Without getting too complicated, we explain . . .

HOW an SSB system works

Most enthusiasts seem to understand that an SSB transmission comprises one sideband only, and no carrier; that to resolve it, the receiver needs to replace the carrier with a locally generated signal. The real gap, which this article seeks to fill, is an understanding of how an SSB signal is generated in the first place.

by W. EDMUND HOOD W2FEZ*

Suppressed carrier, single sideband transmission has so completely overtaken amateur radiotelephony on the HF (high frequency) bands that conventional AM signals are a rarity. In fact, many stations aren't even equipped to tune in an AM signal.

The advantages of single sideband are well worth the trouble of acquiring the more precise and complex equipment needed to produce and properly receive these signals. It does, however, burden the novice amateur with having to learn much more today than he might have needed 20 years ago, in order to reach an average level of understanding in the communications art.

When voice modulation of a radio signal was first accomplished, it was accepted simply that the level of the signal had been modulated, or caused to vary in a manner corresponding to the frequency and amplitude of the voice. Hence the term "amplitude modulation". However, it soon emerged that the concept, while not necessarily wrong, was certainly simplistic and incomplete.

In fact, the process of amplitude modulating a signal actually produces a pattern of frequencies comprising the original carrier plus adjacent signals, referred to as "sidebands", having a frequency and amplitude dependant on the superimposed audio. The modulated envelope which may be seen on the screen of a cathode ray oscilloscope is virtually a graphical summation of the carrier and sidebands. (See Fig. 1).
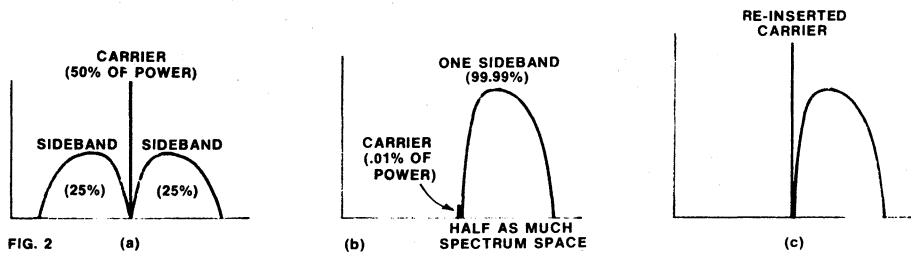
Thus, when we dissect an AM signal mathematically and examine it component by component, we discover that the so-called "carrier" remains constant in its level. We also find that a narrow band of spectrum above and below the carrier frequency is occupied by a symmetrical complex of signals resulting from the combination of the carrier with the voice — the sidebands.

This is consistent with the familiar theory that, when any two signals are combined in a nonlinear device, the output of the device usually contains the original two signals and two new signals, whose frequencies are the sum and the difference of the frequencies

A conventional AM transmitter output stage (a) modulated by an audio signal (b) produces a modulated waveform (c) as it appears on a CRO screen. However, upon analysis, (c) turns out to be equivalent to (d). The combination of the original carrier, with level unchanged, and two groups of sidebands, whose energy and distribution varies instantaneously with the audio signal.
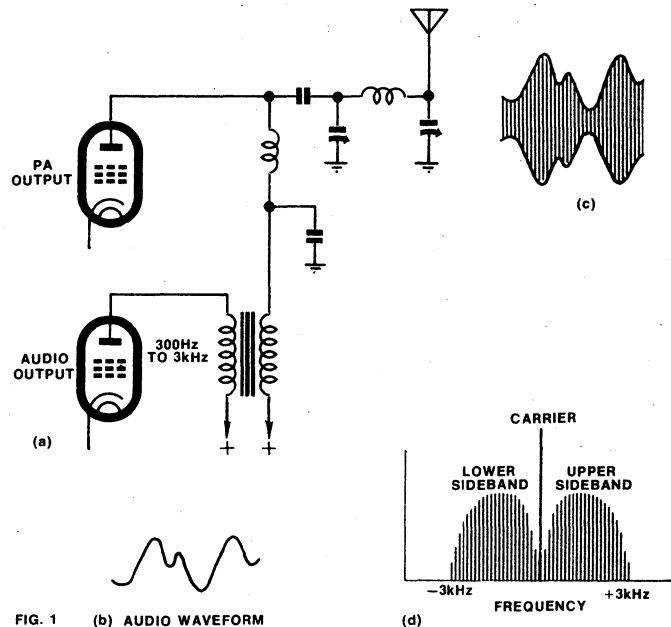
FIG. 1   (b) AUDIO WAVEFORM   (d)

of the original signals.

