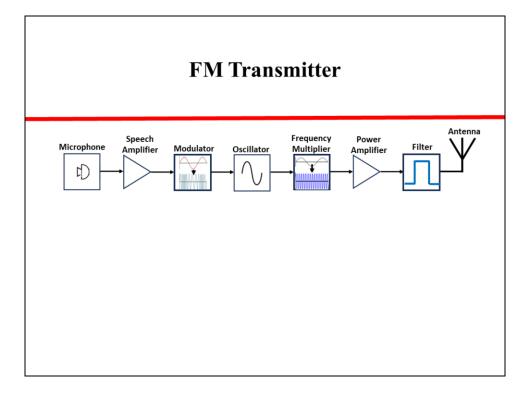
Old-style T/R switch used for separate transmitter / receiver setups.

















In a frequency modulation transmitter, the input to the speech amplifier is connected to the:

- modulator
- power amplifier
- frequency multiplier
- microphone

In a frequency modulation transmitter, the input to the speech amplifier is connected to the:

- modulator
- power amplifier
- frequency multiplier
- microphone
- < microphone >

In a frequency modulation transmitter, the microphone is connected to the:

- power amplifier
- oscillator
- speech amplifier
- modulator

In a frequency modulation transmitter, the microphone is connected to the:

- power amplifier
- oscillator
- speech amplifier
- modulator
- < speech amplifier >

In a frequency modulation transmitter, the is in between the speech amplifier and the oscillator.

- modulator
- power amplifier
- microphone
- frequency multiplier

In a frequency modulation transmitter, the is in between the speech amplifier and the oscillator.

- modulator
- power amplifier
- microphone
- frequency multiplier
- < modulator >

In a frequency modulation transmitter, the is located between the modulator and the frequency multiplier.

- power amplifier
- microphone
- oscillator
- speech amplifier

In a frequency modulation transmitter, the is located between the modulator and the frequency multiplier.

- power amplifier
- microphone
- oscillator
- speech amplifier
- < oscillator >

In a frequency modulation transmitter, the is located between the oscillator and the power amplifier.

- modulator
- frequency multiplier
- microphone
- speech amplifier

In a frequency modulation transmitter, the is located between the oscillator and the power amplifier.

- modulator
- frequency multiplier
- microphone
- speech amplifier
- < frequency multiplier >

In a frequency modulation transmitter, the is located between the frequency multiplier and the antenna.

- modulator
- speech amplifier
- oscillator
- power amplifier

In a frequency modulation transmitter, the is located between the frequency multiplier and the antenna.

- modulator
- speech amplifier
- oscillator
- power amplifier
- < power amplifier >

In a frequency modulation transmitter, the power amplifier output is connected to the:

- frequency multiplier
- microphone
- modulator
- antenna

In a frequency modulation transmitter, the power amplifier output is connected to the:

- frequency multiplier
- microphone
- modulator
- antenna
- < antenna >

In a CW transmitter, the output from the is connected to the driver/buffer.

- telegraph key
- power supply
- master oscillator
- power amplifier

In a CW transmitter, the output from the is connected to the driver/buffer.

- telegraph key
- power supply
- master oscillator
- power amplifier
- < master oscillator >

What amount of transmitter power should radio amateurs use at all times?

- The minimum legal power necessary to communicate
- 25 watts PEP output
- 250 watts PEP output
- 2000 watts PEP output

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- 25 watts PEP output
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- 2000 watts PEP output
- < The minimum legal power necessary to communicate >

The maximum percentage of modulation permitted in the use of radiotelephony by an amateur station is:

- 100 percent
- 90 percent
- 75 percent
- 50 percent

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- 100 percent
- 90 percent
- 75 percent
- 50 percent
- < 100 percent >

The DC power input to the anode or collector circuit of the final RF stage of a transmitter, used by a holder of an Amateur Radio Operator Certificate with Advanced Qualification, shall not exceed:

- 250 watts
- 500 watts
- 750 watts
- 1000 watts

The DC power input to the anode or collector circuit of the final RF stage of a transmitter, used by a holder of an Amateur Radio Operator Certificate with Advanced Qualification, shall not exceed:

- 250 watts
- 500 watts
- 750 watts
- 1000 watts
- < 1000 watts >

The operator of an amateur station, who is the holder of a Basic Qualification, shall ensure that the station power, when expressed as RF output power measured across an impedance matched load, does not exceed:

• 560 watts peak-envelope power, for transmitters producing any type of single sideband emission

• 2500 watts peak power

• 1000 watts carrier power for transmitters producing other emissions

• 150 watts peak power

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What is the maximum transmitting output power an amateur station may use on 3750 kHz if the operator has Basic and Morse code qualifications?

