
بسم الله الرحمن الرحيم

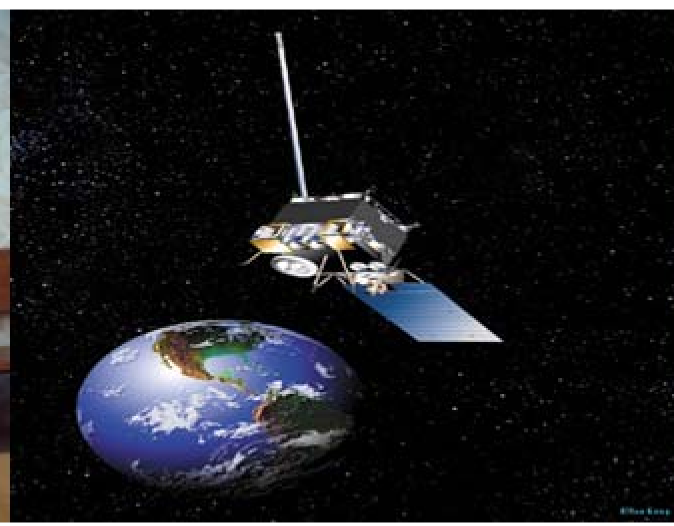
Analog and Digital Communication
(ET-323)

Lecture 8,9,10,11

(Chapter 4)

By

Engr. Haq Nawaz


AMPLITUDE MODULATION (AM)

Chapter Overview

- Introduction
- Baseband and Carrier Communication
- Amplitude Modulation
  - Double Sideband (DSB) and DSB-SC
  - Amplitude Modulation (AM) with carrier
  - Quadrature Amplitude Modulation (QAM)
  - Single Sideband (SSB)
  - Vestigial Sideband (VSB)
- Carrier Acquisition
- Superheterodyne AM Receiver
- Television

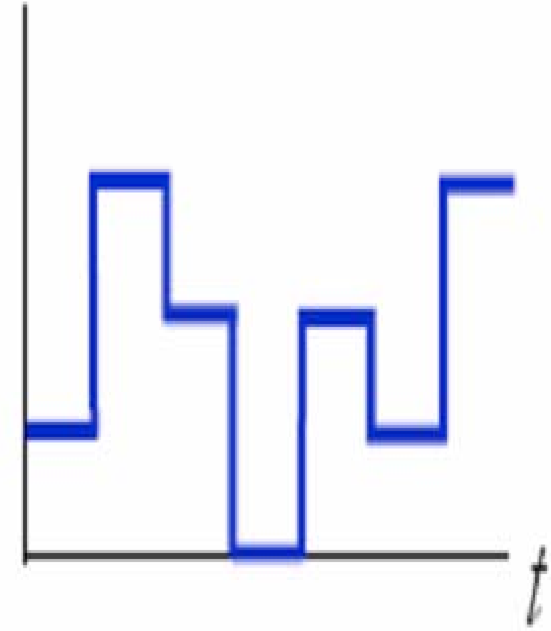
Introduction

- Analog signals are electrical replicas of the original signals such as audio and video.
- Analog signals may be converted into digital signals for transmission.
- Digital signals also originate in the form of computer and other data.
- In general, a digital signal is a coded version of the original data or analog signal.
- Modulation is a process that causes a shift in the range of frequencies in a signal.
- Modulation is used to gain certain advantages:
  - Ease of Radiation
  - Simultaneous Transmission of Several Signals
  - Effecting the Exchange of SNR with B
- Modulation is used to transmit analog as well as digital baseband signals.
- Before discussing modulation, it is important to distinguish between communication that does not use modulation (baseband communication) and communication that uses modulation (carrier communication).

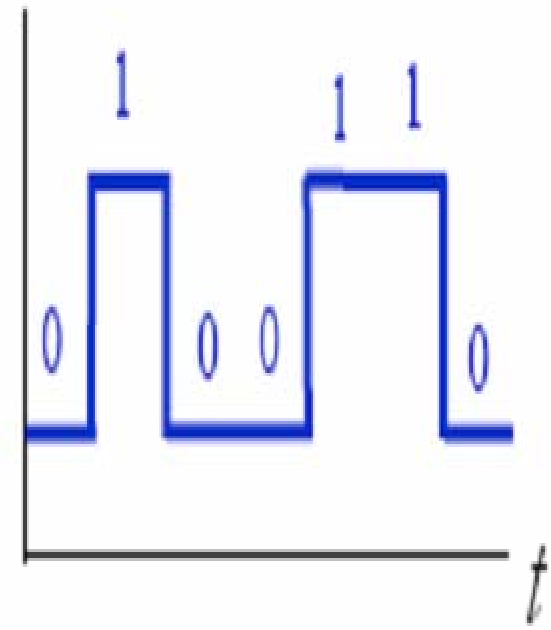
Baseband Communication

- The term baseband is used to designate the band of frequencies of the signal delivered by the source or the input transducer.
- In telephony, the baseband is the audio band (band of voice signals) of 0 to 4kHz.
- In television, the baseband is the video band occupying 0 to 4.5 MHz.
- For digital data or PCM using bipolar signaling at a rate of Rb pulses per second, the baseband is 0 to Rb Hz.
- In baseband communication, baseband signals are transmitted without modulation, that is, without any shift in the range of frequencies of the signal.
- Because the baseband signals have sizable power at low frequencies, they cannot be transmitted over a radio link but are suitable for transmission over a pair of wires, coaxial cables, or optical fibers.
- Local telephone communication, short-haul pulse-code modulation (PCM) (between two exchanges), and long- distance PCM over optical fibers use baseband communication.

