Chapter 6: Single Sideband Systems

Chapter 6 Objectives

At the conclusion of this chapter, the reader will be able to:

- Explain the advantages of SSB over conventional AM transmission.
- Describe the various types of SSB transmission.
- Explain the operation of circuits used to generate SSB signals.
- Given the block diagram of a SSB transmitter, describe the nature of the signal at each point in the circuit.
- Draw a block diagram of a SSB receiver.
- Explain how SSB signals are demodulated.
- Given the block diagram of a SSB transceiver, follow the signal flow for receive and transmit conditions.
- Describe typical troubleshooting procedures for SSB transceivers.

AM is a widely used method of communication. AM transmitters are relatively simple, and AM receivers can be built very inexpensively with just a handful of components. Despite this, AM suffers from very poor efficiency. As was demonstrated in chapter 3, at best only 33% of the transmitted energy in an AM signal actually carries information (the sidebands). The remainder (about 67%) of the energy is used to transmit the *carrier frequency component*, which itself carries no information.

Furthermore, an AM signal contains two sidebands, each of which carry the same information. Having two sidebands causes the required transmitter bandwidth to be twice as wide as absolutely necessary.

By modifying the AM signal in various ways, we can get various types of *sideband* transmissions. Transmitters and receivers for sideband are more complex than their AM equivalents. However, there are situations where the added complexity is justified. Where many stations must share a limited range of frequency space, sideband techniques are useful since they can reduce bandwidth. Where voice communications must be carried out over a noisy path, sideband methods prove superior to AM and FM.

6-1 SSB Versus AM: Types of Sideband Signals

Figure 6-1 shows a typical AM signal, which can be thought of as a DSB-FC (double-sideband, full-carrier) signal.

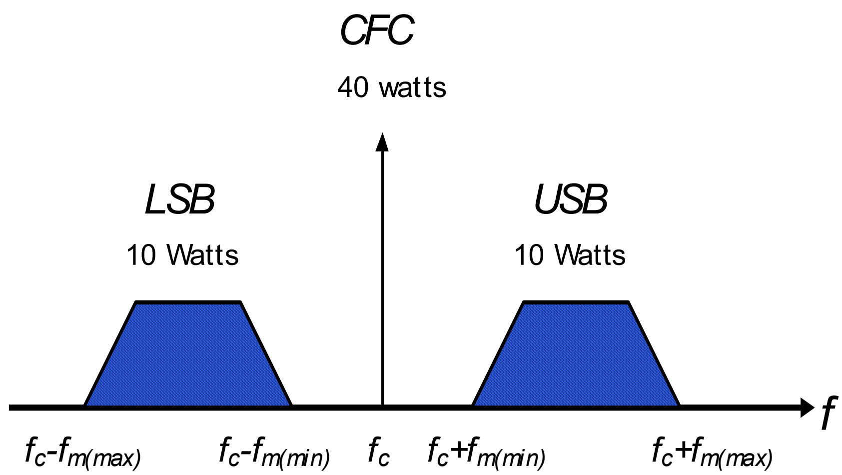

Figure 6-1: A Conventional AM (DSB-FC) Signal

A conventional AM signal consists of a *carrier frequency component* or *CFC*, a *lower sideband*, and an *upper sideband*. Notice how we have shown the sidebands as *ranges* of frequency, rather than individual frequencies. This is because a real information signal has a *range* of frequencies rather than just a single frequency.

The minimum and maximum information frequencies are marked as $f_{m(min)}$ and $f_{m(max)}$. Therefore, the bandwidth of a conventional AM signal can be expressed as:

(6-1) $$BW_{AM} = 2f_{m(\max)}$$

The Minimum Bandwidth for Speech

In the 1960s, Bell Laboratories conducted research to determine the minimum range of frequencies needed for understandable voice communications. The research was important because the telephone companies were sending many different conversations at the same time through a single copper telephone wire. The technique of sending multiple pieces of information at the same time is called *multiplexing*. Each conversation would be assigned a different range of frequencies. If a certain amount of bandwidth were available on the wire, how many conversations could be sent at once? That would depend, of course, on the bandwidth required for each person's voice. Bell's research efforts demonstrated that most of the information power in human speech is contained in the frequency range 300 Hz to 3000 Hz. Frequencies below 300 Hz add bass "presence" to voice, but little intelligibility. Most of the energy above 3000 Hz is from the unvoiced speech sounds, such as *s*, *f*, and so on. *Therefore, most systems that are intended to send only human voice are designed to reproduce the frequency range 300 Hz ($f_{m(min)}$) to 3000 Hz ($f_{m(max)}$).*

Example 6-1

Speech is to be sent using an AM transmitter. The available carrier power is 40 watts, as in Figure 6-1. The transmitter is operating at 100% modulation.

a) What is the *total power?*
b) What is the power of the information?
c) What bandwidth will be needed?

Solution
The total power can be calculated as:

$$P_t = P_c(1+\frac{m^2}{2}) = 40W(1+\frac{1^2}{2}) = \underline{\underline{60\ watts}}$$

The 60 watts is the total available power from the transmitter.

The sideband power is the same thing as the information power. Since the carrier power is 40 W, the *sideband* power will be:

$$P_{\inf o} = P_{side} = Pc\left(\frac{m^2}{2}\right) = 40W\left(\frac{1^2}{2}\right) = \underline{\underline{20\ watts}}$$

Notice how the information power is only 33% of the total power being transmitted! Sending speech requires a bandwidth from 300 Hz to 3000 Hz. The highest information frequency $f_{m(max)}$ is 3000 Hz. Therefore,

$$BW = 2f_{m(\max)} = 2(3000Hz) = \underline{\underline{6kHz}}$$

Note that the minimum information doesn't affect bandwidth at all in a conventional AM transmission.

DSB-SC Operation

The carrier frequency component (CFC) uses up most of the available transmitter power in a conventional AM transmission. By using a special circuit called a *balanced modulator*, we can produce an AM signal with sidebands but no CFC. Such a signal is called a *double-sideband suppressed-carrier* emission, and is pictured in Figure 6-2.

