CHAPTER

6 Amplitude and frequency modulation

Modulation means *variation* of the amplitude or the phase (or both) of an otherwise constant sinusoidal RF *carrier wave* in order that the signal carry information: digital data or analog waveforms such as audio or video. In this chapter we look at pure amplitude modulation (AM) and pure frequency modulation (FM). Historically, these were the first methods to be used for communications and broadcasting. While still used extensively, they are giving way to modulation schemes, mostly digital, some of which amount to simultaneous AM and FM.

The simplest form of AM is *on/off keying*. This binary digital AM (full on is a data "1" and full off is a data "0") can be produced with a simple switch, originally a telegraph key in series with the power source or the antenna. The first voice transmissions used a carbon microphone as a variable resistor in series with the antenna to provide a continuous range of amplitudes. With AM, the frequency of the carrier wave is constant, so the zero crossings of the RF signal are equally spaced, just as they are for an unmodulated carrier. The simplest FM uses just two frequencies; the carrier has frequency f_0 for data "zero" and $f_0 + \Delta f$ for data "one." FM is usually generated by a VCO (voltage-controlled oscillator). For binary FSK (frequency shift keying), the control voltage has only two values: one produces f_0 and the other produces $f_0 + \Delta f$. FM broadcasting uses a continuous range of frequencies; the instantaneous frequency is determined by the amplitude of the audio signal. With FM, the amplitude of the carrier wave signal is constant. Figure 6.1 shows an unmodulated carrier wave, an AM-modulated wave, an FM-modulated wave, and a wave with simultaneous AM and FM modulation.

Phase modulation is a type of frequency modulation, since frequency is the time derivative of phase.

Amplitude and frequency (or phase) exhaust the list of carrier wave properties that can be modulated. The fractional bandwidth of RF signals is low enough that if one zooms in on a stretch of several cycles, the waveform is essentially sinusoidal and can be described by just its amplitude and frequency. Schemes such as single-sideband suppressed carrier (SSBSC), double-sideband

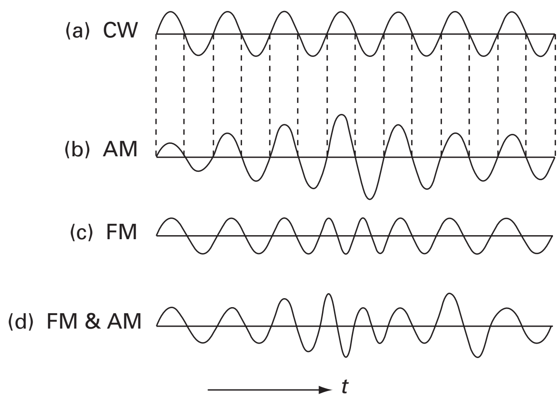


Figure 6.1. (a) Unmodulated wave; (b) with AM modulation; (c) with FM modulation; (d) with simultaneous AM and FM.

suppressed carrier AM (DSBSC), quadrature amplitude modulation (QAM), and others, produce simultaneous amplitude and frequency modulation. A sine-wave generator with provisions for both amplitude and phase modulation can, in principle, produce a radio signal with *any* specified type of modulation, for example, the comb of 52 individually modulated subcarriers used in Wi-Fi data links. Note: the term *digital modulation* means that the modulation uses a finite number of modulation states (often just two) rather than a continuum of states and that the time spent in any state is an integral multiple of a fundamental symbol time (baud).

6.1 Amplitude modulation

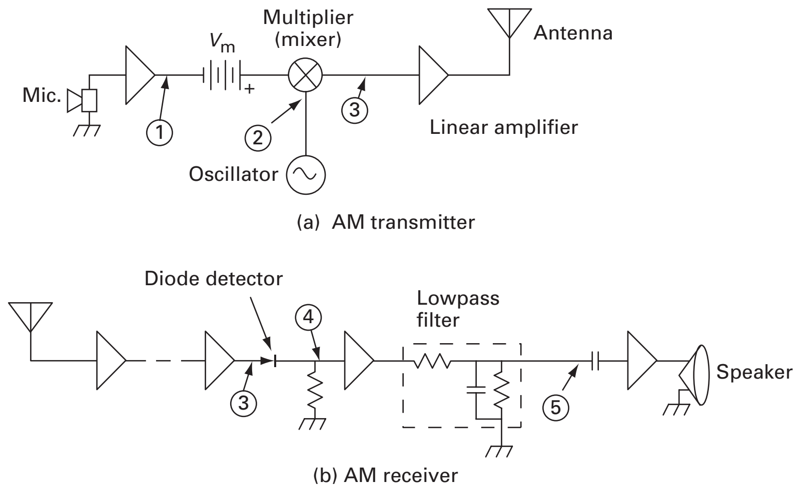
Amplitude modulation, the first scheme used for radio broadcasting, is still used in the long-wave, middle-wave, and short-wave broadcast bands.[1] Let us examine AM, first in the time domain and then in the frequency domain.

