CHAPTER

8 Suppressed-carrier AM and quadrature AM (QAM)

Viewed just in the time domain, ordinary AM, as used in broadcasting, seems so obvious that one would scarcely imagine how it might be done otherwise. But viewed in the frequency domain, as in Chapter 6, this system shows some obvious inefficiencies. First, most of the average transmitted power (about 95% when transmitting typical audio material) is in the carrier, which is a spike or "delta function" in the frequency domain. Since its amplitude and frequency are constant, it carries virtually no information. The information is in the sidebands. Could broadcasters just suppress the carrier to reduce their electric power costs by 95%? Second, the upper and lower sidebands are mirror images of each other, so they contain the same information. Could they not suppress (filter away) one sideband, making room for twice as many stations on the AM band? The answer to both questions is yes but, in both cases, the simple AM receiver, with its envelope detector, will no longer work properly. Economics favored the simplicity of the traditional AM receiver until it because possible to put all the receiver signal processing on an integrated-circuit chip, where the additional complexity can have negligible cost. In this chapter we examine alternate AM systems that remove the carrier and then at AM systems that reduce the signal bandwidth or double the information carried in the original bandwidth.

8.1 Double-sideband suppressed-carrier AM

Let us look at a system that removes the carrier at the transmitter and regenerates it at the receiver. It is easy enough to modify the transmitter to eliminate the carrier. Review the circuit diagram given previously for an AM transmitter (Figure 6.2a). If we replace the bias battery by a zero-volt battery (a wire), the carrier disappears. The resulting signal is known as *Double-Sideband Suppressed-Carrier* (DSBSC). To restore the missing carrier we might try the receiver circuit shown in Figure 8.1. The locally generated carrier, from a *beat frequency oscillator* (BFO), is simply added back into the IF signal, just ahead

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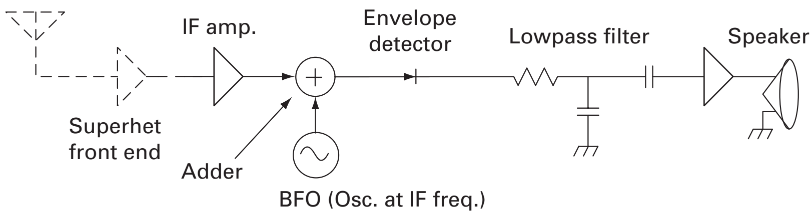


Figure 8.1. Hypothetical DSBSC receiver: additive carrier reinsertion and envelope detection.

of the envelope detector. The energy saved by suppressing the carrier can increase battery lifetime in walkie-talkies by a factor of maybe 20. The modulation schemes used in many cell phones (after the first generation) likewise do away with battery-draining constant carriers.

In a superheterodyne receiver, we only have to generate this carrier at one frequency, the intermediate frequency (IF). As you would expect, the added carrier must have the right frequency. But it must also have the right phase. Suppose, for example, that the modulation is a single audio tone at 400 Hz. If the replacement carrier phase is off by 45°, the output of the envelope detector will be severely distorted. And if the phase is off by 90° the output of the detector will be a tone at 800 Hz (100% distortion!). These example waveforms are shown in Figure 8.2.

Correct carrier reinsertion is fairly easy if the transmitter provides a low-amplitude "pilot" carrier to provide a phase reference. In the model transmitter of Figure 6.2(a) we would go back to using a bias battery, but it would have a low voltage. In principle, the receiver could have a tuned IF amplifier with a narrowband gain peak at the center frequency to bring the pilot carrier up to full level. The most common method, however, is to use a VCO for the BFO in Figure 8.1, together with a feedback circuit to phase lock it to the pilot carrier.

8.2 Single-sideband AM

The requirement for correct BFO phasing is not critical if only one of the two sidebands is transmitted. The sidebands are mirror images of each other so they carry the same information and one will do. We will discuss later three methods to eliminate one of the sidebands from a DSBSC signal, but the first and most obvious method is to use a bandpass filter to select the desired sideband. The resulting signal is known as *Single-Sideband Suppressed-Carrier* (SSBSC) or simply as *Single-Sideband* (SSB). Take the previous example of a single 400-Hz audio tone. The SSB transmitter will put out a single frequency, 400 Hz above the frequency of the (suppressed) carrier if we have selected the upper sideband. This signal will appear in the receiver 400 Hz away from the IF center frequency. Suppose we are still using the BFO and envelope detector. The