For example, if we take a signal at 3.9MHz and combine it with a 1kHz signal, we obtain the original two signals plus a signal at 3.901MHz, resulting from the addition of 3.9MHz and 1kHz; we also find a signal at 3.899MHz, resulting from the subtraction of the 3.9MHz and 1kHz signals. If we were to attempt to radiate these signals, the 1kHz would not, of course, radiate, but the other three would. We would have a 3.9MHz carrier and a sideband at 1kHz above and below it.

Like 1d, 2a above depicts a conventional AM signal. In an SSB system, on sideband and the carrier are eliminated and the power capabilities of the output stage concentrated on the remaining sideband (b). A signal equivalent to the missing carrier can be reinserted at the receiver (c), allowing the original audio to be recovered.

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This is an AM signal of a 1kHz tone.

Now, suppose the carrier were eliminated and just the two sidebands transmitted. If we were to insert a corresponding signal or "carrier" at the receiver, it would combine with the 3.899MHz and 3.901MHz signals to reproduce the 1kHz tone. The original carrier, then, is therefore not really essential to recover the 1kHz tone.

To go one step further, we can also eliminate either one of the sidebands and still recover the 1kHz tone. Whether we eliminate the upper or the lower is simply a matter of preference.

If, instead of the 1kHz tone, we were using sidebands representing the complex waveform of the human voice, we can recover the voice in the same way, just as long as we insert a carrier at the correct frequency in the receiver.

The major disadvantage of the system is that the receiver must be very stable (i.e. free from tuning drift) and must include a stable oscillator to substitute for the missing carrier — usually in the IF system just ahead of the detector.

Any drift in receiver tuning tends to alter the pitch of the voice, as received, and may even render it unintelligible. While a certain amount of slow drift can be corrected by careful and frequent re-tuning, this is not practical where the drift is fast, or erratic, or where the tuning mechanism is "jumpy".

Fortunately, with advancing technology, frequency stability is much less of a problem than it once was. Which is just as well, because SSB offers certain very definite advantages.

When an AM signal is transmitted, the overall signal — assuming voice input — would typically cover a portion of the spectrum 6kHz or more wide. The transmitter power would be divided, half of it producing the carrier and half of it producing the two sidebands (one quarter of the power to each sideband).

Now the carrier doesn't do anything, except help the receiver demodulate the signal. If we get rid of it (Fig. 2) we have twice as much power available to transmit the sidebands, which contain all the information in the signal. But, if we get rid of one of the sidebands, there is that much more power available for the other one. (Because of the complex waveform of the sidebands, we only have the effect of two to three times the effective power in the signal, rather than four, as we might have expected.)

Then there's the narrower bandwidth. A single sideband signal occupies less than half the spectrum space required by an equivalent AM signal. This makes room for more stations and reduces the amount of atmospheric noise picked up by a receiver with appropriately narrow bandwidth.
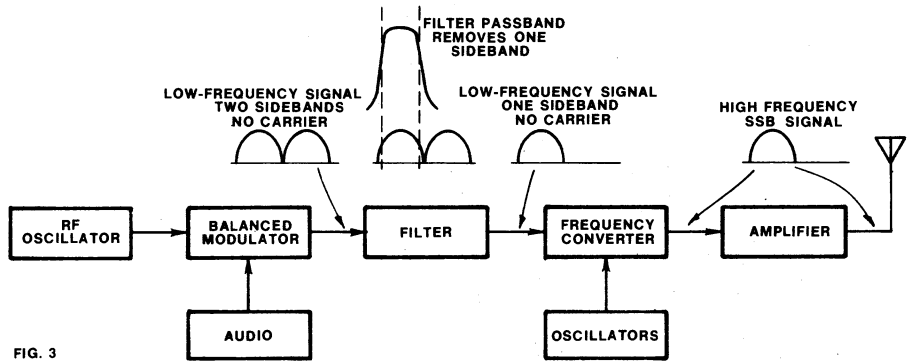
FIG. 3

In a typical SSB transmitter, the carrier is eliminated by the balanced modulator, leaving only the two sidebands. One of these is eliminated by a filter. The remaining sideband is heterodyned to the desired frequency, amplified and radiated.