- 560 watts PEP output for SSB operation
- 1000 watts PEP output for SSB operation
- 1500 watts PEP output for SSB operation
- 2000 watts PEP output for SSB operation

What is the maximum transmitting output power an amateur station may use on 3750 kHz if the operator has Basic and Morse code qualifications?

- 560 watts PEP output for SSB operation
- 1000 watts PEP output for SSB operation
- 1500 watts PEP output for SSB operation
- 2000 watts PEP output for SSB operation
- < 560 watts PEP output for SSB operation >

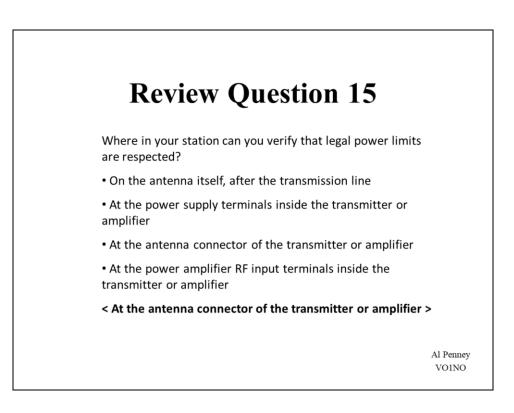
What is the most FM transmitter power a holder of only Basic Qualification may use on 147 MHz?

- 1000 watts DC input
- 200 watts PEP output
- 250 W DC input
- 25 watts PEP output

What is the most FM transmitter power a holder of only Basic Qualification may use on 147 MHz?

- 1000 watts DC input
- 200 watts PEP output
- 250 W DC input
- 25 watts PEP output
- < 250 W DC input >

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In S13.15 you will find the classic procedure for measuring the power output of a transmitter. However standards change and there is a move towards accepting power measurements at the antenna terminals of the transmitter. This is why many SWR meters, which are placed in the transmission path from transmitter to antenna, have a dual function and can serve as a power meter with the flick of a switch.

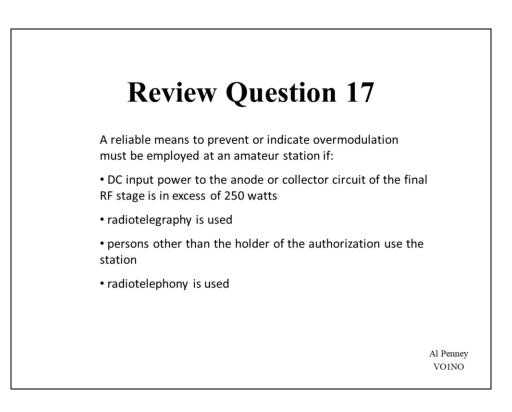
When operating on frequencies below 148 MHz:

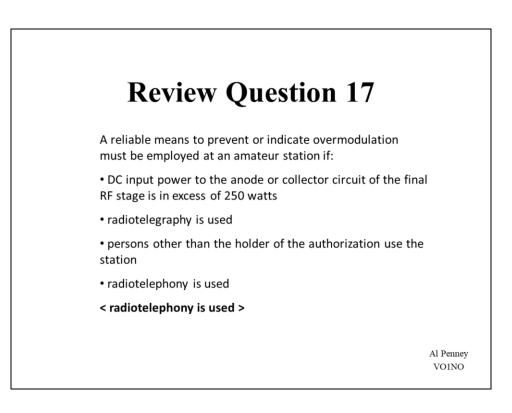
- an overmodulation indicator must be used
- the frequency stability must be comparable to crystal control
- the bandwidth for any emission must not exceed 3 kHz
- the frequency stability of the transmitter must be at least two parts per million over a period of one hour

When operating on frequencies below 148 MHz:

- an overmodulation indicator must be used
- the frequency stability must be comparable to crystal control
- the bandwidth for any emission must not exceed 3 kHz
- the frequency stability of the transmitter must be at least two parts per million over a period of one hour

< the frequency stability must be comparable to crystal control >





An amateur station using radiotelephony must install a device for indicating or preventing:

- overmodulation
- resonance
- antenna power
- plate voltage

An amateur station using radiotelephony must install a device for indicating or preventing:

- overmodulation
- resonance
- antenna power
- plate voltage
- < overmodulation >

If you contact another station and your signal is extremely strong and perfectly readable, what adjustment might you make to your transmitter?