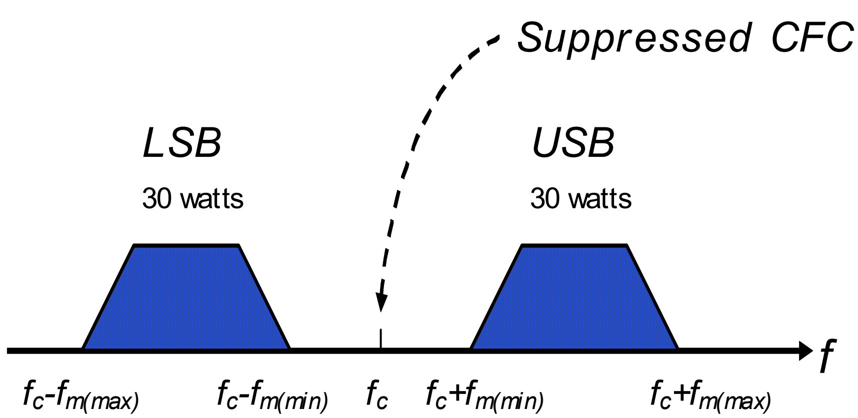

Figure 6-2: A DSB-SC Signal

The carrier frequency component is *gone* from the signal above, but something else has happened. Closely compare Figures 6-1 and 6-2. The *sidebands* have suddenly grown in power! Why is this possible?

The total available power was 60 watts for the original AM signal of Figure 6-2. When we removed the 40 watt carrier, those 40 watts of power became available for transmitting information. Therefore, *all* of the transmitted power is information power -- a great improvement over conventional AM.

However, the resulting DSB-SC signal can no longer be properly demodulated by a diode AM detector. Without a carrier frequency component to act as a "reference" signal, a diode detector produces garbled information. Therefore, an oscillator circuit called a *beat frequency oscillator*, or *BFO*, must be added to the detector circuit to reinsert the missing carrier signal.

Since we are now transmitting additional information power, we say that we have gained a *decibel power advantage* over a conventional AM transmitter. The decibel power advantage can be computed as follows:

(6-2) **Decibel power advantage,** $dB = 10\log\left(\left\langle\frac{P_2}{P_1}\right\rangle\left\langle\frac{BW_1}{BW_2}\right\rangle^2\right)$

Where P_2 is the new information power, P_1 is the original information power, BW_2 is the new bandwidth, and BW_1 is the original bandwidth. Note that the squared bandwidth ratio takes into account the *combined* advantage for both the transmitter and receiver. The ratio of powers is *not* squared!

Example 6-2

If the total power of the transmission of Figure 6-2 is 60 watts (same as Figure 6-1), and again human speech is to be transmitted, calculate:
a) The bandwidth of the DSB-SC signal.
b) The decibel power advantage of the DSB-SC signal over the AM-FC signal.

Solution
As you can see from Figure 6-2, the bandwidth of the signal will remain unchanged, since we're still sending two sidebands. The bandwidth will therefore remain at (2)(3000 Hz) or 6 kHz.

Equation 6-2 can be used to calculate the decibel advantage. The information power in the AM signal is 20 watts, and the information power in the DSB signal is 60 watts. There's no change in bandwidth between the two modes:

$$dB = 10\log\left(\left\langle\frac{P_2}{P_1}\right\rangle\left\langle\frac{BW_1}{BW_2}\right\rangle^2\right) = 10\log\left(\left\langle\frac{60W}{20W}\right\rangle\left\langle\frac{6KHz}{6KHz}\right\rangle^2\right) = \underline{\underline{+4.77dB}}$$

In other words, for the conventional AM transmitter to be as effective as the DSB-SC transmitter, the AM transmitter would have to be operating at a power level that is +4.77 dB (three times) stronger than the DSB unit. In other words, the *total* power level (Pt) of the AM transmitter would have to be *180 watts* to equal the information carrying capacity of the DSB-SC unit. This is a great power savings!

SSB-SC Operation

We can improve the efficiency of transmission even further by eliminating one of the redundant sidebands. It does not matter which one is removed; when the upper sideband is kept, we say that we're operating in *USB* mode; when the lower sideband is kept, we're in *LSB* mode. Most people simply refer to this mode as *SSB*; they assume that the carrier is suppressed.

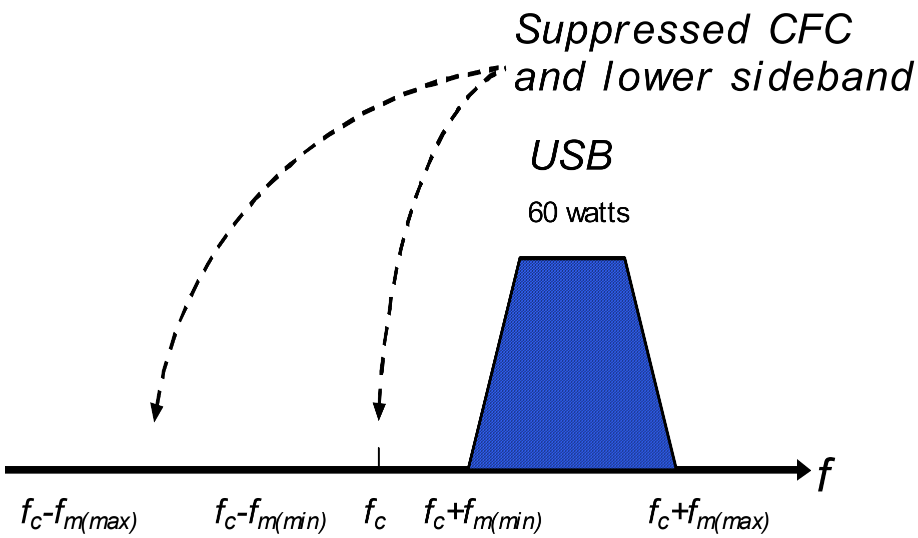

Figure 6-3: A SSB-SC Signal in USB Mode

A SSB signal has a bandwidth that is slightly less than *half* of a corresponding AM or DSB-SC signal. Look how the entire 60 watts is now concentrated in a much more narrow "slice" of spectrum. This characteristic gives a SSB transmitter much more "talk power" than an AM transmitter of the same power level!

Example 6-3

If the SSB signal of Figure 6-3 is sending a voice signal (300 Hz - 3 kHz), calculate:
a) The bandwidth
b) The decibel power advantage of the SSB signal over the AM signal of Figure 6-1.
c) The power level an AM transmitter would need in order to have equivalent performance.