Demonstration that Detection of Double Sideband A.M. Requires the Correct Carrier Phase

$t := 0, 0.1..\,300$

carrier(t) := sin(t)

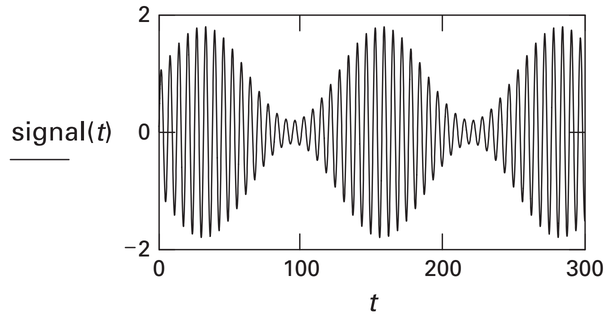
audio(t) := 0.8·sin(0.05·t) Audio is sine wave at 1/20 carrier frequency

dssc(t) := audio(t)·carrier(t) Double sideband suppressed carrier (sidebands alone)

Note that dssc(t) = 1/2(cos(0.95t) – cos(1.05)t)

signal(t) := carrier(t) + dssc(t) Carrier plus sidebands = (1 + audio)(carrier) = normal AM

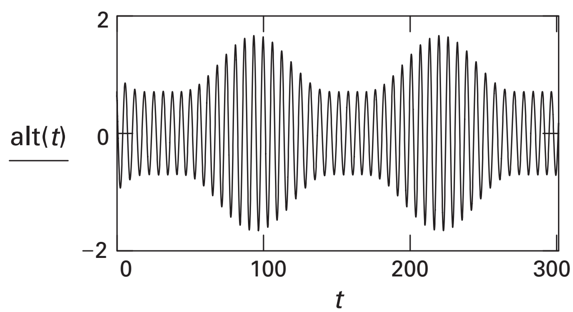
(a) Sidebands plus correct carrier. (note that the envelope is the sinewave audio)



alt(t) :=dssc(t) + cos(t + $\frac{\pi}{4}$) Give the re-injected carrier a 45° phase error.

(b) Sidebands with carrier 45° out of phase

(Note envelope distortion)



alt2(t) := dssc(t) + cos(t) Here the re-injected carrier has a 90° phase error.

(c) Sidebands with carrier 90° out of phase

(Note that the envelope has twice the audio frequency – total distortion)

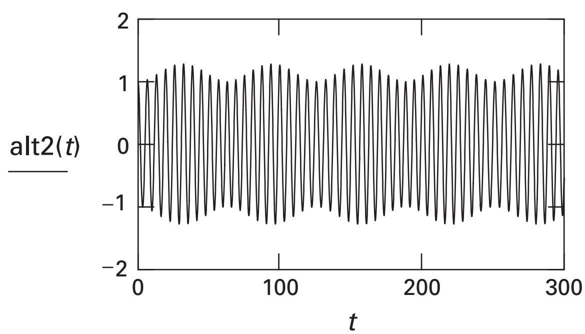


Figure 8.2. Detection of DSSC by adding a local carrier.

envelope of the signal-plus-BFO will indeed be an undistorted sine wave at 400 Hz. So far, so good. But suppose the signal had been two tones, one at 400 Hz and one at 600 Hz. When the BFO carrier is added, the resulting envelope will have tones at 400 and 600 Hz but it will also have a component at 200 Hz (600 − 400) which is distortion. Unwanted signals are produced by the IF components beating with each other; their strength is proportional to the product of the two IF signals. The strength of each wanted component, on the other hand, is proportional to the product of its amplitude times the amplitude of the BFO. If the BFO amplitude is much greater than the IF signal the distortion can be reduced.

8.3 Product detector

The distortion described above can be avoided by using a *product detector* instead of an envelope detector. In the receiver shown in Figure 8.3, the product detector is just the familiar multiplier (a.k.a. mixer).

The output of this detector is, by the distributive law of multiplication, the sum of the products of each IF signal component times the BFO signal. There are no cross-products of the various IF signal components. Product detectors are also called "baseband mixers" or "baseband detectors," because they translate the sideband components all the way down to their original audio frequencies.