- Turn on your speech processor
- Reduce your SWR
- Continue with your contact, making no changes
- Turn down your power output to the minimum necessary

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- Turn on your speech processor
- Reduce your SWR

>

- Continue with your contact, making no changes
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- < Turn down your power output to the minimum necessary
 - Al Penney
 - VOINO

In a typical CW transmitter, the is the primary source of direct current.

- master oscillator
- power supply
- driver/buffer
- power amplifier

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- power supply
- driver/buffer
- power amplifier
- < power supply >

In a CW transmitter, the is between the master oscillator and the power amplifier.

- driver/buffer
- audio amplifier
- power supply
- telegraph key

In a CW transmitter, the is between the master oscillator and the power amplifier.

- driver/buffer
- audio amplifier
- power supply
- telegraph key
- < driver/buffer >

In a CW transmitter, the is in between the driver/buffer stage and the antenna.

- power amplifier
- power supply
- telegraph key
- master oscillator

In a CW transmitter, the is in between the driver/buffer stage and the antenna.

- power amplifier
- power supply
- telegraph key
- master oscillator
- < power amplifier >

In a CW transmitter, the output of the is transferred to the antenna.

- driver/buffer
- power supply
- master oscillator
- power amplifier

In a CW transmitter, the output of the is transferred to the antenna.

- driver/buffer
- power supply
- master oscillator
- power amplifier
- < power amplifier >

In a single sideband transmitter, the output of the is connected to the balanced modulator.

- mixer
- radio frequency oscillator
- variable frequency oscillator
- linear amplifier

In a single sideband transmitter, the output of the is connected to the balanced modulator.

- mixer
- radio frequency oscillator
- variable frequency oscillator
- linear amplifier
- < radio frequency oscillator >

In a single sideband transmitter, the output of the is connected to the filter.

- mixer
- radio frequency oscillator
- balanced modulator
- microphone

In a single sideband transmitter, the output of the is connected to the filter.

- mixer
- radio frequency oscillator
- balanced modulator
- microphone
- < balanced modulator >

In a single sideband transmitter, the is connected to the speech amplifier.

- radio frequency oscillator
- filter
- mixer
- microphone

In a single sideband transmitter, the is connected to the speech amplifier.

- radio frequency oscillator
- filter
- mixer
- microphone
- < microphone >

In a single sideband transmitter, the output of the is connected to the balanced modulator.

- filter
- variable frequency oscillator
- linear amplifier
- speech amplifier

In a single sideband transmitter, the output of the is connected to the balanced modulator.

- filter
- variable frequency oscillator
- linear amplifier
- speech amplifier
- < speech amplifier >

In a single sideband transmitter, the output of the variable frequency oscillator is connected to the .

- balanced modulator
- linear amplifier
- mixer
- antenna

In a single sideband transmitter, the output of the variable frequency oscillator is connected to the .

- balanced modulator
- linear amplifier
- mixer
- antenna
- < mixer >

In a single sideband transmitter, the output of the is connected to the mixer.

- linear amplifier
- antenna
- variable frequency oscillator
- radio frequency oscillator

In a single sideband transmitter, the output of the is connected to the mixer.

- linear amplifier
- antenna
- variable frequency oscillator
- radio frequency oscillator
- < variable frequency oscillator >

In a single sideband transmitter, the is in between the mixer and the antenna.

- variable frequency oscillator
- balanced modulator
- radio frequency oscillator
- linear amplifier

In a single sideband transmitter, the is in between the mixer and the antenna.

- variable frequency oscillator
- balanced modulator
- radio frequency oscillator
- linear amplifier
- < linear amplifier >

In a single sideband transmitter, the output of the linear amplifier is connected to the:

- speech amplifier
- antenna
- filter
- variable frequency oscillator

In a single sideband transmitter, the output of the linear amplifier is connected to the:

- speech amplifier
- antenna
- filter
- variable frequency oscillator
- < antenna >

Which list of emission types is in order from the narrowest bandwidth to the widest bandwidth?