Solution
a) Since we're only sending one sideband, equation 6-1 doesn't apply. Instead, we can fall back on the basic definition of bandwidth:

$$BW = f_{\max} - f_{\min}$$

b) In Figure 6-3, f_{min} is equal to $f_c + f_{m(min)}$, and f_{max} is equal to $f_c + f_{m(max)}$, so we can state:

$$BW = f_{\max} - f_{\min} = (f_c + f_{m(\max)}) - (f_c + f_{m(\min)}) = f_{m(\max)} - f_{m(\min)}$$
$$BW = f_{m(\max)} - f_{m(\min)} = 3000Hz - 300Hz = \underline{\underline{2.7\ kHz}}$$

Notice that the bandwidth is just slightly less than one half of the bandwidth required for an AM-FC transmission.

Equation 6-2 calculates decibel power advantage:

$$dB = 10\log\left(\left\langle\frac{P_2}{P_1}\right\rangle\left\langle\frac{BW_1}{BW_2}\right\rangle^2\right) = 10\log\left(\left\langle\frac{60W}{20W}\right\rangle\left\langle\frac{6KHz}{2.7KHz}\right\rangle^2\right) = \underline{\underline{+11.7\ dB}}$$

Note that we're not transmitting any additional information power; we're just packing the information into a smaller bandwidth "space."

c) The power ratio expressed by an 11.7 dB advantage can be found by simply expressing the terms inside the dB formula for power advantage:

$$G_p = \left\langle\frac{P_2}{P_1}\right\rangle\left\langle\frac{BW_1}{BW_2}\right\rangle^2 = \left\langle\frac{60W}{20W}\right\rangle\left\langle\frac{6KHz}{2.7KHz}\right\rangle^2 = \underline{\underline{14.8:1}}$$

The AM transmitter would need about 14.8 times the power of the SSB transmitter in order to have equivalent performance. That is a total power of (14.8)(60W) or 888 watts.

SSB is much more efficient than AM. In general, most people agree that there is better than a 10 dB (10:1) power advantage for SSB transmission over AM.

Vestigial Sideband (VSB) Mode

One of the disadvantages of DSB-SC and SSB transmission is that neither of these signals can be demodulated with a diode detector. A mode that combines the advantage of conventional AM with the reduced bandwidth requirements of SSB is called *vestigial sideband*, or *VSB*. A VSB signal is shown below in Figure 6-4:

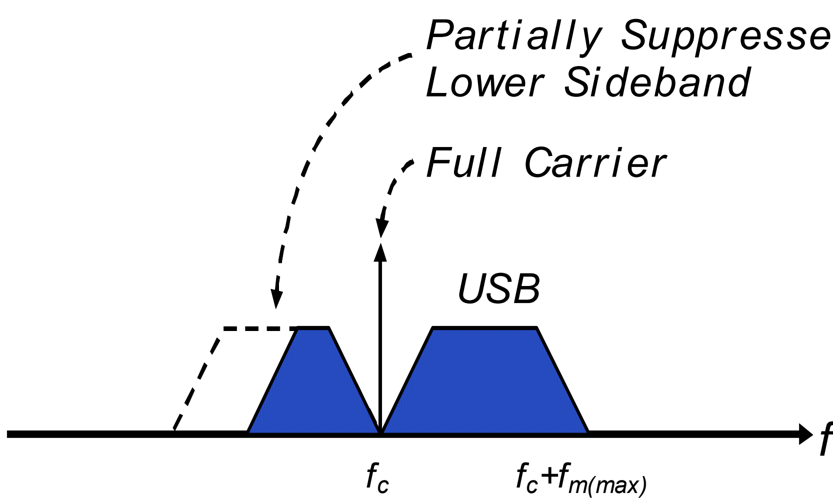

Figure 6-4: VSB Operation (USB Mode)

In VSB, the carrier is left intact (thus allowing detection with a diode detector), and most of the lower sideband is filtered out. The resulting signal has a reduced bandwidth when compared to a conventional AM signal. Analog television uses VSB to transmit the picture because the information frequencies contained within the video signal range from near DC (30 Hz) to approximately 4 MHz. Transmitting a video signal by conventional AM would therefore require an 8 MHz bandwidth just for the picture! By using VSB, the bandwidth is reduced to slightly less than 5 MHz -- which leaves room to squeeze in an FM sound carrier. The details of television transmission will be covered in a later chapter.

Percentage of Modulation and Other Myths

You might recall that for an AM signal, the modulation index (and hence percentage of modulation) could be calculated as follows:

$$m = \frac{Vm}{Vc}$$

In a DSB-SC or SSB-SC signal, there is no carrier transmitted; only sidebands are sent. Using the formula above for SSB or DSB signals causes a problem, for we can't divide by a *zero Vc* to get an answer. Percentage of modulation is meaningless!

SSB and DSB transmitter outputs are measured according to their *peak envelope power*, or *PEP*. PEP is calculated by finding the average (RMS) power delivered during one RF cycle at the *peak* or *maximum* of the modulation envelope. Sideband transmitters are always rated in this manner; as long as the operator of the transmitter does not exceed the rated PEP level of the unit, no "overmodulation" occurs and a clean modulated signal will be delivered to the antenna.

Example 6-4

What is the PEP of the AM and DSB-SC signals in Figure 6-5? Each is being measured across a 50 ohm dummy load.

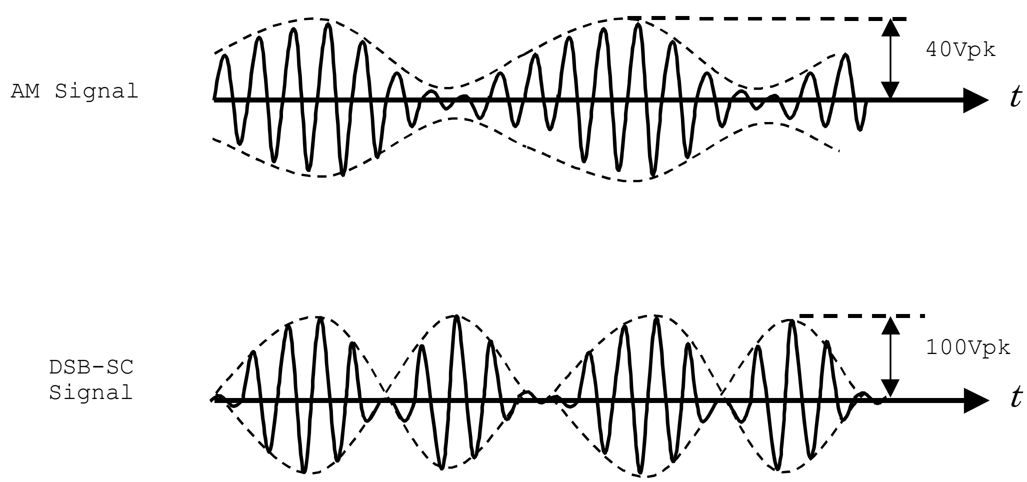

Figure 6-5: AM and DSB-SC signals for PEP measurement

Solution

To calculate PEP, find the RMS power during one RF cycle at the crest of the modulation envelope. For the AM signal, the crest voltage is 40 Vpk, so we can find the RMS voltage:

$$V_{rms} = \frac{V_{pk}}{\sqrt{2}} = \frac{40Vpk}{\sqrt{2}} = \underline{28.28\,V}$$