While a product detector does not produce cross-products (intermodulation distortion) of the IF signal components, the injected carrier should ideally have the correct phase. The wrong phase is of no consequence in our single-tone example. But when the signal has many components, their relative phases are important. A waveform will become distorted if all its spectral components are given an identical phase shift (see Problem 8.3). It is only when every component is given a phase shift proportional to its frequency that a waveform is not distorted but only delayed in time. Therefore a single-sideband transmitter, like the double-sideband transmitter, should really transmit a pilot carrier and the receiver should lock its BFO phase to this pilot. Some SSB systems do just that. But for voice communications it is common to use no pilot. Speech remains intelligible and almost natural-sounding even when the BFO phase is

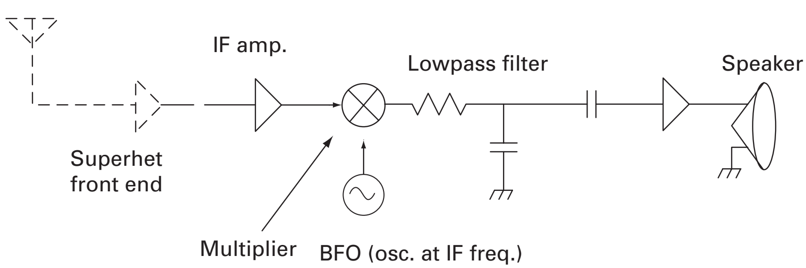


Figure 8.3. SSBSC receiver using a product detector.

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wrong.[1] The BFO can even be slightly off frequency. When the frequency is too high, a demodulated upper sideband (USB) voice signal has a lower than natural pitch. When it is too low, the pitch is higher than natural.

Finally, what happens if we try to use a product detector with a free-running (not phase locked) BFO to receive DSBSC? If the BFO phase is off by 90°, the detector produces no output. If the BFO phase happens to be correct, the output will have maximum amplitude. For an intermediate-phase error the amplitude will be reduced.

Advantages of SSB

Single-sideband, besides not wasting power on a carrier, uses only half the bandwidth, so a given band can hold twice as many channels. Halving the receiver bandwidth also halves the background noise so there is a 3-dB improvement over conventional AM in signal-to-noise ratio. When a spectrum is crowded with conventional AM signals, their carriers produce annoying beat notes in a receiver (a carrier anywhere within the receiver passband appears to be a sideband belonging to the spectrum of the desired signal). With single-sideband transmitters there are no carriers and no beat notes. Radio amateurs, the military, and aircraft flying over oceans use SSB for short-wave (1.8–30 MHz) voice communication.

8.4 Generation of SSB

There are at least three well-known methods to generate single-sideband. In the filter method of Figure 8.4, a sharp bandpass filter removes the unwanted sideband. This filter usually uses crystal or other high-Q mechanical resonators and has many sections. It is never tunable, so the SSB signal is generated at a single frequency (a transmitter IF frequency) and then mixed up or down to the desired frequency. The filter, in as much as it does not have infinitely steep skirts, will cut off some of the low-frequency end of the voice channel.

The phasing method uses two multipliers and two phase-shift networks to implement the trigonometric identity:

$$\cos(\omega_c t)\cos(\omega_a t) \pm \sin(\omega_c t)\sin(\omega_a t) = \cos([\omega_c \mp \omega_a]t). \tag{8.1}$$

The subscripts “a” and “c” denote the audio and (suppressed) carrier frequencies. If the audio signal is $\cos(\omega_a t)$, we have to generate the $\sin(\omega_a t)$ needed for the second term. A 90° audio phase shift network can be built with passive components or digital signal processing techniques. The audio phase-shift network is usually implemented with two networks, one ahead of each mixer. Their phase *difference* is close to 90° throughout the audio band. The second term also

[1] Human perception of speech and even music seems to be remarkably independent of phase. When an adjustable allpass filter is used to produce arbitrary phase vs. frequency characteristics (while not affecting the amplitude), test subjects find it difficult or impossible to distinguish phase-modified audio from the original source.