6.1.1 AM in the time domain

Without modulation (when the speech or music is silent) the voltage applied to the antenna is a pure sine wave at the carrier frequency. The power of a radio

[1] The long-wave (LW) band, used only outside the Western Hemisphere, extends from 153 to 179 kHz. The middle-wave (MW) band extends from 520 to 1700 kHz. Short-wave bands are usually identified by wavelength: 75 m, 60 m, 49 m, 41 m, 31 m, 25 m, 19 m, 16 m, 13 m and 11 m. The spacing between LW frequency assignments is 9 kHz. MW spacings are 10 kHz in the Western Hemisphere and 9 kHz elsewhere. Short-wave frequency assignments are less coordinated but almost all short-wave stations operate on frequencies which are an integral number of kHz.

Figure 6.2. Hypothetical AM transmitter and receiver.



station is defined as the transmitter output power when the modulation is zero. The presence of an audio signal changes the amplitude of the carrier. Figure 6.2 shows a hypothetical AM transmitter and receiver.

The audio signal (amplified microphone voltage) has positive and negative excursions but its average value is zero. Suppose the audio voltage is bounded by $+V_m$ and $-V_m$. A dc bias voltage of V_m volts is added to the audio voltage. The sum, $V_m + V_{audio}$, is always positive and is used to multiply the carrier wave, $\sin(\omega_c t)$. The resulting product is the AM signal; the amplitude of the RF sine wave is proportional to the biased audio signal. The simulation in Figure 6.3 shows the various waveforms in the transmitter and receiver of Figure 6.2 corresponding to a random segment of an audio waveform. The biased audio waveform is called the *modulation envelope*. Note that at full modulation where $V_{audio} = +V_m$, the carrier is multiplied by $2V_m$ whereas, at zero modulation, the carrier is multiplied by V_m (bias only). This factor of 2 in amplitude means the fully (100%) modulated signal has four times the power of the unmodulated signal (the carrier wave alone). It follows that the antenna system for a 50 000 W AM transmitter must be capable of handling 200 000 W peaks without breakdown. The average power of the modulated signal is determined by the average square of the modulation envelope. For example, in the case of 100% modulation by a single audio tone, the average power of the modulated signal is greater than the carrier by a factor of $\langle(1+\cos\theta)^2\rangle = \langle 1+2\cos(\theta) + \cos(\theta)^2\rangle = 1 + \langle\cos(\theta)^2\rangle = 3/2$.

Figure 6.2 also shows how the receiver demodulates the signal, i.e., how it recovers the modulation envelope from the carrier wave. The detector is just an ideal rectifier, which eliminates the negative cycles of the modulated RF signal. A simple lowpass filter then produces the average voltage of the positive loops. (Usually this is a simple *RC* filter, but the lowpass filter used in Figure 6.3 is a