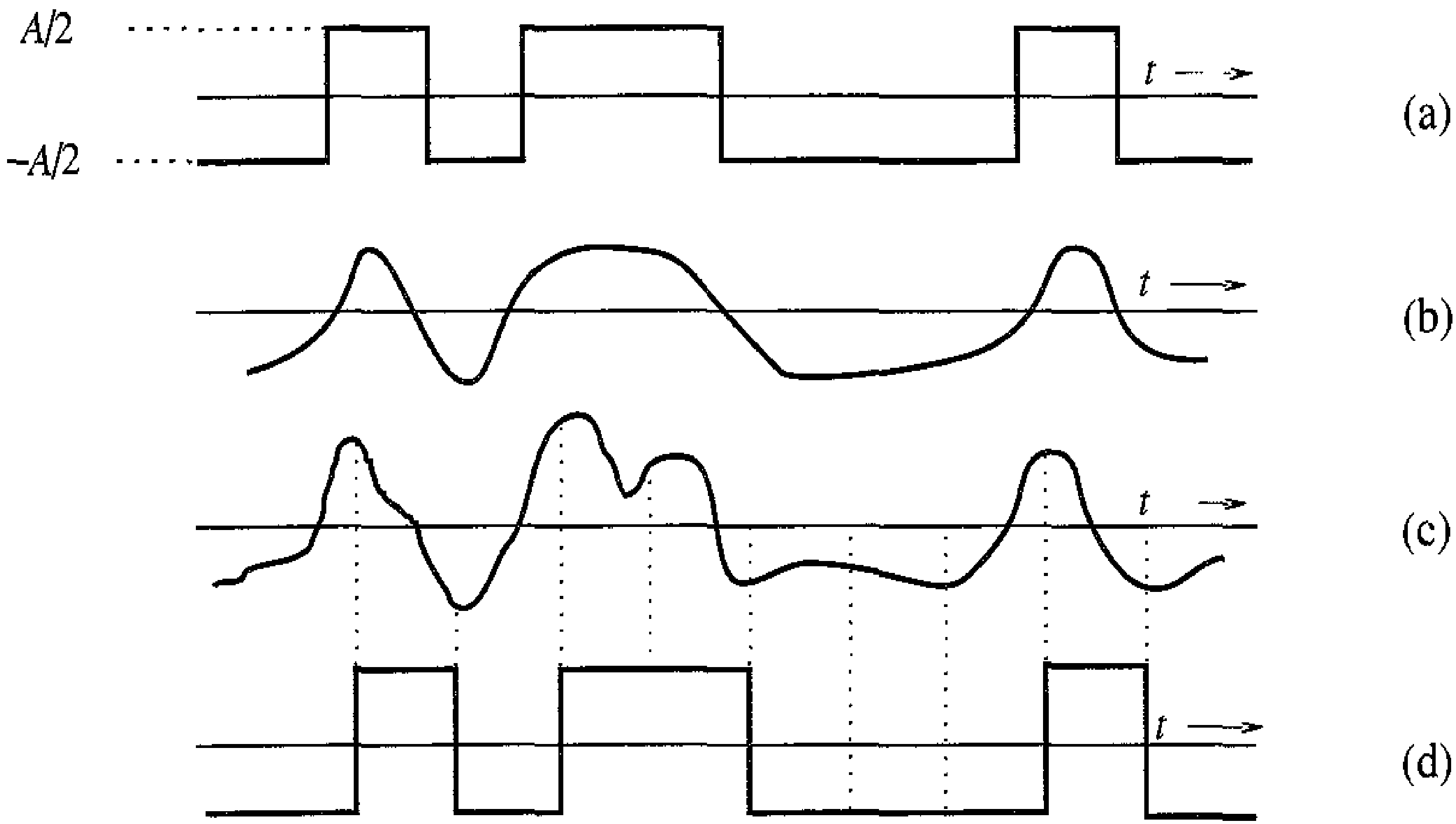
- Digital Signals
  - Values are taken from a discrete set