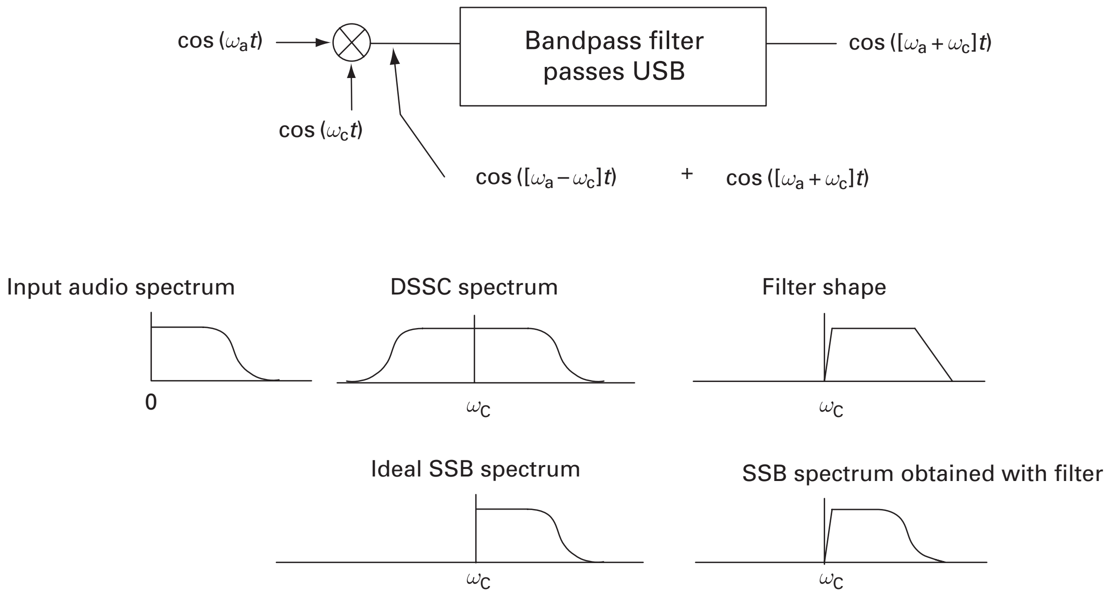


Figure 8.4. SSB generator – filter method.

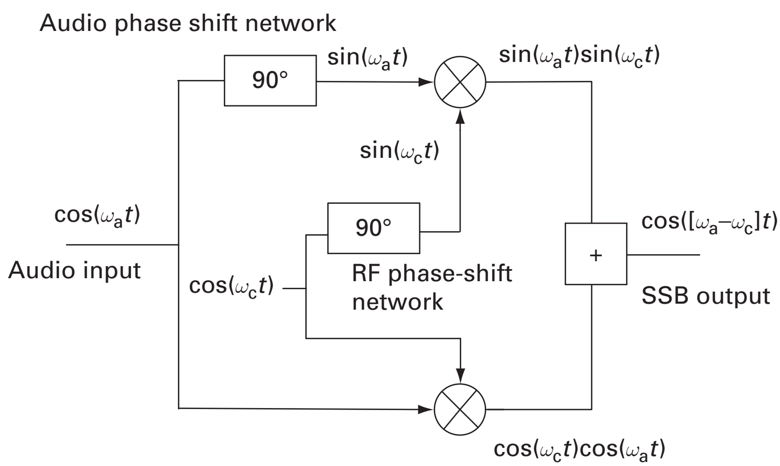


Figure 8.5. SSB Generator – phasing method.

requires that we supply sin($\omega_c t$) but, since ω_c is fixed, anything that provides a 90° shift at this one frequency will suffice. Figure 8.5 illustrates the phasing method used to generate an upper sideband (USB) signal.

If the adder is changed to a subtractor (by inverting the polarity of one input), the output will be the upper rather than the lower sideband.

A third method [2] needs neither a sharp bandpass filter nor phase-shift networks. Weaver's method, shown in Figure 8.6, uses four multipliers and two lowpass filters. The trick is to mix the audio signal with a first oscillator whose frequency, ω_0, is in the center of the audio band. The outputs of the first

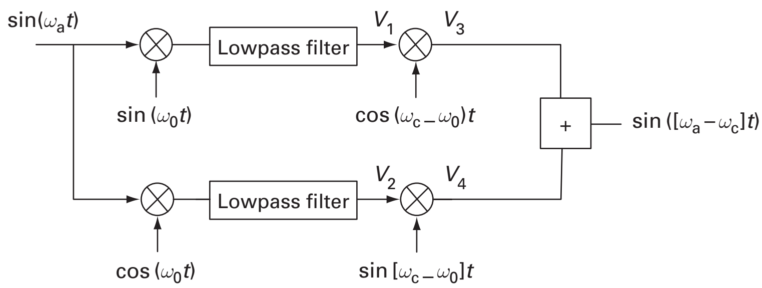


Figure 8.6. SSB generator – Weaver method.

set of mixers have the desired 90° phase difference. The second set of mixers then works the same way as the two mixers in the phasing method.