Now Ohm's law is applied to find the PEP power:

$$PEP = \frac{V_{rms}^2}{R_L} = \frac{28.28V^2}{50\Omega} = \underline{\underline{16\ watts}}$$

For the DSB transmitter, the exact same procedure is used, and we get:

$$V_{rms} = \frac{V_{pk}}{\sqrt{2}} = \frac{100Vpk}{\sqrt{2}} = \underline{70.7V}$$

$$PEP = \frac{V_{rms}^2}{R_L} = \frac{70.7V^2}{50\Omega} = \underline{\underline{100\ watts}}$$

For SSB mobile and fixed-station transmitters, 100 watts PEP is a fairly standard value for power output.

Section Checkpoint

6-1 What frequency component uses up most of the power in a conventional AM signal?
6-2 What determines the bandwidth of an AM signal?
6-3 What is the minimum frequency range for reproduction of speech?
6-4 How is a DSB-SC signal different from an AM-FC signal?
6-5 What signal must be reinserted at the detector circuit in a receiver designed to receive DSB and SSB signals?
6-6 What is meant by the term *decibel power advantage*?
6-7 What type of emission results when the CFC and one sideband are removed?
6-8 How is the bandwidth of an SSB signal computed?
6-9 What is the advantage of VSB over SSB and DSB-SC?
6-10 Why is percentage of modulation meaningless for SSB?
6-11 How is PEP calculated?

6-2 SSB Signal Generation: Filter Method

SSB signals can be generated using several methods. These methods are known as the *filter method*, *phasing method*, and a relatively new approach, the *DSP (digital signal processor) method*. Most modern transceivers use the filter method; as the power of DSP chips rises (and their price tumbles), it is expected that the DSP method will continue to grow in popularity. Figure 6-6 illustrates the filter method.

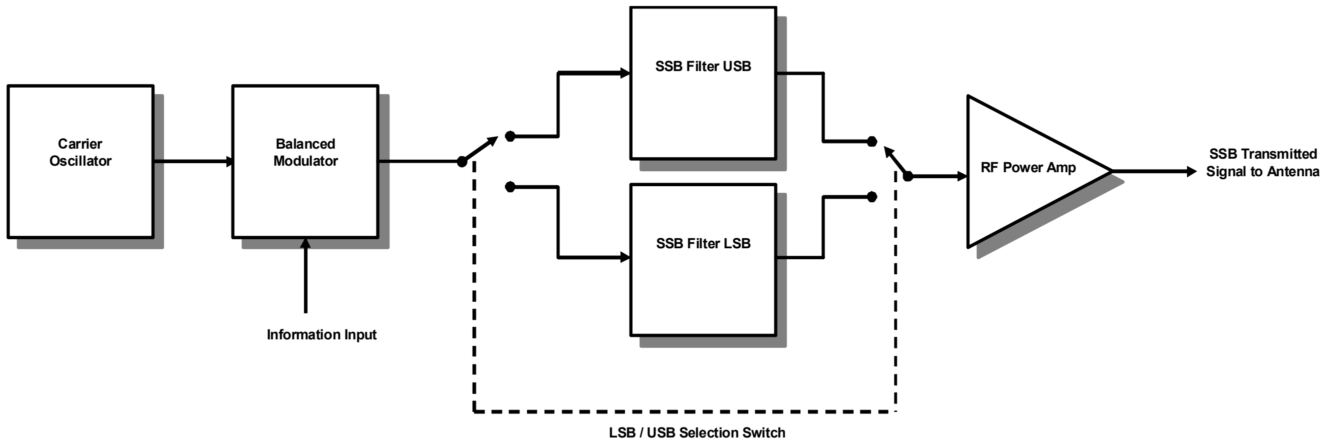

Figure 6-6: Filter Method of SSB Generation

In the filter method, two steps are taken to generate a SSB signal. First, a DSB-SC signal is generated in a *balanced modulator*. Second, the undesired sideband is removed by filtration. An actual SSB transceiver isn't likely to use two filters; it's simply easier to picture the process this way. We'll see soon how it can be done with a single filter.

The resulting SSB signal must be amplified before being sent to the antenna. A *linear* RF power amplifier is required. All SSB transmitters can therefore be considered to be *low-level* transmitters.

Balanced Modulator with Two JFETs

Figure 6-7 shows a balanced modulator constructed from a pair of JFETs. There are two inputs to the circuit, the *carrier* input and the *information* input.