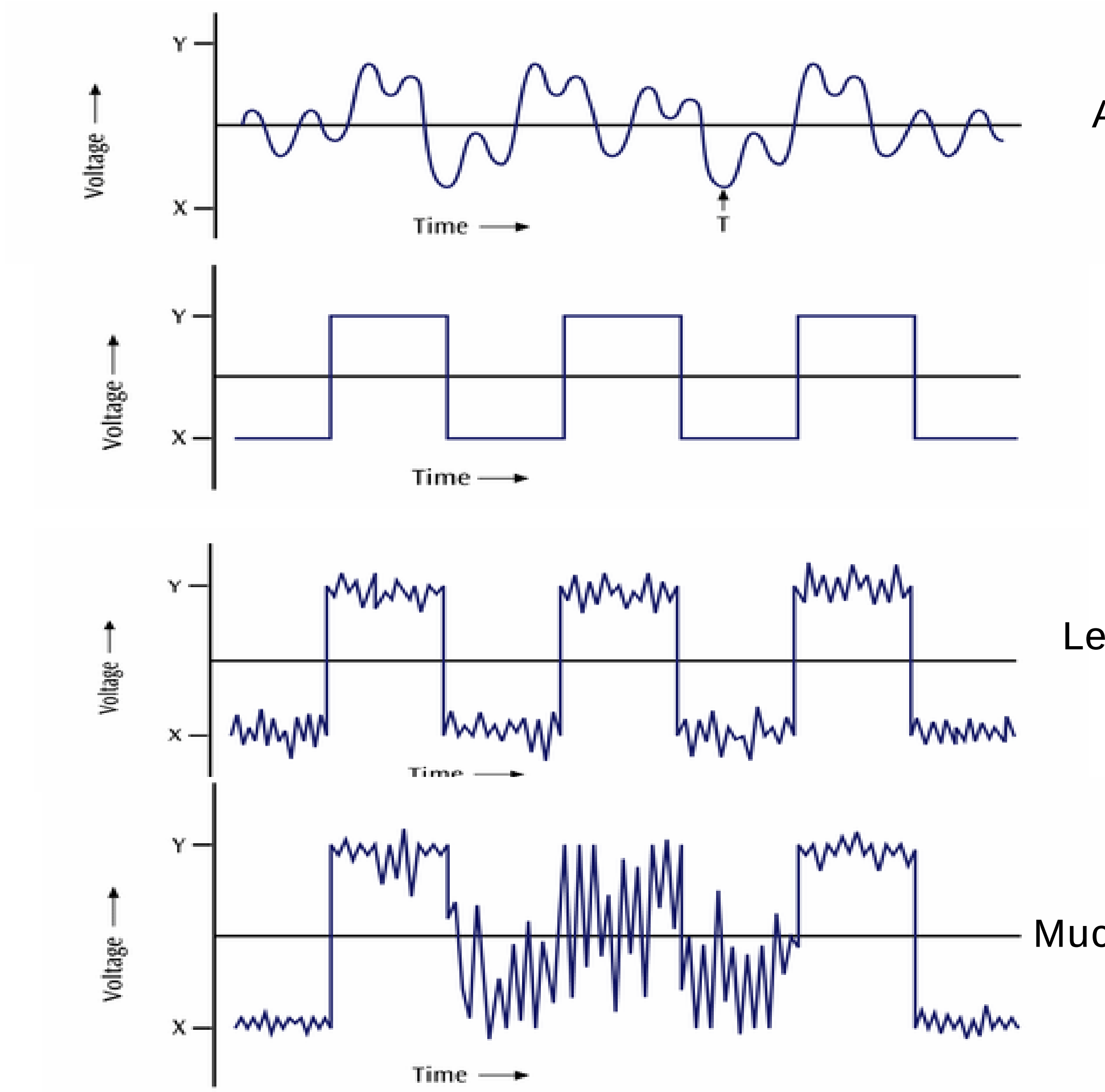


- Binary Signals
  - Digital signals with just two discrete values



Digital Baseband signals transmission



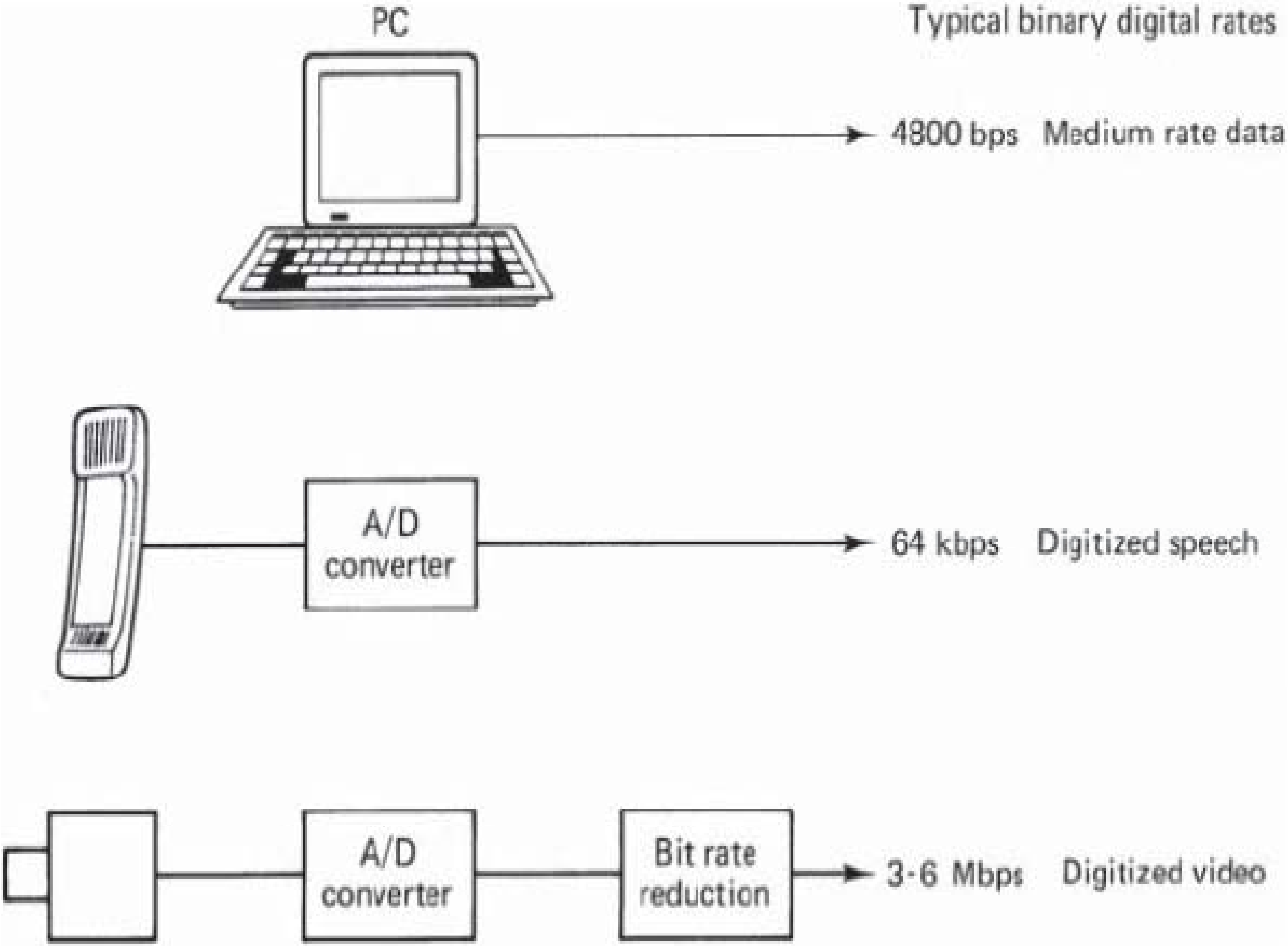


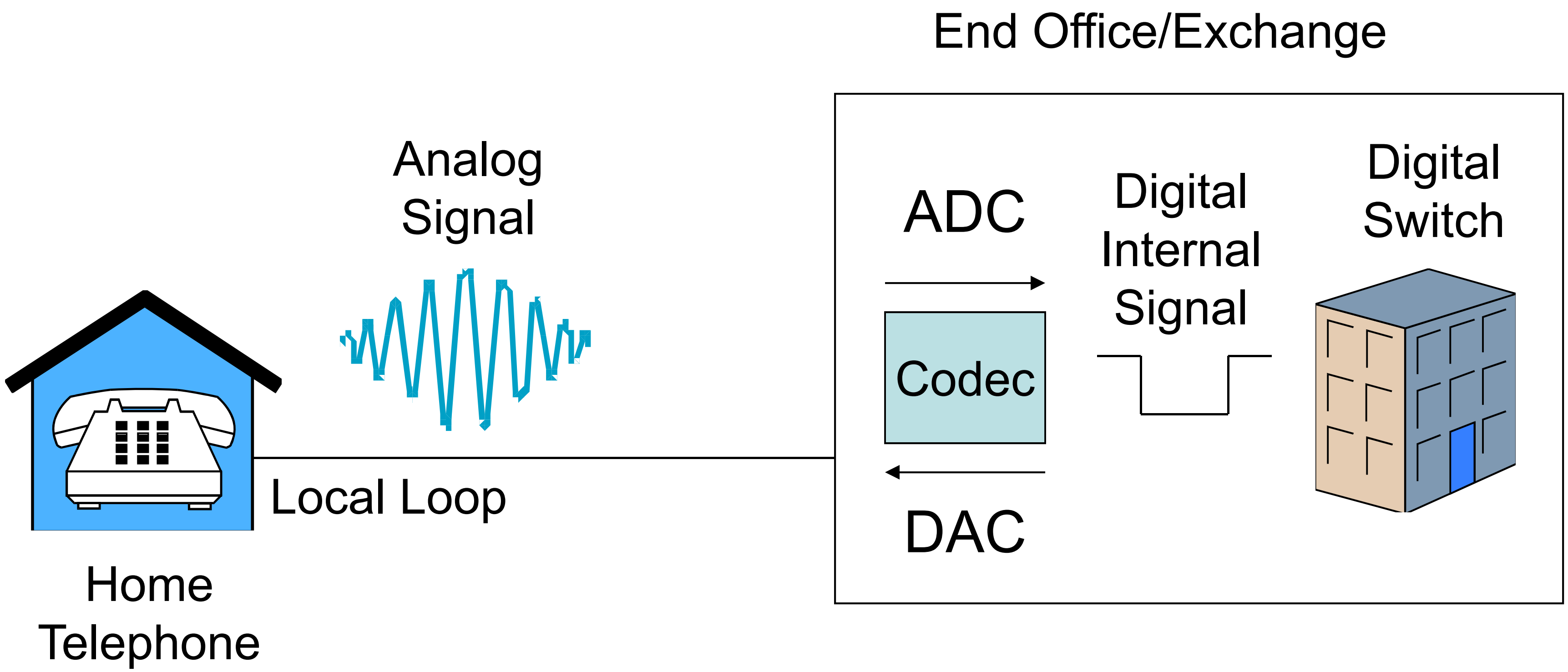
Analog signal

Digital signal

Less Noisy Digital signal

Much Noisy Digital signal


PC
Typical binary digital rates
4800 bps Medium rate data
A/D converter
64 kbps Digitized speech
A/D converter
Bit rate reduction
3·6 Mbps Digitized video


End Office/Exchange
Analog
Signal
ADC
Digital
Internal
Signal
Digital
Switch
Codec
Local Loop
DAC
Home
Telephone

Carrier Communication

- Baseband signals produced by various information sources are not always suitable for direct transmission over a given channel.