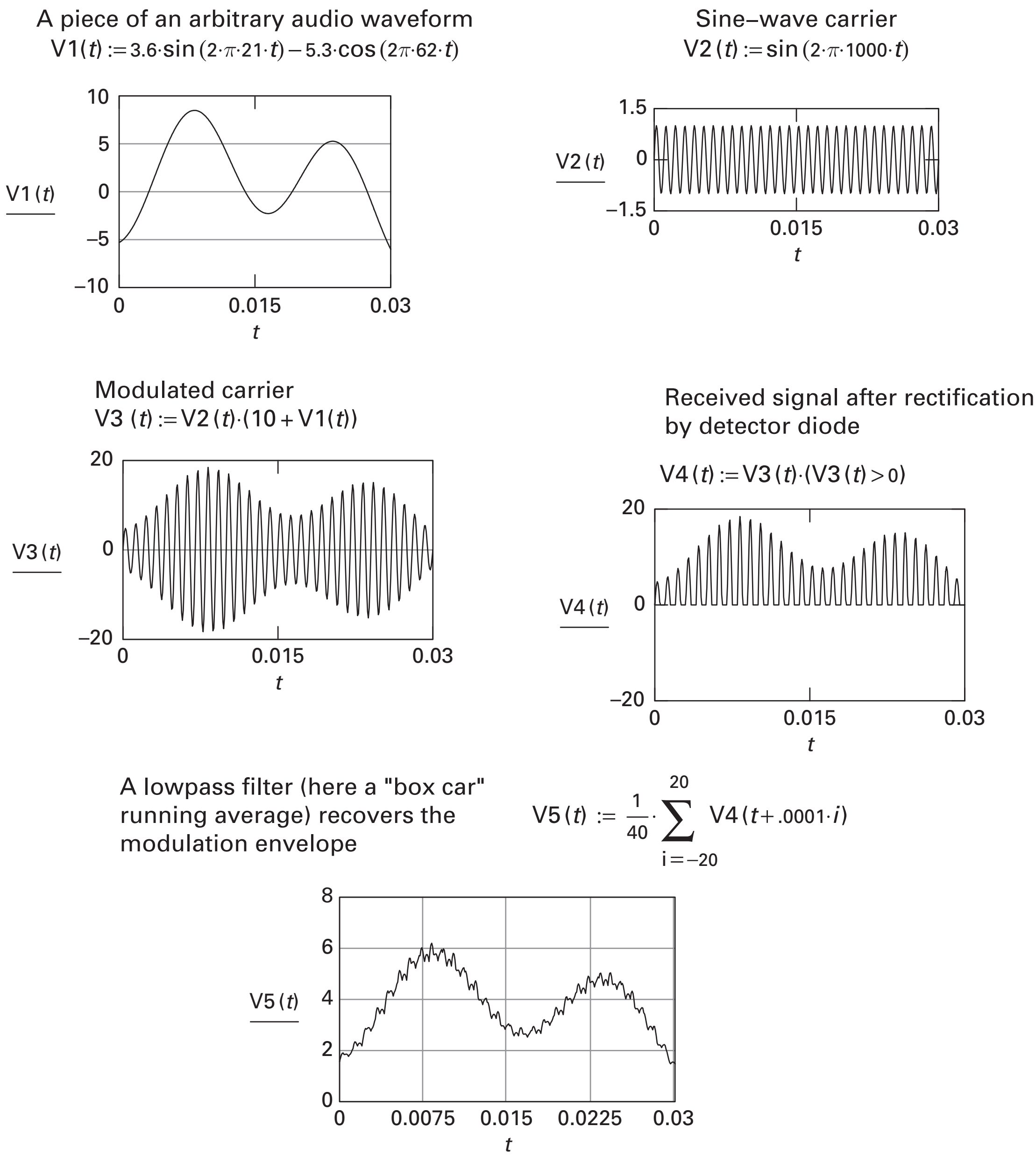


Figure 6.3. Waveforms in the AM system of Figure 6.2.

“boxcar” integrator that forms a running average.) Finally, ac coupling removes the bias, leaving an audio signal identical to the signal from the microphone.

6.1.2 AM in the frequency domain

Let us look at the spectrum of the AM signal to see how the power is distributed in frequency. This is easy when the audio signal is a simple sine wave, say V_a sin

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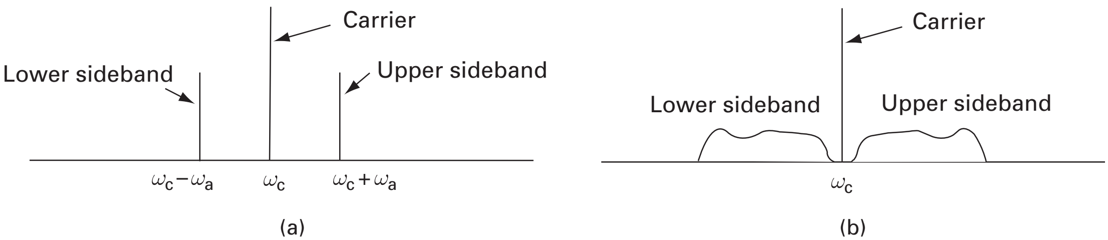

Figure 6.4. Spectrum of an AM power spectrum: (a) modulation by a single-tone modulation; (b) modulation by an audio spectrum.

$(\omega_a t)$. The voltage at point 3 in Figure 6.2 is then $\sin(\omega_c t)(V_m + V_a \sin(\omega_a t)) = V_m \sin(\omega_c t) + 1/2\ V_a \cos([\omega_c - \omega_a]t) - 1/2\ V_a \cos([\omega_c + \omega_a]t)$. These three terms correspond to the *carrier* at ω_c with power $V_m^2/2$, a *lower sideband* at $\omega_c - \omega_a$ with power $V_a^2/8$, and an *upper sideband* at $\omega_c + \omega_a$, also with power $V_a^2/8$. This spectrum is shown in Figure 6.4(a).

Note that the carrier, the component at ω_c, is always present and that its amplitude is independent of the modulation level. At 100% sine-wave modulation $V_a = V_m$ and the average power in each sideband is $\frac{1}{4}$ of the carrier power. The maximum average power is therefore $\frac{3}{2}$ times the carrier power, as we saw earlier from the time domain description. The sidebands, which carry all the information, account here for only $\frac{1}{3}$ of the total power transmitted. With speech waveforms the sidebands may contain even less power.