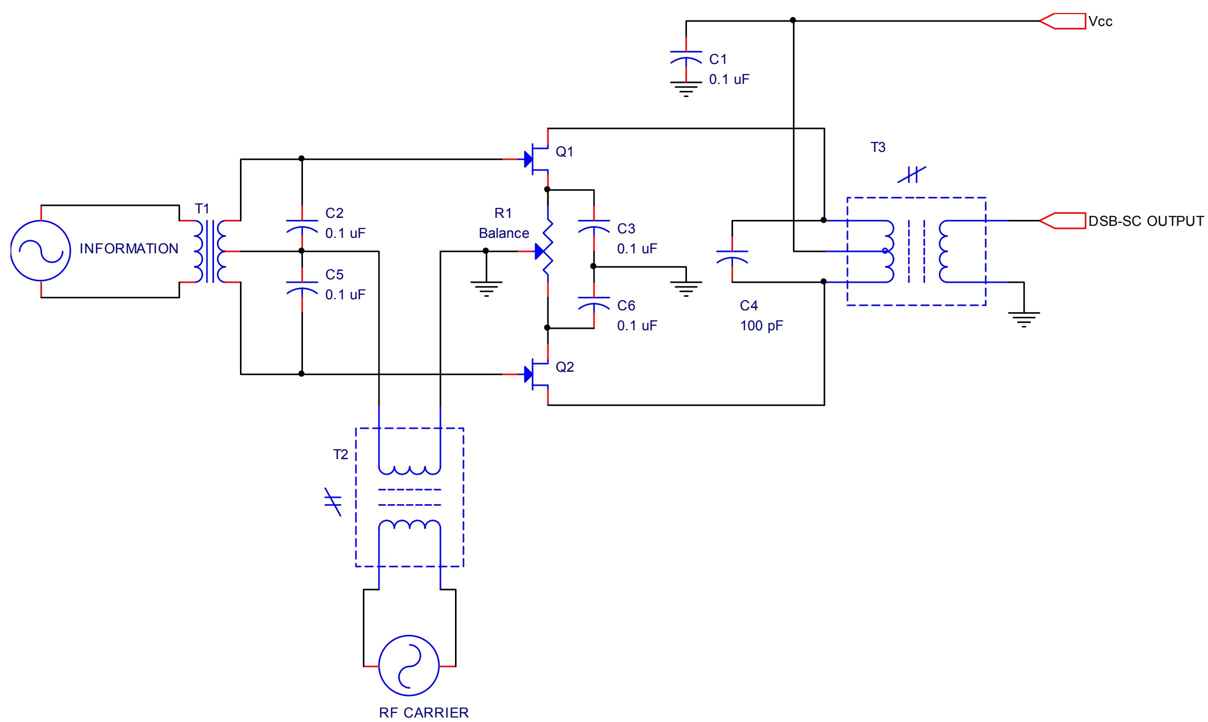

Figure 6-7: A Balanced Modulator Employing Two JFETs

The carrier signal enters the balanced modulator through transformer T_2. The RF signal flows left after leaving the secondary (top coil) of T_2, where it skips the secondary of T_1 (right coil) by passing through RF bypass capacitors C_2 and C_5. This seems self-defeating; why isn't T_1 part of the RF circuit? T_1 is in the circuit for the introduction of an audio signal, as we'll soon see.

The RF carrier signal flows in-phase to the gates of transistors Q_1 and Q_2. This is a little strange; an amplifier built like this *usually* has the input signals fed 180 degrees out of phase to the two transistors. This one has the signals being applied *in phase*. The result is that both Q_1 and Q_2 "push" the signal out their drain pins *in phase* to the ends of transformer T3, the output transformer.

Because of the center tap on T_3, the ends of the center-tapped primary would normally be expected to be 180 degrees out of phase. When Q_2 and Q_3 supply the in-phase RF signal to both halves of T_3, the RF signal *cancels* inside T_3 and *no output is produced!* This is known as "push-push" operation. It's very similar to a teeter-totter. If the same thing happens on both sides of a teeter-totter, it doesn't move. This push-push action *cancels* the RF carrier signal, which is exactly what we want.

That's great, you think; all this amplifier analysis just to get *nothing* out! Let's unbalance the "teeter-totter." Transformer T_1 is an *audio coupling transformer*, and an audio information signal is passed into the left side of T_1. The audio signal is slowly changing with respect to the RF signal. When the audio signal goes positive, the gate of Q_1 becomes more positive, and Q_1's RF gain increases; at the same time, the gate of Q_2 is becoming more *negative*, so Q_2's gain *decreases*. Guess what? Now Q_1 and Q_2 aren't "pushing" exactly alike anymore; the audio signal from T_1 has *unbalanced* them. Since Q_1 and Q_2 are now producing unequal signals, they no longer cancel within T_3, and an output results. The exact opposite happens on the negative half-cycle.

Variable resistor R_1 is included as a *balance* control. Because of all the component tolerances, it's likely that some minor adjustment will be necessary to get Q_1 and Q_2 to have the same gain. R_1 is also known as a *carrier null* control.

To properly align a balanced modulator, a technician removes any audio signal, and adjusts the carrier null (balance) control for *minimum* output from the circuit. Some balanced modulators also include a trimmer capacitor; the two controls are adjusted alternately for the deepest possible null in the output.

Figure 6-8 shows typical waveforms for the balanced modulator above. Note how the gain of Q_1 and Q_2 alternate -- when Q_1's gain is being increased by the intelligence, Q_2's gain is being reduced. This is what causes the imbalance in T_3.

The output waveshape of a DSB-SC signal from a balanced modulator circuit is different than the original information. The envelope has a "cat's eye" shape, with a period *one-half* of the information period. (There are two complete cat's-eyes for each cycle of information). The point of zero crossing always has a sharp wedge shape in a DSB-SC waveform. These features are important; they show proper operation of the balanced modulator.