- These signals are usually further modified to facilitate transmission. This conversion process is known as modulation.

- In this process, the baseband signal is used to modify some parameter of a high-frequency carrier signal.

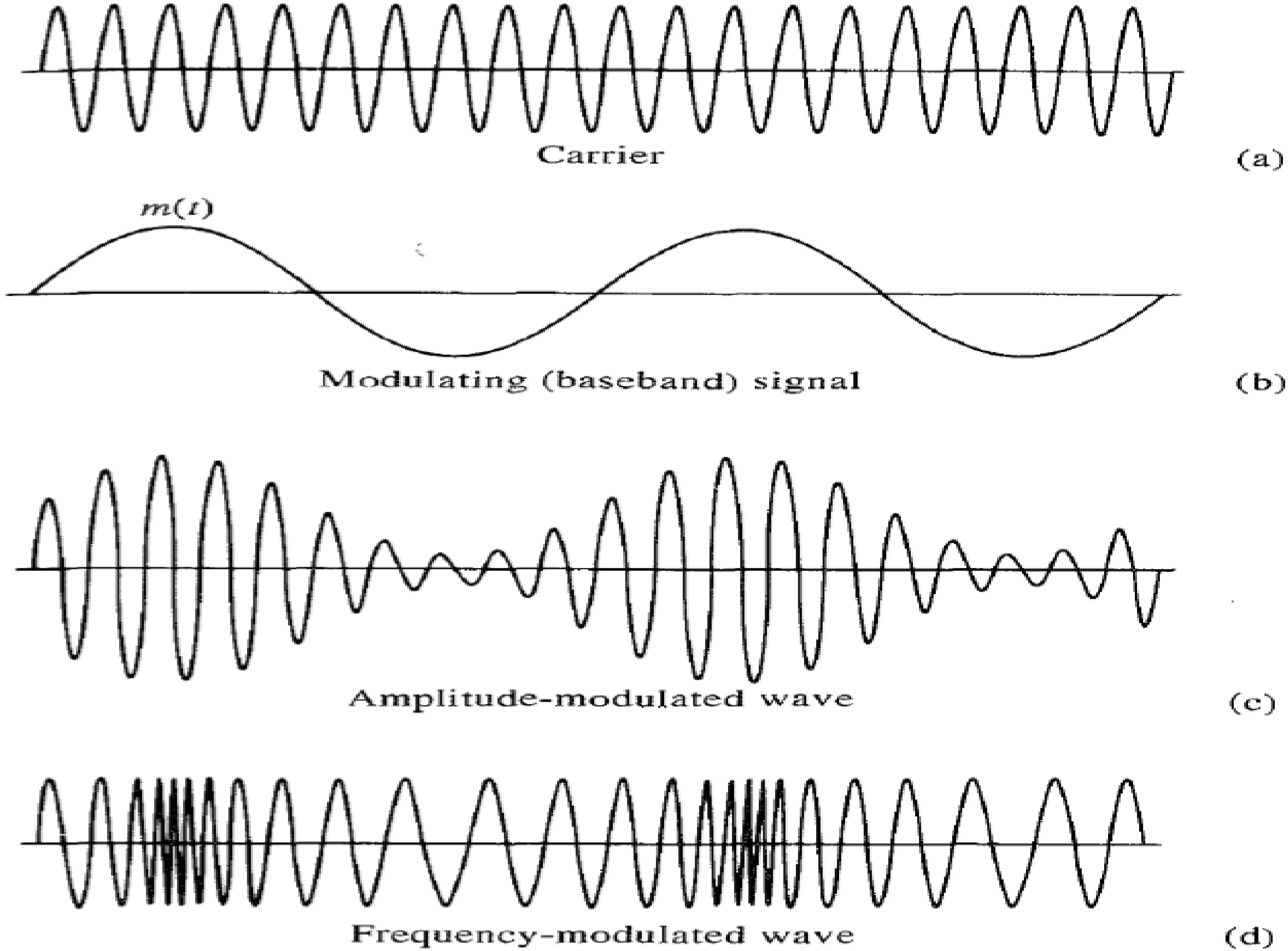
- A carrier is a sinusoid of high frequency, and one of its parameters such as amplitude, frequency, or phase is varied in proportion to the baseband signal m(t). Accordingly, we have:
  - Amplitude Modulation (AM)
  - Frequency Modulation (FM
  - Phase Modulation (PM)

- The latter two types of modulation are similar, and belong to the class of modulation known as angle modulation.