If, instead of a single sine wave, the audio signal is a superposition of sine waves with different frequencies (and amplitudes), the above analysis is readily extended to show that the upper and lower sidebands become continuous bands of frequency components straddling the carrier symmetrically, as shown in Figure 6.4(b). As the audio signal changes, the shape and size of the sidebands change. A typical audio signal has components at frequencies as high as 10 KHz. AM broadcast stations restrict their audio to not exceed 10 KHz but this still produces a 20 KHz-wide spectrum. The spacing between frequency assignments in the AM broadcast band is only 10 KHz. To prevent overlap, no two stations in a given listening area are assigned the same frequency or adjacent frequencies.

6.2 Frequency and phase modulation

In frequency modulation (FM) and phase modulation (PM) the audio voltage controls the phase angle of the sinusoidal carrier wave while the amplitude remains constant. Since frequency is the time derivative of phase, the two quantities cannot be varied independently; FM and PM are only slightly different methods of *angle modulation*. The advantage of FM (or PM) broadcasting over AM broadcasting is that, under strong signal conditions, the audio signal-to-noise ratio at the output of the receiver can be much higher for FM than for AM. The only price paid for this improvement is increased bandwidth (this topic is discussed in detail in Chapter 23).

6.2.1 FM in the time domain

A simple unmodulated carrier wave is given by $V(t)=\cos(\omega_c t+\phi_0)$ where ω_c and ϕ_0 are constants. The phase, $\phi(t)=(\omega_c t+\phi_0)$, of this cw signal increases linearly with time at a rate ω_c radians per second. In FM, the instantaneous frequency is made to shift away from ω_c by an amount proportional to the modulating audio voltage, i.e., $\omega(t)=\omega_0+k_{osc}V_a(t)$. A linear voltage-controlled oscillator (VCO) driven by the audio signal, as shown in Figure 6.5a, makes an FM modulator.

The excursion from the center frequency, $k_{osc}V_a$, is known as the (radian/s) *deviation*. In FM broadcasting the maximum deviation, defined as 100% modulation, is 75 kHz, i.e., $k_{osc}V_{a_{max}}/2\pi=75\cdot10^3$. Consider the case of modulation by a single audio tone, $V_a(t)=A\cos(\omega_a t)$. The instantaneous frequency of the VCO is then $\omega(t)=\omega_c+k_{osc}A\cos(\omega_a t)$. And, since the instantaneous frequency is the time derivative of phase, the phase is given by the integral

$$\phi(t)=\int\omega(t)\mathrm{d}t=\int(\omega_c+k_{osc}A\cos(\omega_a t))\mathrm{d}t=\omega_c t+\frac{k_{osc}A}{\omega_a}\sin(\omega_a t)+\phi_0. \tag{6.1}$$

From Equation (6.1), we see that, in addition to the linearly increasing phase, $\omega_c t$, of the unmodulated carrier, there is a phase term that varies sinusoidally at the audio frequency. The amplitude of this sinusoidal phase term is $k_{osc}A/\omega_a$. This maximum phase excursion is known as the *modulation index*. The inverse dependence on the modulation frequency, ω_a, results from phase being the integral of frequency. Phase modulation (PM) differs from FM only in that it does not have this ω_a denominator; it could be produced by a fixed-frequency oscillator followed by a voltage-controlled phase shifter. An indirect way to produce a PM signal is to use a standard FM modulator (i.e., a VCO), after first passing the audio signal through a differentiator. The differentiator produces a factor, ω_a, which cancels the ω_a denominator produced by the VCO. This scheme is shown in Figure 6.6.

Similarly, a PM transmitter can be adapted to produce FM. A possible receiver for PM would be the receiver of Figure 6.5(b) with an integrator after

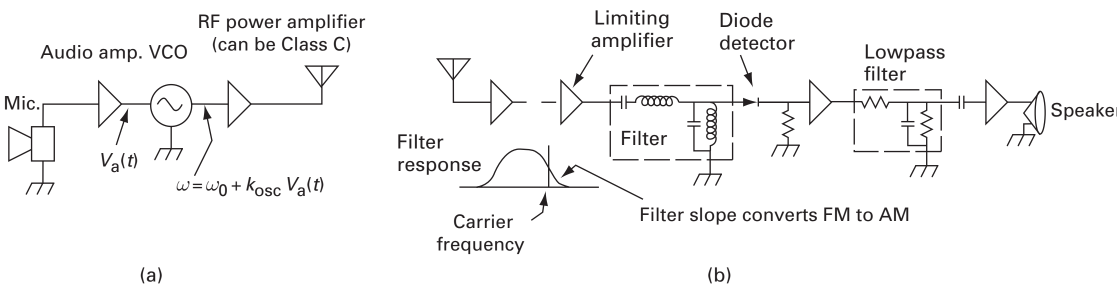

Figure 6.5. Basic FM system: (a) transmitter; (b) receiver.