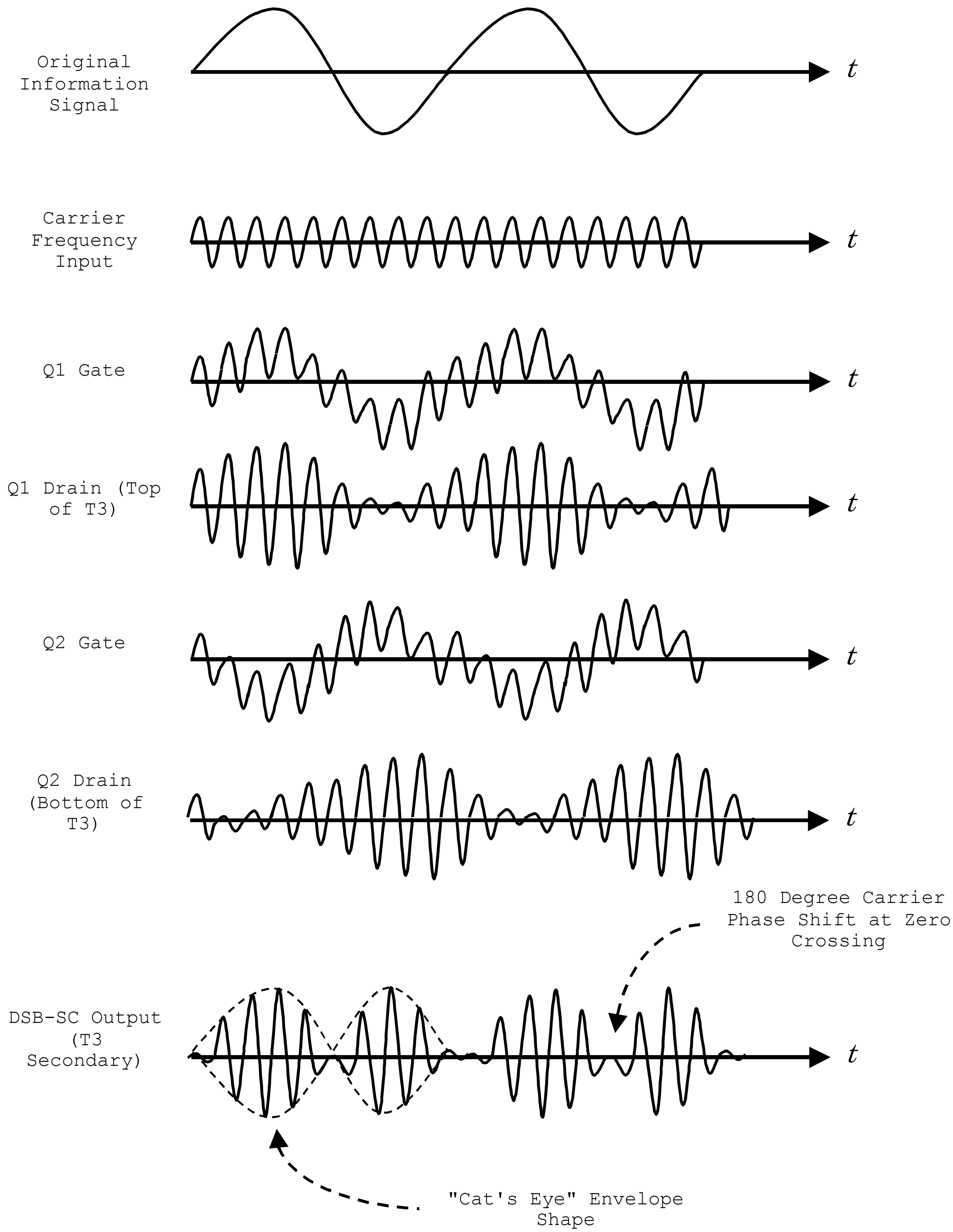

Figure 6-8: Balanced Modulator Waveforms for the Circuit of Figure 6-7

Other Balanced Modulator Circuits

Two other balanced modulator circuits are very popular; they are the *diode modulator* and *integrated circuit modulator*. The diode modulator is probably the most popular of all balanced modulator circuits; one is shown in Figure 6-9:

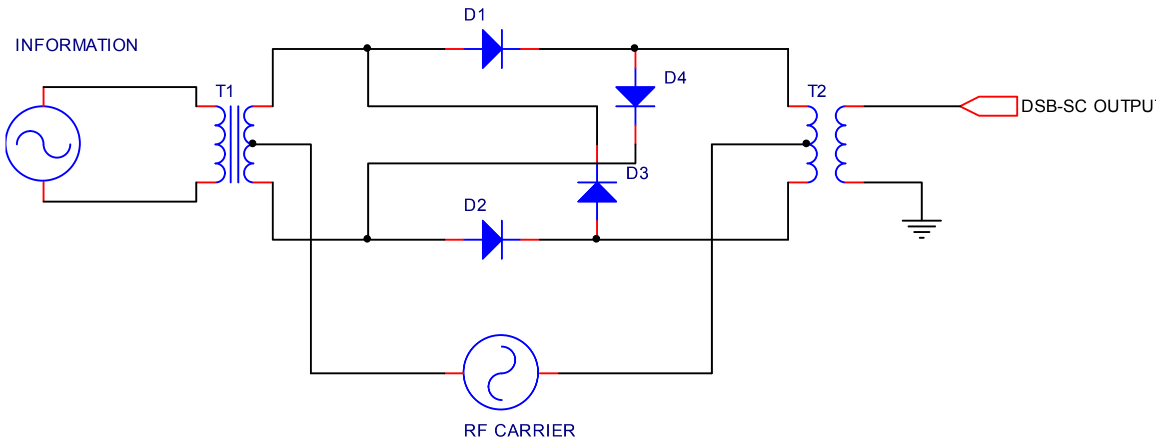

Figure 6-9: A Double-Balanced Diode Modulator

In the diode modulator of Figure 6-9, the RF carrier signal is applied in the middle of the circuit between T_1 and T_2. For a moment, assume that the information voltage is zero. Under this condition, diodes D_1 and D_2 will conduct equally on the positive half cycle of the RF carrier, and equal currents will flow through them to the top and bottom of T_2. You know what that means: Equal signal to the top and bottom halves of a center-tapped transformer means no output!

The same action occurs on the negative half-cycle of the RF carrier; instead, diodes D_3 and D_4 now conduct, again equally, and again, with no resulting output from T_2.

In order to get output, something needs to shove diode pairs D_1-D_2 and D_3-D_4 off balance. If we apply an information voltage into T_1, that will do the trick! On the positive half-cycle of the *information*, diodes D_1 and D_4 are forward biased, while diodes D_2 and D_3 are reverse-biased. You're not misreading this! This imbalances the diodes, and now the RF currents in the upper and lower halves of T_3 will be unequal. The exact opposite happens on the negative half cycle of the information. The result is a double-sideband, suppressed carrier at the output.

The diode modulator is very popular; commercial equipment often uses a hybrid form in a sealed metal can, which provides very high carrier rejection and excellent reliability.

Integrated Circuit Modulators

The Analog Devices AD834, Signetics NE602A, and Motorola MC1496 are examples of integrated circuit balanced mixers that can function either as balanced modulators, balanced mixers, or SSB demodulators. These devices are very common in modern equipment. Figure 6-10 is the data sheet for the MC1496.

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MOTOROLA

MC1496, B

Balanced Modulators/ Demodulators

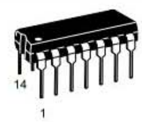
BALANCED MODULATORS/DEMODULATORS

SEMICONDUCTOR TECHNICAL DATA

These devices were designed for use where the output voltage is a product of an input voltage (signal) and a switching function (carrier). Typical applications include suppressed carrier and amplitude modulation, synchronous detection, FM detection, phase detection, and chopper applications. See Motorola Application Note AN531 for additional design information.

- Excellent Carrier Suppression –65 dB typ @ 0.5 MHz
  –50 dB typ @ 10 MHz
- Adjustable Gain and Signal Handling
- Balanced Inputs and Outputs
- High Common Mode Rejection –85 dB typical

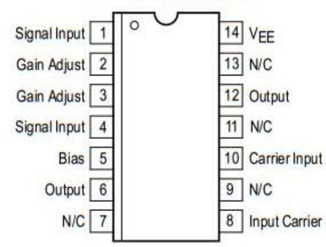
This device contains 8 active transistors.