- CW, RTTY, SSB voice, FM voice
- CW, SSB voice, RTTY, FM voice
- CW, FM voice, RTTY, SSB voice
- RTTY, CW, SSB voice, FM voice

Which list of emission types is in order from the narrowest bandwidth to the widest bandwidth?

- CW, RTTY, SSB voice, FM voice
- CW, SSB voice, RTTY, FM voice
- CW, FM voice, RTTY, SSB voice
- RTTY, CW, SSB voice, FM voice
- < CW, RTTY, SSB voice, FM voice >

What circuit has a variable-frequency oscillator connected to a driver and a power amplifier?

- A crystal-controlled AM transmitter
- A single-sideband transmitter
- A digital radio transmitter
- A VFO-controlled CW transmitter

What circuit has a variable-frequency oscillator connected to a driver and a power amplifier?

- A crystal-controlled AM transmitter
- A single-sideband transmitter
- A digital radio transmitter
- A VFO-controlled CW transmitter
- < A VFO-controlled CW transmitter >

What type of modulation system changes the amplitude of an RF wave for the purpose of conveying information?

- Frequency modulation
- Amplitude modulation
- Phase modulation
- Amplitude-rectification modulation

What type of modulation system changes the amplitude of an RF wave for the purpose of conveying information?

- Frequency modulation
- Amplitude modulation
- Phase modulation
- Amplitude-rectification modulation
- < Amplitude modulation >

Morse code is usually transmitted by radio as:

- an interrupted carrier
- a series of key-clicks
- a continuous carrier
- · a voice-modulated carrier

Morse code is usually transmitted by radio as:

- an interrupted carrier
- a series of key-clicks
- a continuous carrier
- a voice-modulated carrier
- < an interrupted carrier >

An RF oscillator should be electrically and mechanically stable. This is to ensure that the oscillator does not:

- cause undue distortion
- drift in frequency
- become over modulated
- generate key-clicks

An RF oscillator should be electrically and mechanically stable. This is to ensure that the oscillator does not:

- cause undue distortion
- drift in frequency
- become over modulated
- generate key-clicks
- < drift in frequency >

What is the term for the average power supplied to an antenna transmission line during one RF cycle, at the crest of the modulation envelope?

- Average radio-frequency power
- Peak transmitter power
- Peak envelope power
- Peak output power

What is the term for the average power supplied to an antenna transmission line during one RF cycle, at the crest of the modulation envelope?

- Average radio-frequency power
- Peak transmitter power
- Peak envelope power
- Peak output power
- < Peak envelope power >

What is the usual bandwidth of a single sideband amateur signal?

- Between 2 and 3 kHz
- 1 kHz
- 2 kHz
- Between 3 and 6 kHz

What is the usual bandwidth of a single sideband amateur signal?

- Between 2 and 3 kHz
- 1 kHz
- 2 kHz
- Between 3 and 6 kHz
- < Between 2 and 3 kHz >

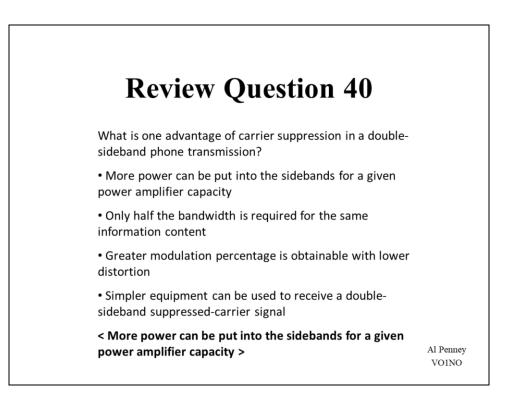
In a typical single-sideband phone transmitter, what circuit processes signals from the balanced modulator and sends signals to the mixer?

- RF amplifier
- Carrier oscillator
- Filter
- IF amplifier

In a typical single-sideband phone transmitter, what circuit processes signals from the balanced modulator and sends signals to the mixer?

- RF amplifier
- Carrier oscillator
- Filter
- IF amplifier
- < Filter >

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What happens to the signal of an over modulated singlesideband or double-sideband phone transmitter?