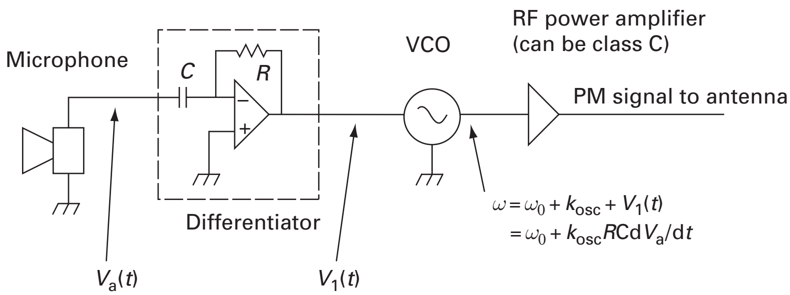

Figure 6.6. Phase modulation via frequency modulation.

the diode detector to undo the differentiation done in the transmitter of Figure 6.6. (See Chapter 18 for more FM and PM detectors.) In the simple FM receiver of Figure 6.5(b), a bandpass filter converts FM into AM because the carrier frequency is placed on the slope of the filter. When the carrier frequency increases, the filter response decreases the amplitude. After this filter, the rest of the receiver is an AM receiver. A limiting amplifier ahead of the filter is a refinement to eliminate amplitude noise, as well as to make the audio volume independent of signal strength. More refined AM and FM detectors are presented in Chapter 18.

6.2.2 Frequency spectrum of FM

Using the expression for phase from Equation (6.1) (neglecting ϕ_0) and using the usual expansion for cos(a+b), the complete FM signal (VCO output) for single-tone modulation becomes

$$V(t) = \cos(\phi(t)) = \cos(\phi_m \sin(\omega_a t))\cos(\omega_c t) - \sin(\phi_m \sin(\omega_a t))\sin(\omega_c t) \tag{6.2}$$

where $\phi_m = k_{osc}A/\omega_a$ is the modulation index. Let us first consider the case where the modulation index is small.

6.2.3 Narrowband FM or PM

At very low modulation levels, both FM and PM produce a power spectrum similar to AM, i.e., a carrier with an upper sideband ω_a above the carrier and a lower sideband ω_a below the carrier. To see this, we expand Equation (6.2) for very small ϕ_m, obtaining

$$V(t) \approx \left(1 - \frac{\phi_m^2 \sin^2(\omega_a t)}{2}\right)\cos(\omega_c t) - \phi_m \sin(\omega_a t)\sin(\omega_c t). \tag{6.3}$$

The first term is an almost constant amplitude spike at the carrier frequency, much like the carrier of an AM signal. You can verify directly that the amplitude of $V(t)$ remains constant by summing the squares of the coefficients of $\cos(\omega_c t)$ and $\sin(\omega_c t)$. Expanding the right-hand term gives the two sidebands,

$$V(t) \approx \cos(\omega_c t)\left(1 - \frac{\phi_m^2 \sin^2(\omega_a t)}{2}\right) + (\phi_m/2)\cos[(\omega_c + \omega_a)t] - (\phi_m/2)\cos[(\omega_c - \omega_a)t]. \tag{6.4}$$

This differs from an AM signal in that the sidebands are equivalent to having multiplied the audio signal not by $\cos(\omega_c t)$, the carrier, but by $\sin(\omega_c t)$. Note also that, in this limit of small modulation index, the amplitude of the sidebands is small and the width of the spectrum stays fixed at $2\omega_a$, just as with AM.

6.2.4 Wideband FM spectrum

Looking at Equation (6.2), we see that the signal consists of the products of $\cos(\omega_c t)$ and $\sin(\omega_c t)$, multiplied respectively by the baseband signals $\cos[\phi_m \sin(\omega_a t)]$ and $\sin[\phi_m \sin(\omega_a t)]$. When ϕ_m is not small, these products are similar and each consists of upper and lower sidebands around ω_c, with frequency components spaced from the carrier frequency by integral multiples of ω_a. The $\cos(\omega_c t)$ and $\sin(\omega_c t)$ carriers are suppressed and the spectrum has no distinct central carrier spike. An exact analysis uses Fourier analysis of the cos (sin) and sin(sin) terms[2] to find the comb of sidebands, but a simpler argument can give an estimate of the bandwidth of the FM signal. Referring again to Equation (6.2), the baseband modulating signals produce sidebands. In a time equal to one-quarter of an audio cycle, the expression $\phi_m \sin(\omega_a t)$ changes by ϕ_m radians, so the phase of the baseband modulating signals changes by ϕ_m radians. Dividing this phase change by the corresponding time interval, $1/(4f_a) = \pi/(2\omega_a)$, the average sideband frequency is given by $2\phi_m \omega_a/\pi$. Substituting $\phi_m = k_{osc} A/\omega_a$, the average frequency is $2k_{osc}A/\pi = (2/\pi) \times$ deviation. Therefore, the one-sided bandwidth of the FM signal, when the modulation index is large, is roughly equal to the maximum deviation and the full bandwidth is about twice the maximum deviation. If the deviation is reduced, the bandwidth goes down proportionally at first but then, in the narrowband regime, stays constant at twice the audio bandwidth, as shown above.