D SUFFIX
PLASTIC PACKAGE
CASE 751A
(SO–14)

P SUFFIX
PLASTIC PACKAGE
CASE 646

PIN CONNECTIONS



ORDERING INFORMATION

Device	Operating Temperature Range	Package
MC1496D	T_A = 0°C to +70°C	SO–14
MC1496P	T_A = 0°C to +70°C	Plastic DIP
MC1496BP	T_A = –40°C to +125°C	Plastic DIP

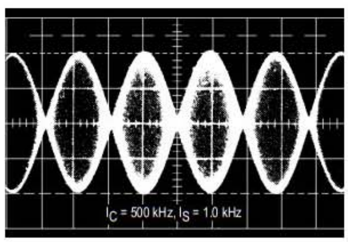


Figure 1. Suppressed Carrier Output Waveform

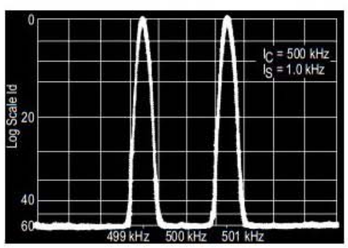


Figure 2. Suppressed Carrier Spectrum

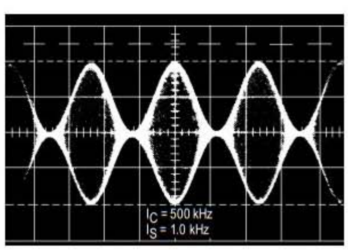
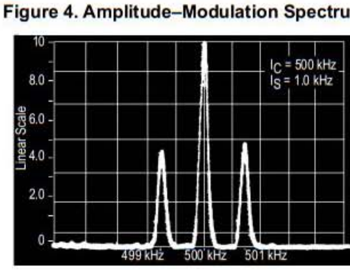


Figure 3. Amplitude Modulation Output Waveform

Figure 4. Amplitude–Modulation Spectrum



 Rev 4

Figure 6-10: Data Sheet for the MC1496 Integrated Circuit Modulator (Copyright of Motorola, used by permission)

MC1496, B

MAXIMUM RATINGS (T_A = 25°C, unless otherwise noted.)

Rating	Symbol	Value	Unit
Applied Voltage (V6 – V8, V10 – V1, V12 – V8, V12 – V10, V8 – V4, V8 – V1, V10 – V4, V6 – V10, V2 – V5, V3 – V5)	ΔV	30	Vdc
Differential Input Signal	V8 – V10 V4 – V1	+5.0 $\pm(5 + I5R_e)$	Vdc
Maximum Bias Current	I_5	10	mA
Thermal Resistance, Junction–to–Air Plastic Dual In–Line Package	$R_{\theta JA}$	100	°C/W
Operating Temperature Range	T_A	0 to +70	°C
Storage Temperature Range	T_{stg}	–65 to +150	°C

NOTE: ESD data available upon request.

ELECTRICAL CHARACTERISTICS (V_{CC} = 12 Vdc, V_{EE} = –8.0 Vdc, I5 = 1.0 mAdc, R_L = 3.9 kΩ, R_e = 1.0 kΩ, T_A = T_{low} to T_{high}, all input and output characteristics are single–ended, unless otherwise noted.)

Characteristic	Fig.	Note	Symbol	Min	Typ	Max	Unit
Carrier Feedthrough V_C = 60 mVrms sine wave and offset adjusted to zero, f_C = 1.0 kHz f_C = 10 MHz V_C = 300 mVpp square wave: offset adjusted to zero, f_C = 1.0 kHz offset not adjusted, f_C = 1.0 kHz	5	1	V_{CFT}	 – – – –	 40 140 0.04 20	 – – 0.4 200	µVrms mVrms
Carrier Suppression f_S = 10 kHz, 300 mVrms f_C = 500 kHz, 60 mVrms sine wave f_C = 10 MHz, 60 mVrms sine wave	5	2	V_{CS}	 40 –	 65 50	 – –	dB k
Transadmittance Bandwidth (Magnitude) (R_L = 50 Ω) Carrier Input Port, V_C = 60 mVrms sine wave f_S = 1.0 kHz, 300 mVrms sine wave Signal Input Port, V_S = 300 mVrms sine wave $\lvert V_C \rvert$ = 0.5 Vdc	8	8	BW_{3dB}	 – –	 300 80	 – –	MHz
Signal Gain (V_S = 100 mVrms, f = 1.0 kHz; $\lvert V_C \rvert$= 0.5 Vdc)	10	3	A_{VS}	2.5	3.5	–	V/V
Single–Ended Input Impedance, Signal Port, f = 5.0 MHz Parallel Input Resistance Parallel Input Capacitance	6	–	 r_{ip} c_{ip}	 – –	 200 2.0	 – –	 kΩ pF
Single–Ended Output Impedance, f = 10 MHz Parallel Output Resistance Parallel Output Capacitance	6	–	 r_{op} c_{oo}	 – –	 40 5.0	 – –	 kΩ pF
Input Bias Current $I_{bS} = \frac{I1 + I4}{2}$; $I_{bC} = \frac{I8 + I10}{2}$	7	–	 I_{bS} I_{bC}	 – –	 12 12	 30 30	µA
Input Offset Current I_{ioS} = I1–I4; I_{ioC} = I8–I10	7	–	 $\lvert I_{ioS} \rvert$ $\lvert I_{ioC} \rvert$	 – –	 0.7 0.7	 7.0 7.0	µA
Average Temperature Coefficient of Input Offset Current (T_A = –55°C to +125°C)	7	–	$\lvert TC_{Iio} \rvert$	–	2.0	–	nA/°C
Output Offset Current (I6–I9)	7	–	$\lvert I_{oo} \rvert$	–	14	80	µA
Average Temperature Coefficient of Output Offset Current (T_A = –55°C to +125°C)	7	–	$\lvert TC_{Ioo} \rvert$	–	90	–	nA/°C
Common–Mode Input Swing, Signal Port, f_S = 1.0 kHz	9	4	CMV	–	5.0	–	Vpp
Common–Mode Gain, Signal Port, f_S = 1.0 kHz, $\lvert V_C \rvert$= 0.5 Vdc	9	–	ACM	–	–85	–	dB
Common–Mode Quiescent Output Voltage (Pin 6 or Pin 9)	10	–	V_{out}	–	8.0	–	Vpp
Differential Output Voltage Swing Capability	10	–	V_{out}	–	8.0	–	Vpp
Power Supply Current I6 +I12 I14	7	6	I_{CC} I_{EE}	– –	2.0 3.0	4.0 5.0	mAdc
DC Power Dissipation	7	5	P_D	–	33	–	mW

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Figure 6-10 - Continued

Filtering Out the Undesired Sideband

Once a DSB-SC signal has been created, the unwanted sideband must be removed. Since the upper and lower sidebands occupy different frequency ranges, a bandpass filter can be used to allow only the desired sideband through. The result will be a SSB signal.