Referring to Figure 8.6,

$$V_1(t) = \text{lowpass } [\sin(\omega_a t)\sin(\omega_0 t)] = {}^1\!/\!_2 \cos([\omega_a - \omega_0]t) \tag{8.2}$$

$$V_2(t) = \text{lowpass } [\sin(\omega_a t)\cos(\omega_0 t)] = {}^1\!/\!_2 \sin([\omega_a - \omega_0]t) \tag{8.3}$$

$$\begin{aligned} V_3(t) &= {}^1\!/\!_2 \cos[(\omega_c - \omega_0)t]\cos[(\omega_a - \omega_0)t] \\ &= {}^1\!/\!_4 \cos([\omega_a - \omega_c]t) + {}^1\!/\!_4 \cos([\omega_a + \omega_c - 2\omega_0]t) \end{aligned} \tag{8.4}$$

$$\begin{aligned} V_4(t) &= {}^1\!/\!_2 \sin[(\omega_c - \omega_0)t]\sin[(\omega_a - \omega_0)t] \\ &= {}^1\!/\!_4 \cos[(\omega_a - \omega_c)t] - {}^1\!/\!_4 \cos[(\omega_a + \omega_c - 2\omega_0)t] \end{aligned} \tag{8.5}$$

$$V_{\text{ou}}\text{t}(t) = V_3(t) + V_4(t) = {}^1\!/\!_2 \cos\,[(\omega_c - \omega_a)t]. \tag{8.6}$$

We see from Equation (8.6) that this particular arrangement generates the lower sideband.

8.5 Single-sideband with class C, D, or E amplifiers

The three methods described above for generating a single-sideband signal are done in low-level circuitry. Linear amplifiers then produce the required power for the antenna. There is, however, a way to use a class-C or class-D amplifier with simultaneous phase modulation and amplitude modulation to produce SSB. This follows from the fact that any narrowband signal (CW, AM, FM, PM, SSB, DSB, narrowband noise, etc.) is essentially a sinusoid at the center frequency, ω_0, whose phase and amplitude vary on a time-scale longer than $(\omega_0)^{-1}$. Suppose we have generated SSB at low level. We can envelope-detect it, amplify it, and use the amplified envelope to AM-modulate the class-C or class-D amplifier. At the same time we can phase-modulate the amplifier with the phase determined from the low-level SSB signal. The low-level signal is simply amplitude-limited and then used to drive the modulated amplifier. The point is that the high-power SSB signal is produced with an amplifier that has close to 100% efficiency. Moreover, an existing AM transmitter can be converted to single-sideband by making only minor modifications.

8.6 Quadrature AM (QAM)

We noted above that a product detector used for DSBSC will produce no output if the BFO phase is wrong by 90°. It follows that two independent DSBSC signals can be transmitted over the same channel. At the receiver, the phase of the BFO selects one or the other. This is known as quadrature AM (QAM). Note that the carrier can be suppressed, but retaining at least a low-power pilot carrier provides a phase reference for the receiver. Figure 8.7 shows the transmitter (a) and receiver (b) for a QAM system. Here, the receiver uses a narrowband filter to isolate the pilot carrier and then amplifies it to obtain a reconstituted BFO or LO, depending on whether the receiver is a superheterodyne or uses direct conversion to baseband. In practice, carrier regeneration is almost always done with a phase lock loop.

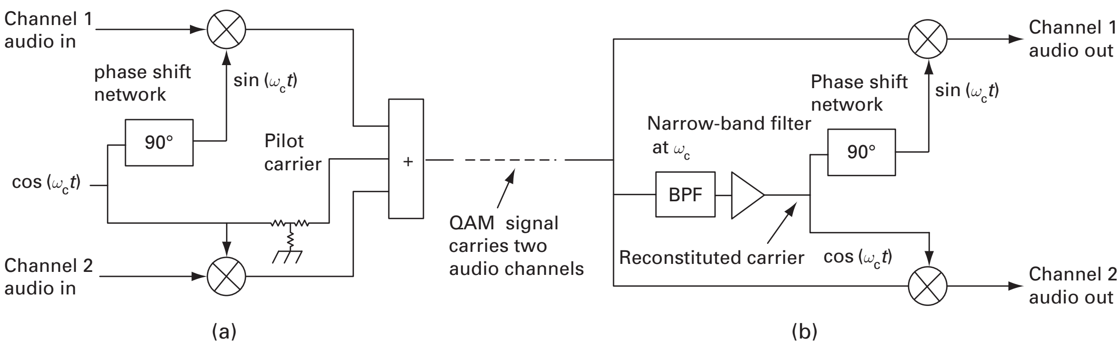


Figure 8.7. QAM:
(a) transmitter; (b) receiver.