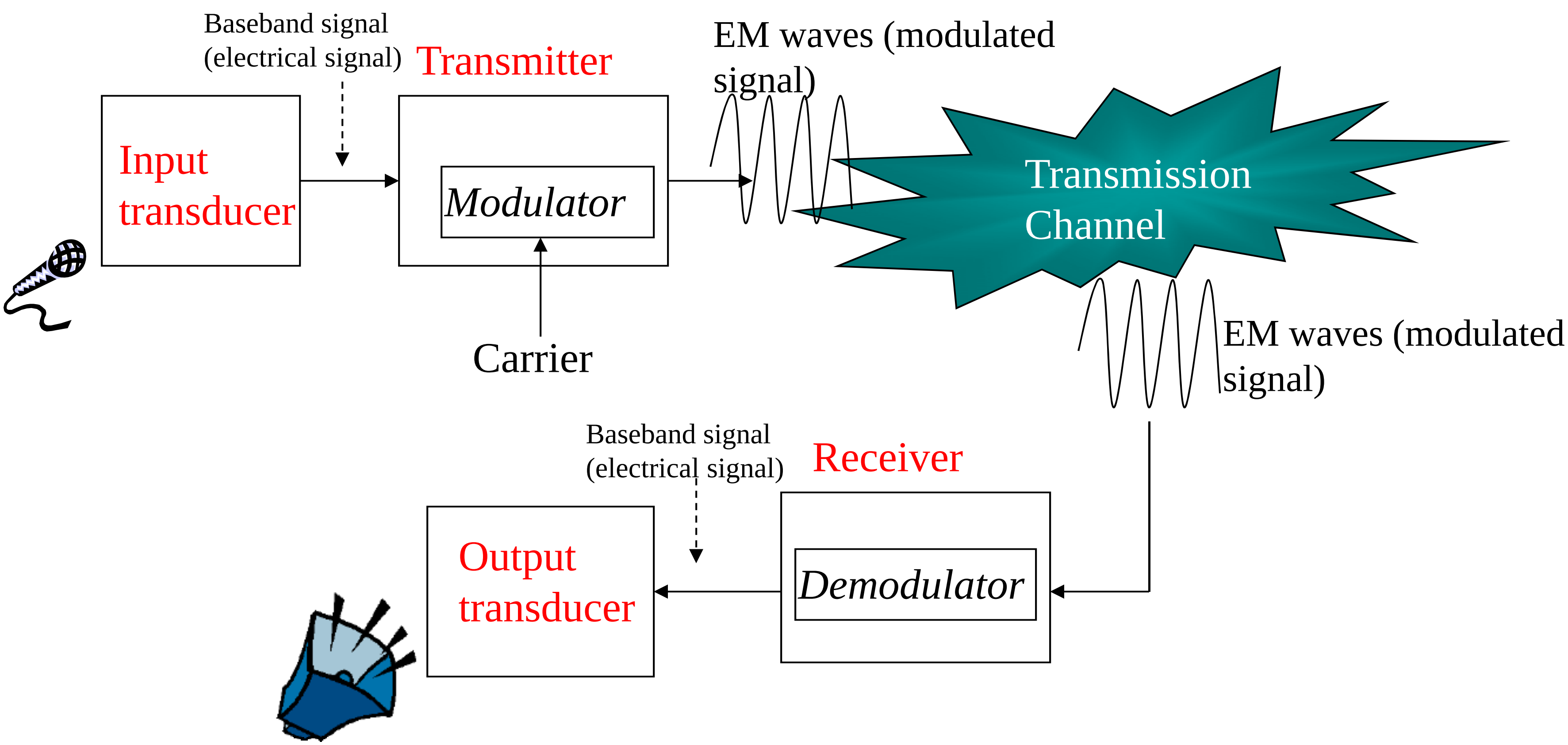
- Modulation can be helpful in utilizing the vast spectrum of frequencies available because of technological advances.

- By modulating several baseband signals and shifting their spectra to nonoverlapping bands, one can use all the available bandwidth through frequency division multiplexing (FDM).

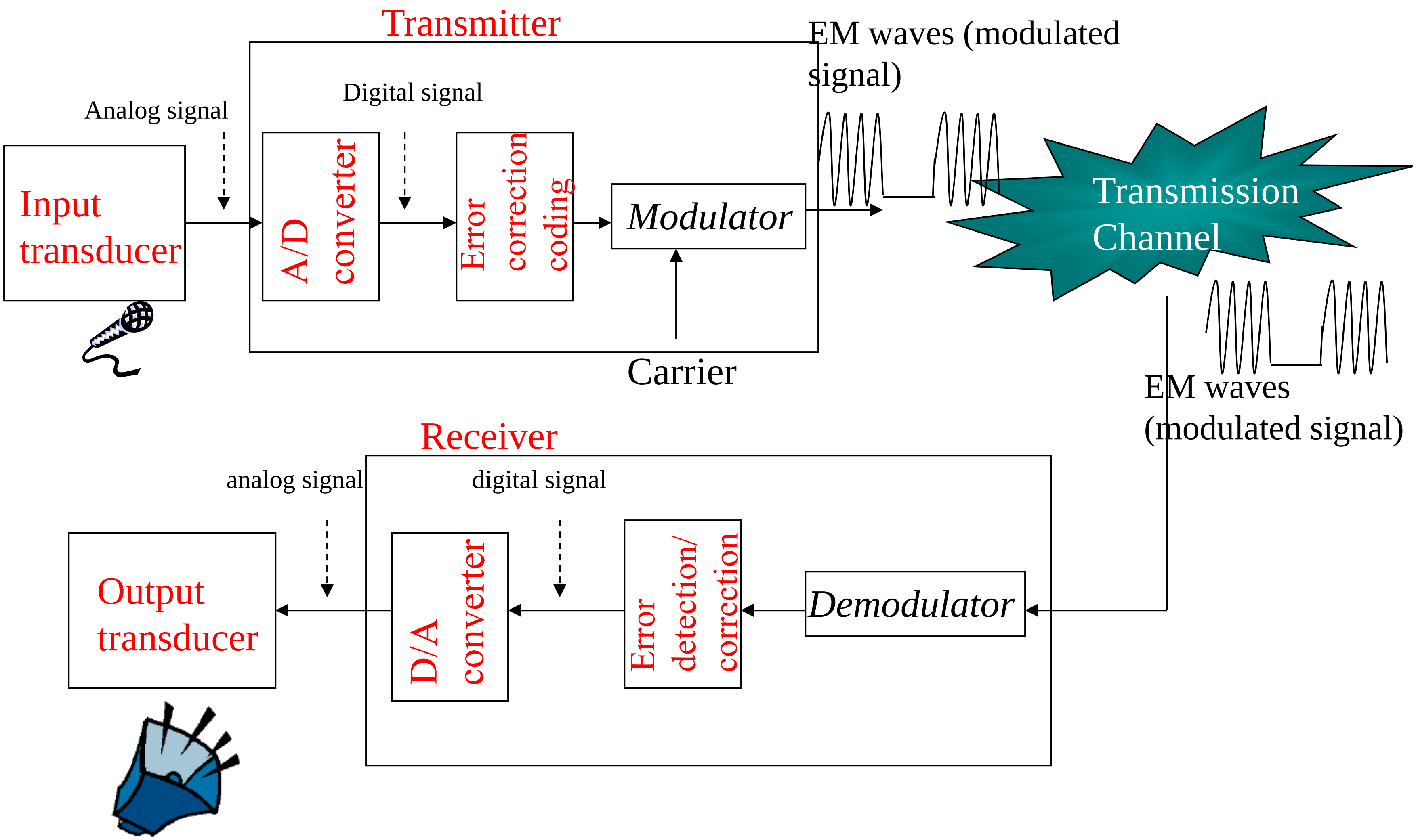
- Long- haul communication over a radio link also requires modulation to shift the signal spectrum to higher frequencies in order to enable efficient power radiation using antennas of reasonable dimensions.