A typical balanced modulator circuit. Being fed to the common emitter circuit, the carrier cancels in the push-pull output system.
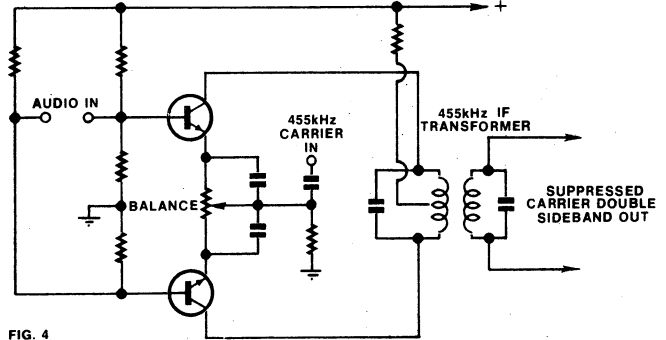
FIG. 4

A suppressed carrier, single sideband signal can be generated by several methods, and most of them produce an equally good end product. It is there, however, that the comparison ends. The phasing method of generating a single sideband signal is so complex that it is seldom used. In fact, I've only seen one amateur rig using this method, and that was produced 10 or 15 years ago by Heathkit.

The most popular method of single sideband generation is called the filter method, and it is this method that I will cover in detail. (See Fig. 3.)

The heart of any single sideband transmitter is a device called a balanced modulator. There is a wide variety of balanced modulator circuits available, some more complex than others, and each one is somebody's favourite. While I don't wish to push one type over the others, I will only cover a sampling here. A balanced modulator, incidentally, is also often used in a receiver to mix the incoming signal with that of the local oscillator to produce the IF signal.

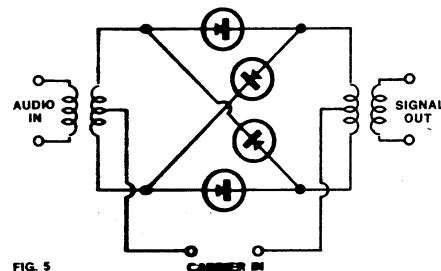Here is one of the easier to understand balanced modulator circuits. (See Fig. 4.) Note that the audio is fed to the two transistors in a push-pull arrangement. The carrier, however, is fed to the transistors in parallel. At the output, the carrier is of the same amplitude and polarity at opposite ends of the winding. Thus it cancels itself out, leaving only the two sidebands. Any small amount of carrier that gets by, due to unequal characteristics of the transistors, is eliminated by adjusting the balancing potentiometer.

The four-diode ring modulator circuit shown in Fig. 5 needs no balancing potentiometer, so long as the diodes are well matched in their characteristics. Its operation is a bit more difficult to explain, but it operates by being thrown in and out of balance by the audio signal. It enjoys its greatest popularity as a radio frequency mixer circuit. The RF transformers are broadband toroids, and, in receivers employing this circuit, they are small enough to fit on your fingernail.

The two-diode balanced modulator of Fig. 6 is an easy one to make and often represents the best compromise in simplicity and efficiency. Any of the above circuits have carrier suppression of 50dB or better, when properly built and balanced. They are not, by any means, the only balanced modulator circuits in use, but are representative.

A balanced modulator, whichever circuit is used, produces the two sidebands characteristic of an AM signal, but no carrier. This brings us halfway to our goal. Now we must eliminate one of the sidebands. This is done with a very selective filter, hence the name "filter method".

FIG. 5

A four diode ring modulator, relying for its efficiency on balanced components.

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Two kinds of filters enjoy considerable popularity in this application. One is made using quartz crystals of the proper frequencies, connected in a lattice or similar arrangement. (See Fig. 7.) The other consists of two magnetic transducers, and a number of mechanically resonant discs. Mechanical filters, generally made by Collins, are more popular in commercial systems and in the more expensive amateur transmitters.

Whichever type of filter is used, it must pass a band of frequencies about 3kHz wide for normal speech transmissions and must reject all frequencies outside that band, with a very high amount of attenuation. (See Fig. 8.)

The nominal frequency of the RF signal fed into the balanced modulator is selected to be just outside the passband of the filter. The filter passes the desired sideband almost without loss and eliminates the other sideband and any residual carrier. The selection of either the upper or the lower sideband is accomplished by changing the (residual) carrier frequency from one side of the filter passband to the other.

FREQUENCY CHANGE

Single sideband signals are usually generated at a fairly low frequency and converted later to the desired transmission frequency. This is accomplished by mixing the low frequency signals with one of a higher frequency and then tuning out the unwanted products.

In one brand of commercial unit, for example, the signal is originally generated in the neighbourhood of 455kHz and then mixed with a signal close to 1955kHz. This results in a signal at 1500kHz and another at 2410kHz. The 1500kHz signal is selected and the other rejected by the tuned circuits. This signal is amplified and then mixed with a still higher frequency.

If, for example, we wanted the final product at 3.9MHz, we could obtain it by combining the 1.5MHz signal with a 5.4MHz signal.

We should note here that, each time we mix and use the difference frequency, rather than the sum, the sidebands invert. That is, the upper sideband would become a lower one and vice versa. It's okay to do this this way, as long as we are aware of this phenomenon.