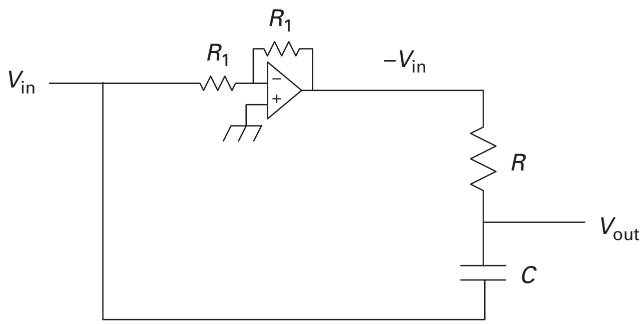
The U.S. NTSC and European PAL standards for color television use QAM to transmit a pair of video color difference signals. In the NTSC system, these so-called I and Q signals (In-phase and Quadrature) are multiplied by sine and cosine versions of a 3.579-MHZ oscillator signal and the resulting DSBSC signals are added to the normal video ("luminance") signal. Other uses of QAM include delivery of digital TV signals to set-top converters in some cable television systems and data distribution in some LANs.

The QAM system of Figure 8.7 could also be used to broadcast AM stereo sound, with both the L + R (left plus right) and L − R signals fitting into the channel used traditionally for L + R alone. However, such a QAM stereo signal would not be compatible with the envelope detector in standard AM receivers. Several compatible analog systems were developed and used for AM stereo, but at this writing, both AM and FM stations are adopting compatible *hybrid digital* (HD) systems in which the digital signals occupy spectral space on either side of the conventional analog signal. This system, which uses *COFDM* modulation, is discussed in Chapter 22.

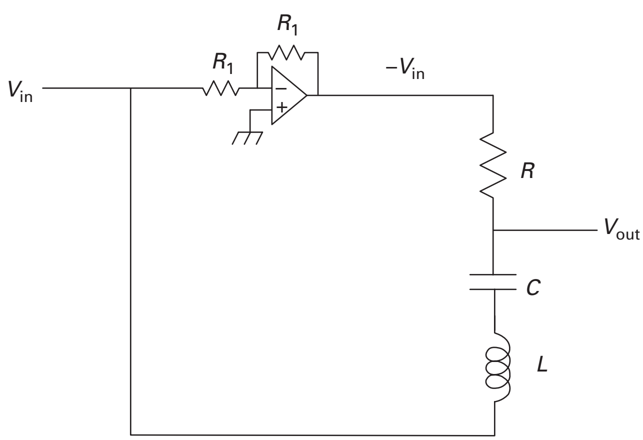
Problems

Problem 8.1. Demonstrate for yourself the kind of phase distortion that will occur when the BFO in a product detector does not have the same phase as the suppressed carrier. Use the Fourier decomposition of a square wave: $V(\mathrm{t}) = \sin(\omega t) + 1/3 \sin(3\omega t) + 1/5 \sin(5\omega t) + \ldots$. Have your computer plot $V(\theta) = \sin(\theta) + 1/3 \sin(3\theta) + 1/5 \sin(5\theta) + \cdots + 1/9 \sin(9\theta)$ for θ from 0 to 2π. Then plot $V'(\theta) = \sin(\theta{+}1) + 1/3 \sin(3\theta{+}1) + 1/5 \sin(5\theta{+}1) + \cdots + 1/9 \sin(9\theta{+}1)$, i.e., the same function but with every Fourier component given an equal phase shift (here 1 radian). Would you expect these waveforms to sound the same? Consider the case in which the phase shift is 180°. This just inverts the waveform. Is this a special case or is an inverted audio waveform actually distorted?

Problem 8.2. The phase-shift networks used in the phasing method of single-sideband generation have a flat amplitude vs. frequency response – they are known as *allpass* filters. Allpass filters have equal numbers of poles (all in the left-hand plane) and zeros (all in the right-hand plane). Find the amplitude and phase response of the network shown below, which is a first-order allpass filter. (Note that the inverting op-amp is used only to provide an inverted version of the input.)



Problem 8.3. The most general allpass filter can be obtained by cascading first-order allpass sections (Problem 8.2) with second-order allpass sections. Find the amplitude and phase response of the network shown below, a second-order allpass filter.



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