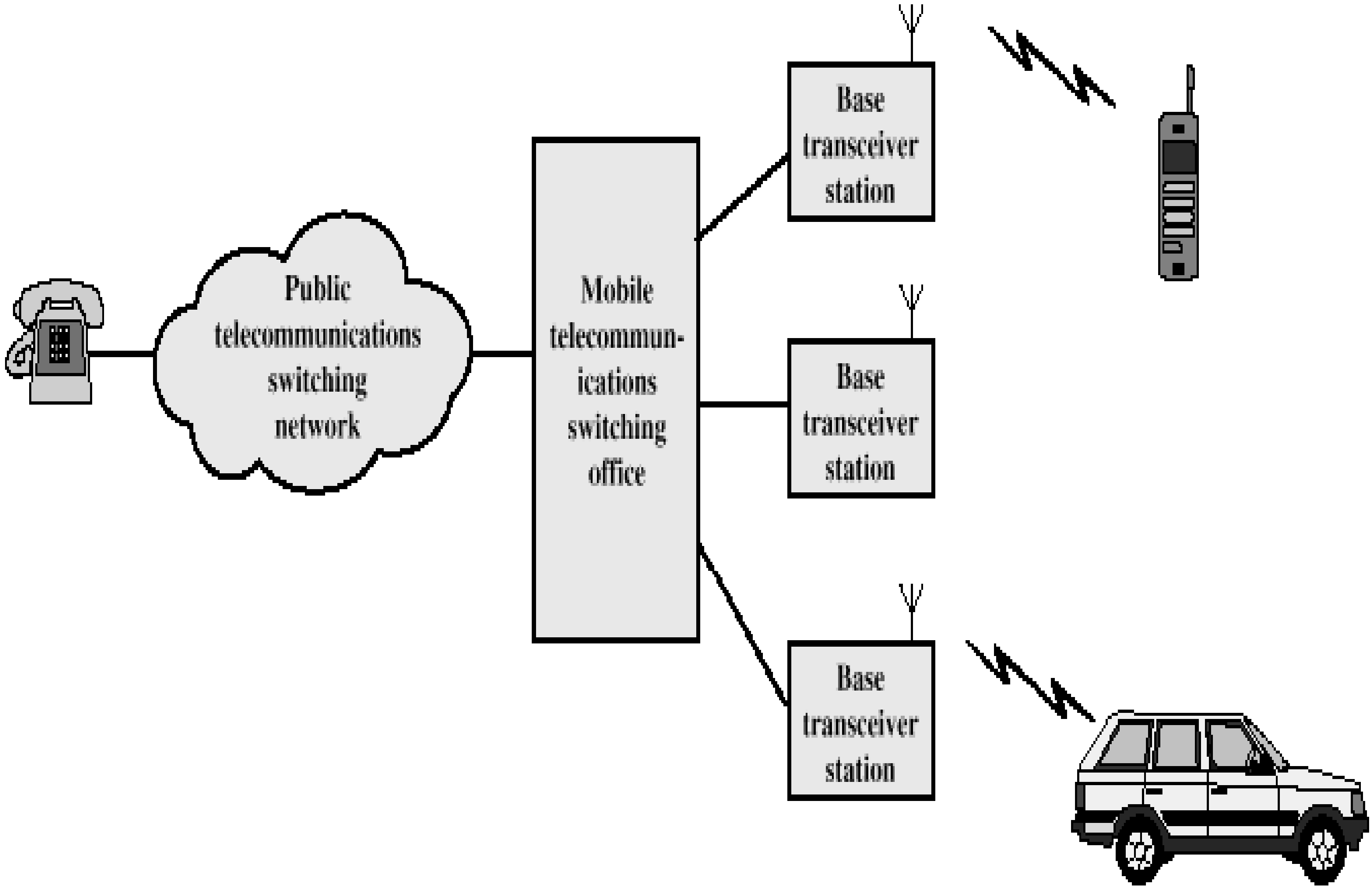

Carrier
(a)
m(t)
Modulating (baseband) signal
(b)
Amplitude-modulated wave
(c)
Frequency-modulated wave
(d)