Once we have the signal converted to the proper frequency, all we have to do is to amplify it to the desired power level. Here again, we have to be careful what kind of amplifier we use. Amplifiers are classed according to the way the grid of the valve or the base of the transistor is biased.

The most efficient of the radio frequency power amplifiers, in terms of

At right: A two-diode balanced modulator, suggested as good compromise between simplicity and efficiency.
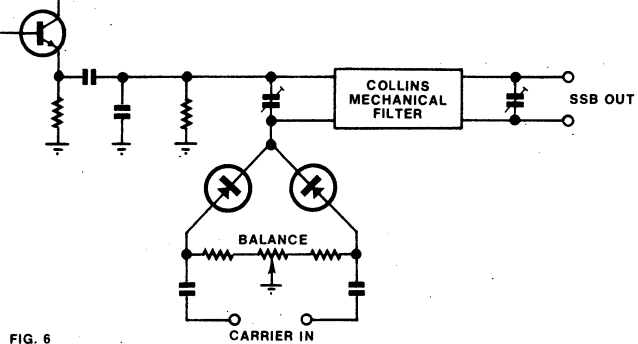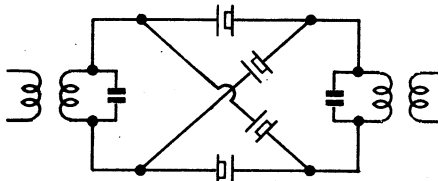
FIG. 6

FIG. 7

Mechanical filters (Fig. 6) are efficient but also expensive and, as an alternative, crystal lattice filters, as above, are often preferred for amateur equipment.

DC input power, is biased as class C. This means that the voltage on the grid or the base is set so that current flows only during the peaks of input voltage. This is all right for CW operation, but wouldn't work very well with a single sideband signal due to the distortion that would occur.

A class B amplifier is biased so that current flows only during the positive half of the input cycle.

A class A amplifier is biased so that current flows at all times. It is the least efficient, but produces the least amount of harmonic distortion. Amplifiers biased at class A, class B, or in between are known as linear amplifiers, and this is the only kind that can be used when the input signal is already modulated.
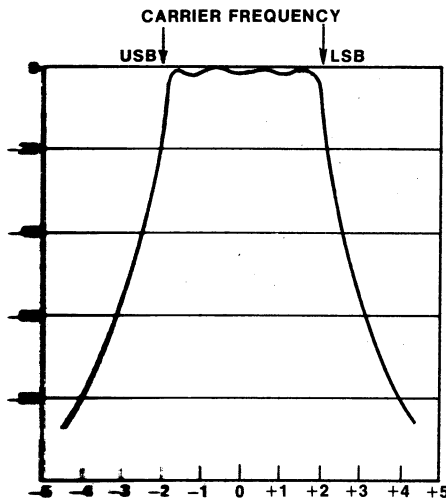
The features just described represent

CARRIER FREQUENCY

FIG. 8

The whole performance of an SSB transmitter depends on the filter characteristic shown here.

the characteristics common to all single sideband transmitters. If you can grasp them, you are more than halfway to a general understanding of the mode.

Receivers designed for signals of this type differ from conventional receivers only in that they are more selective than their ancestors, and the detector circuit, instead of being a simple diode, is a balanced mixer. This enables the most efficient demodulation of the signal. Older receivers can receive single sideband signals as long as they have a CW oscillator. However, it does require careful tuning and adjustment of the pitch control or the CW oscillator, and, in the ultimate, success or otherwise depends on both the mechanical and electrical design.

The frequency conversion provisions common to SSB receivers and transmitters make transceiver design the most economical way to go, though not necessarily the most versatile. The carrier oscillator continues to operate in the receive mode, thereby enabling demodulation of the signal, and, since the same conversion oscillators are used in both modes, transmit and receive frequencies are automatically locked together.

When operating single sideband with separate transmitter and receiver, you should remember that very careful tuning is essential to get your transmitter and receiver on precisely the same frequency.

Some transmitters include a feature that produces just enough carrier to enable spotting the frequency on the receiver. With others, you may have to speak into the microphone while fine-tuning the vfo (in the spot mode, not in transmit, please) until the voice sounds natural in the receiver. Do this after having first tuned the receiver to the desired signal.

Upper sideband is generally preferred on the 14MHz amateur band and above, while lower sideband is commonly used for 7MHz and below. Other than custom, however, there is no rule dictating which sideband to use. A little practice, and the novice should very quickly be joining us after he gets the appropriate ticket. This article hasn't covered the entire extent of single sideband operation, but I hope that it has, at least, helped to get the beginner over the hump.