- It has higher fidelity and improved signal-to-noise ratio
- It becomes distorted and occupies more bandwidth
- It becomes stronger with no other effects

• It occupies less bandwidth with poor high-frequency response

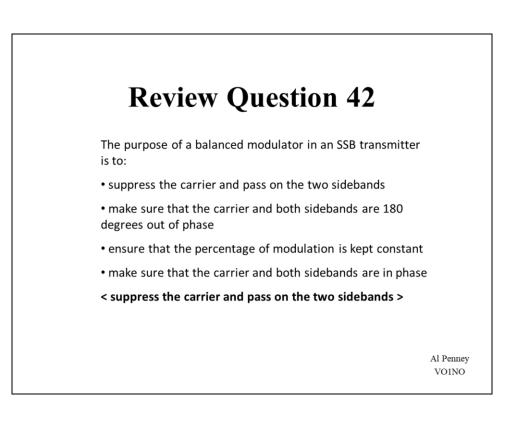
What happens to the signal of an over modulated singlesideband or double-sideband phone transmitter?

- It has higher fidelity and improved signal-to-noise ratio
- It becomes distorted and occupies more bandwidth
- It becomes stronger with no other effects
- It occupies less bandwidth with poor high-frequency response

< It becomes distorted and occupies more bandwidth >

The purpose of a balanced modulator in an SSB transmitter is to:

- suppress the carrier and pass on the two sidebands
- make sure that the carrier and both sidebands are 180 degrees out of phase
- ensure that the percentage of modulation is kept constant
- make sure that the carrier and both sidebands are in phase



In a SSB transmission, the carrier is:

- transmitted with one sideband
- inserted at the transmitter
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What kind of emission would your FM transmitter produce if its microphone failed to work?

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Why is FM voice best for local VHF/UHF radio communications?

- It is more resistant to distortion caused by reflected signals
- Its RF carrier stays on frequency better than the AM modes
- It provides good signal plus noise to noise ratio at low RF signal levels
- The carrier is not detectable

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All amateur stations, regardless of the mode of transmission used, must be equipped with:

- a DC power meter
- an overmodulation indicating device
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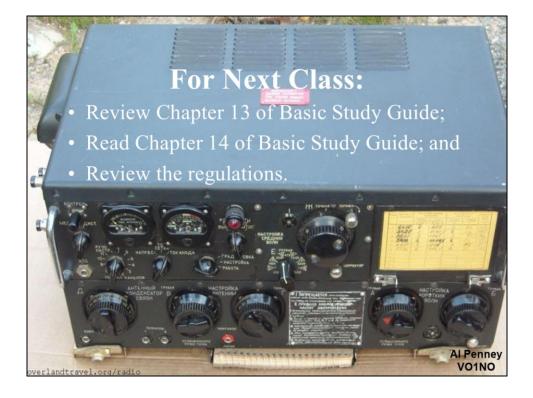
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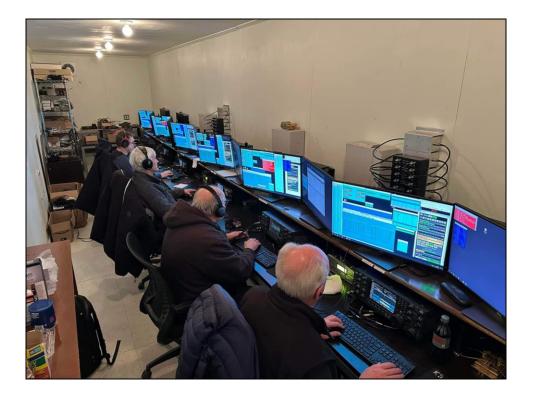




Superstation K1LZ

Left monitor: DXLog and antenna switching. Right Monitor: RFKit RF2KS amp control

Gerry Hull, W1VE





160/80 run on left, 15/10 inband on right

Summary - K1LZ						
BAND	QSO	DXCC	ZONE	DUPE	POINTS	AVG
160	515	77	21	6	1331	2.6
80	1287	109	29	67	3566	2.8
40	2319	138	38	109	6683	2.9
20	2528	144	39	118	7105	2.8
15	2306	144	39	89	6660	2.9
10	2171	140	37	84	6164	2.8
TOTAL	11126	752	203	473	31509	2.8
FINAL SCORE:30 091 095						