[2] The terms in Equation (6.2) can be expanded into harmonics of the audio frequency by using the Bessel function identities:

$$\cos(\phi \sin(\omega t)) = J_0(\phi) + 2\sum_{n=1}^{n=\infty} J_{2n}(\phi)\cos(2n\omega t)$$

and

$$\sin(\phi \sin(\omega t)) = 2\sum_{n=0}^{n=\infty} J_{2n+1}(\phi)\sin((2n+1)\omega t).$$

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6.2.5 Frequency multiplication of an FM signal

When a signal is passed through a times-N frequency multiplier its phase is multiplied by N. A square-law device, for example, can serve as a frequency doubler since $\cos^2(\phi) = 1/2 + 1/2\cos(2\phi)$. So if the phase of an FM signal, ϕ, (the right-hand expression in Equation 6.1), is multiplied by 2 by using a frequency doubler, both ω_c and kA/ω_a are multiplied by 2, i.e., the frequency and the modulation index are doubled. When a given VCO cannot be linearized well enough within the intended operating range, it can be operated with a low deviation and its signal can be frequency-multiplied to increase the deviation. If the resulting center frequency is too high, an ordinary mixer can shift it back down, preserving the increased deviation.

6.3 AM transmitters

The simple AM transmitter shown in Figure 6.2 is entirely practical. (Of course we would fix it up so that a battery would not be needed to supply the bias voltage.) The linear RF power amplifier would be the major part of this transmitter. We saw in Chapter 3 that a class-B linear amplifier has relatively high efficiency, $\pi/4$ or 78.5%, but that is for a maximum-amplitude sine wave. Let us find the efficiency for the transmitter of Figure 6.2, assuming the average modulation power is 10% of the peak modulation power. (The "crest" factor for speech is often taken to be 16 (12 dB), but broadcasters normally use dynamic range compression to increase "loudness.") Let the output signal be given by

$$V_{\text{out}} = V_{\text{dc}}\tfrac{1}{2}[1 + V_{\text{m}}(t)]\cos(\omega t) \tag{6.5}$$

where $|V_{\text{m}}(t)| \leq 1$. Note that the maximum output voltage is V_{dc}, the power supply voltage. (Refer to the amplifier circuit of Figures 3.7 or 3.8.) Remembering that $\langle\cos^2\rangle = 1/2$ and assuming that the audio signal is an ac signal, i.e., $\langle V_{\text{m}}(t)\rangle = 0$, we can express the average output power:

$$\begin{aligned} P_{\text{out}} &= \langle V_{\text{out}}{}^2\rangle/R = \tfrac{1}{4}V_{\text{dc}}{}^2\langle[1 + V_{\text{m}}(t)]^2\rangle/(2R) \\ &= \tfrac{1}{4}V_{\text{dc}}{}^2(1 + 0.1)/(2R) = 1.1V_{\text{dc}}{}^2/(8R), \end{aligned} \tag{6.6}$$

where R is the load (antenna) resistance and where we have used the fact that $\langle\cos(\omega t)\rangle = 0$. The average power delivered by the dc supply is given by

$$\begin{aligned} \langle V_{\text{dc}}I_{\text{supply}}\rangle &= V_{\text{dc}}\langle|V_{\text{out}}(t)/R|\rangle \\ &= V_{\text{dc}} \cdot \tfrac{1}{2}V_{\text{dc}}\langle[1 + V_{\text{m}}(t)/R\rangle 2/\pi = V_{\text{dc}}^2/(\pi R), \end{aligned} \tag{6.7}$$

where we have used the fact that the average of a positive loop $\sin(x)$ is equal to $2/\pi$. Calculating the efficiency, we find

$$\eta = \frac{\text{power out}}{\text{power in}} = (1.1)/V_{\mathrm{dc}}{}^2/(8R) \div V_{\mathrm{dc}}^2/\pi = 1.1\pi/8 = 43.2\%. \qquad (6.8)$$

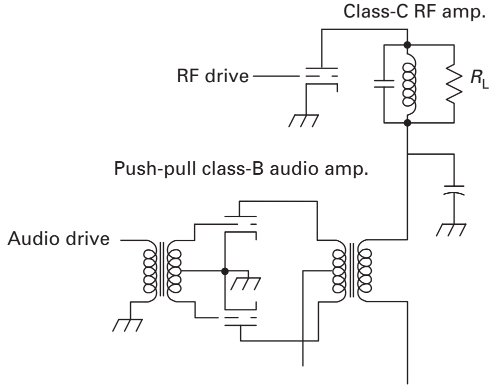
Therefore, when running this AM transmitter, only 43.2% of the prime power gets to the antenna; the rest is dissipated as heat in the class-B amplifier.