Example 6-5

The SSB transmitter of Figure 6-6 is to transmit a speech signal (300-3000 Hz) on a *suppressed carrier frequency* of 8.000 MHz.

a)What will the frequency of the carrier oscillator be?
b) What will the passband of the USB filter need to be?
c) What will the passband of the LSB filter need to be?
d) Comment on the bandwidth needed for the LSB and USB filters.

Solution
The carrier oscillator operates at the suppressed carrier frequency, so by inspection, f_c=8.000 MHz

To calculate the USB filter passband, use the information in Figure 6-3. The limits of the passband can be calculated as:

$$f_{\min} = f_c + f_{m(\min)} = 8.000MHz + 300Hz = \underline{\underline{8.0003\ MHz}}$$

The upper limit is calculated in the same way:

$$f_{\max} = f_c + f_{m(\max)} = 8.000MHz + 3000Hz = \underline{\underline{8.0030\ MHz}}$$

The USB filter needs to pass 8.0003 MHz to 8.0030 MHz.
The LSB filter frequencies are calculated in the opposite manner:

$$f_{\max} = f_c - f_{m(\min)} = 8.000MHz - 300Hz = \underline{\underline{7.9997\ MHz}}$$

$$f_{\min} = f_c - f_{m(\max)} = 8.000MHz - 3000Hz = \underline{\underline{7.997\ MHz}}$$

The LSB filter must pass 7.997 MHz to 7.9997 MHz.
The bandwidth of the USB filter is (8.003 MHz - 8.0003 MHz), or 2.7 kHz.
The LSB filter bandwidth is (7.9997 MHz - 7.997 MHz), or 2.7 kHz.

Both USB and LSB filters use the same bandwidth, which is the same as the bandwidth of the information signal.

The job of filtering out the undesired sideband is actually much more difficult in terms of practical circuit components than it might initially seem. This is because a fairly precise filter bandpass *shape* is required. Attaining the proper shape requires tuned circuits with a very high Q; the Q required can be more than 5,000. The ideal and typical bandpass curves of the USB filter for Example 6-5 is shown in Figure 6-11.

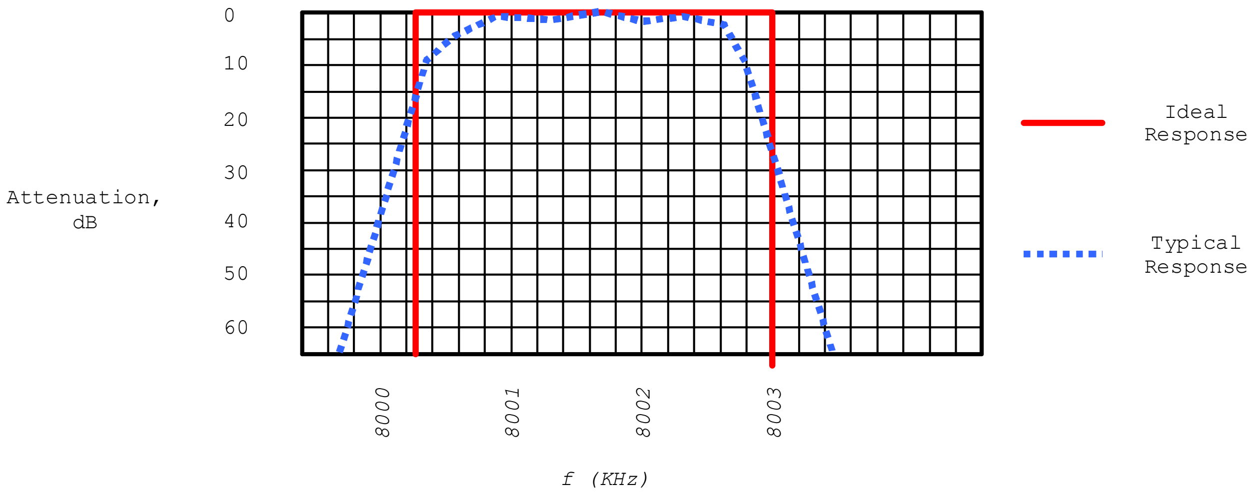

Figure 6-11: Ideal and Typical Responses for the USB Filter of Example 6-5

In the filter response curves of Figure 6-11, which can be obtained by connecting the filter under test to a *network analyzer*, the vertical scale is in units of *decibel attenuation*. Recall that to attenuate a signal is to make it weaker. A zero dB attenuation means that the signal is not made weaker at all; it passes with no power loss. The higher the attenuation, the greater the power loss.

The solid red line represents the *ideal* frequency response of the SSB filter. The ideal filter has infinite attenuation outside the desired passband (8000.3 kHz - 8003 kHz), and *zero* passband attenuation. The resulting ideal response is *rectangular*.

In real life, there's no such thing as a perfect filter. The blue response curve is typical of a real filter. Note how the *skirts* (the sloped portions on the left and right) aren't perfectly vertical. The frequency response within the *passband* (the top middle portion) isn't exactly "flat" either; there is a small amount of gain variation called *passband ripple*. The passband ripple is about 2 dB in this example.

The "goodness" of a filter is often expressed as the filter's *shape factor*, or SF. The shape factor of a filter is defined as:

$$\text{(6-3)} \quad SF = \frac{BW_{(-60dB)}}{BW_{(-6dB)}}$$

The shape factor is the ratio of the bandwidth at 60 dB attenuation to the bandwidth at 6 dB attenuation.

Example 6-6

Calculate the shape factor for the ideal and typical filter curves from Figure 6-11.

Solution

For the ideal curve, the bandwidths at 6 and 60 dB attentuation are *identical*, and from the graph, these bandwidths can be read as 2.7 kHz (8,003 kHz - 8000.3 kHz). Therefore, the shape factor of the ideal filter is:

$$SF = \frac{BW_{(-60dB)}}{BW_{(-6dB)}} = \frac{2.7KHz}{2.7KHz} = \underline{\underline{1:1}}$$