Almost all AM transmitters obtain higher efficiency by using class-C, D, or E RF amplifiers. These amplifiers are not linear in the normal sense, that is, the output signal amplitude is not a constant multiple of the input signal amplitude. But, for a fixed input RF amplitude, the output amplitude *is* proportional to the supply voltage. These amplifiers can therefore be used as high-power *multipliers* that form the product of the power supply voltage times a unit sine wave at the RF frequency. Furthermore, the efficiency of these amplifiers, which is high, is essentially independent of the supply voltage. (These amplifiers are discussed in detail in Chapter 9.) Let us look at the overall efficiency of transmitters using these amplifiers.

6.3.1 AM transmitter using a class-C RF amplifier and a class-B modulator

In the traditional AM transmitter, audio voltage developed by a high-power class-B audio amplifier (the modulator) is added to a dc bias and the sum of these voltages powers a class-C RF amplifier. As explained above, a class-C RF amplifier acts as a high-power multiplier. In the traditional tube-type circuit of Figure 6.7, the dc bias is fed through the secondary of the modulation transformer. Audio voltage produced by the modulator appears across the secondary winding and adds to the bias voltage. With no audio present, the class-B audio amplifier consumes negligible power and the bias voltage supply provides power for the carrier. At 100% sine-wave modulation, the modulator must supply audio power equal to half the bias supply power. A 50 000 watt transmitter thus requires a modulator that can supply 25 000 W of audio power. (Again, this result is for a single tone, but is essentially the same for speech or music.)

Figure 6.7. Class-B plate-modulated AM transmitter.



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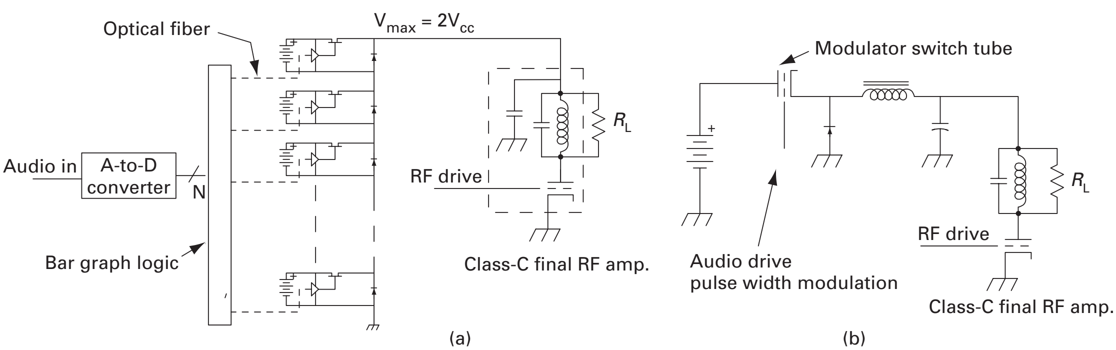
Let us find the efficiency of this transmitter. We will assume the class-C amplifier has an efficiency of 80% and that, as before, the average modulation power is 10% of the maximum possible modulation power. If the normalized output carrier power is 1 W, the dc bias supply must provide 1/0.8 W and the modulator must supply 0.1/0.08 W. To handle peaks, the maximum power from the modulator must be 1/0.08 W. To find the efficiency of the class-B modulator, we must know not only its peak output voltage, but also the mean of the absolute value of its output voltage. (Note that this last piece of information was not needed in the analysis of the class-B RF amplifier transmitter discussed above.) Let us just assume the modulating signal is a single audio sine wave whose power is 0.1/0.8 W. With that assumption, the modulator will have an efficiency of 0.35. The total input power, carrier plus modulation, will therefore be 1/0.8 +(0.1/0.35)/0.8 = 1.61 W. The output power will be 1 + 0.1, so the overall efficiency of this transmitter is 68%, a significant improvement over the transmitter using a linear class-B RF amplifier.

6.3.2 Class-C RF amplifier with a switching modulator

There are several newer methods to produce AM with even higher efficiency. All use switching techniques. The modulator shown in Figure 6.8(a) is just a high-power digital-to-analog converter. It uses solid-state switches to add the voltage of many separate low-voltage power supplies, rather than tubes or transistors to resistively drop the voltage of a single high-voltage supply. Modulators of this type can, in principle, be 100% efficient, since the internal switch transistors are either fully on or fully off. The modulator of Figure 6.8(b) is a pulse rate modulator whose output voltage is equal to the supply voltage multiplied by the duty factor of the switch tube.

This type of circuit, whose efficiency can also approach 100%, is discussed in detail with switching power supplies in Chapter 29. Again specifying an average modulation power of 10%, the combination of an 80% efficient class-C RF amplifier together with a switching modulator, makes an AM transmitter with an overall efficiency of 80%.

Figure 6.8. Class-C RF amplifier with (a) high-power D-to-A modulator and (b) duty cycle switching modulator.



6.3.3 Class-D or E amplifiers with switching modulators

Class D and class-E RF amplifiers can approach 100% efficiency. Like the class-C amplifier, the amplitude of their output sine wave is proportional to the dc supply voltage. When one of these amplifiers is coupled with a switching modulator, the resulting AM transmitters can approach an overall efficiency of 100% for any modulation level.

6.4 FM transmitters

Since FM modulation is generated by a VCO at a low level, and the signal has a constant amplitude, there is no other modulator nor is there any requirement that the final power amplifier have a linear amplitude response. Class-C amplifiers have been used most often, with efficiencies around 80%.

6.5 Current broadcasting practice

Many $\frac{1}{2}$ MW to 2 MW AM stations operate in the long-wave, medium-wave, and short-wave bands. These superpower transmitters use vacuum tubes. Standard AM broadcast band transmitters in the U.S. are limited to 50 kW. For this power and lower powers, new AM transmitters manufactured in the U.S. use transistors. These transmitters combine power from a number of modular amplifiers in the 1 kW range. The advantages of solid state include lower (safer) voltages, indefinite transistor life rather than expensive tube changes every year or two, and better “availability;” a defective module only lowers the power slightly and can be replaced while the transmitter remains on the air. Most FM transmitters over about 10 kW still use vacuum tubes, but solid-state FM transmitters are available up to about 40 kW.

Stereophonic sound was added to FM broadcasting around 1960. Existing monophonic receivers operated as they had, receiving the left + right (L + R) audio signal. Thus the system is (backwardly) *compatible*, just as color television was compatible with existing black and white television receivers. The L − R signal information is in the demodulated signal, but clustered around 38 kHz, above the audible range. The L − R signal uses double-sideband suppressed carrier AM modulation, which is explained in Chapter 8. The demodulator to recover the L-R signal is discussed in Chapter 18. Stereo was added to AM radio broadcasting around 1980. Several systems competed to become the standard, but the public was largely indifferent, having already opted for FM stereo. Compatible digital broadcasting has recently been introduced, with versions for both the AM and FM bands. In this IBOC (In-Band On-Channel) system, a station’s digital simulcast signal (which can contain different program

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material) shares the same channel with the standard AM or FM signal, mostly using the relatively empty spectral regions near (and somewhat past) the nominal edges of the channel. The digital signal, which can be produced by a separate transmitter and may even use a separate antenna, uses COFDM (Coded Orthogonal Frequency-Division Multiplexing), a modulation system, described in Chapter 22. This system is intended to be a stepping-stone to all-digital radio broadcasting in the traditional AM and FM bands. IBOC broadcasting equipment is equipped to transmit an all-digital mode, to be used if and when the traditional analog broadcasting is discontinued.

Problems

Problem 6.1. An AM transmitter, 100% modulated with an audio sine wave, has sidebands whose total power is equal to half the carrier power. Consider 100% modulation by an audio square wave. What is the ratio of sideband power to carrier power?

Problem 6.2. Suppose you are trying to listen to a distant AM station, but another station on the same frequency is coming in at about the same strength. Will you hear both programs clearly? If not, how will they interfere with each other?

Problem 6.3. During periods where the audio signal level is low, the amplitude of an AM signal varies only slightly from the carrier level. The modulation envelope, which carries all the information, rides on top of the high-power carrier. If the average amplitude could be decreased without decreasing the amplitude of the modulation, power could be saved. Discuss how this might be accomplished at the transmitter and what consequences, if any, it might have at the receiver.

Problem 6.4. Show how a PM transmitter can be used to generate FM.

Problem 6.5. Consider an FM transmitter modulated by a single audio tone. As the modulation level is increased, the spectral line at the carrier frequency decreases. Find the value of the modulation index that makes the carrier disappear completely.

Problem 6.6. One of the methods used for compatible AM stereo was a combined AM/PM system. The left + right (L + R) signal AM-modulated the carrier in the usual way, while the L − R signal was used to phase-modulate the carrier. Draw block diagrams of a transmitter and receiver for this system. Why would PM be preferable to FM for the L − R signal?