Basic Analog communication system

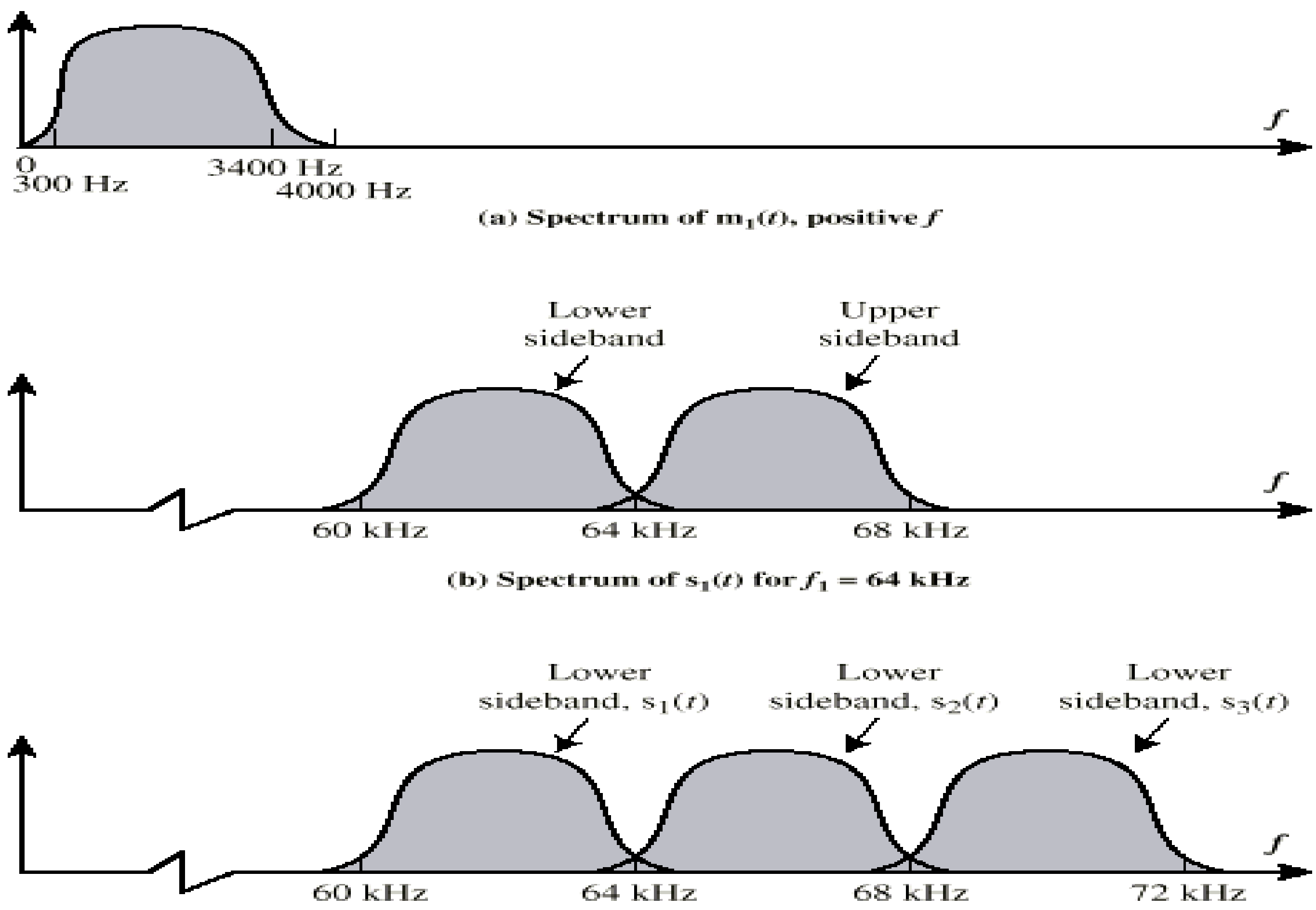


Basic digital communications system




Public
telecommunications
switching
network
Mobile
telecommun-
ications
switching
office
Base
transceiver
station
Base
transceiver
station
Base
transceiver
station

FDM of Three Voice band Signals



(c) Spectrum of composite signal using subcarriers at 64 kHz, 68 kHz, and 72 kHz

Special type of Baseband signals

- Pulse-modulated signals like:
  - Pulse Amplitude Modulation (PAM)
  - Pulse Width Modulation (PWM)
  - Pulse Position Modulation (PPM)
  - Pulse Code Modulation (PCM)
  - Delta Modulation (DM).

- Despite the term modulation, these signals are baseband signals.

- The term modulation is used here in another sense.

- Pulse-modulation schemes are really baseband coding schemes, and they yield baseband signals.

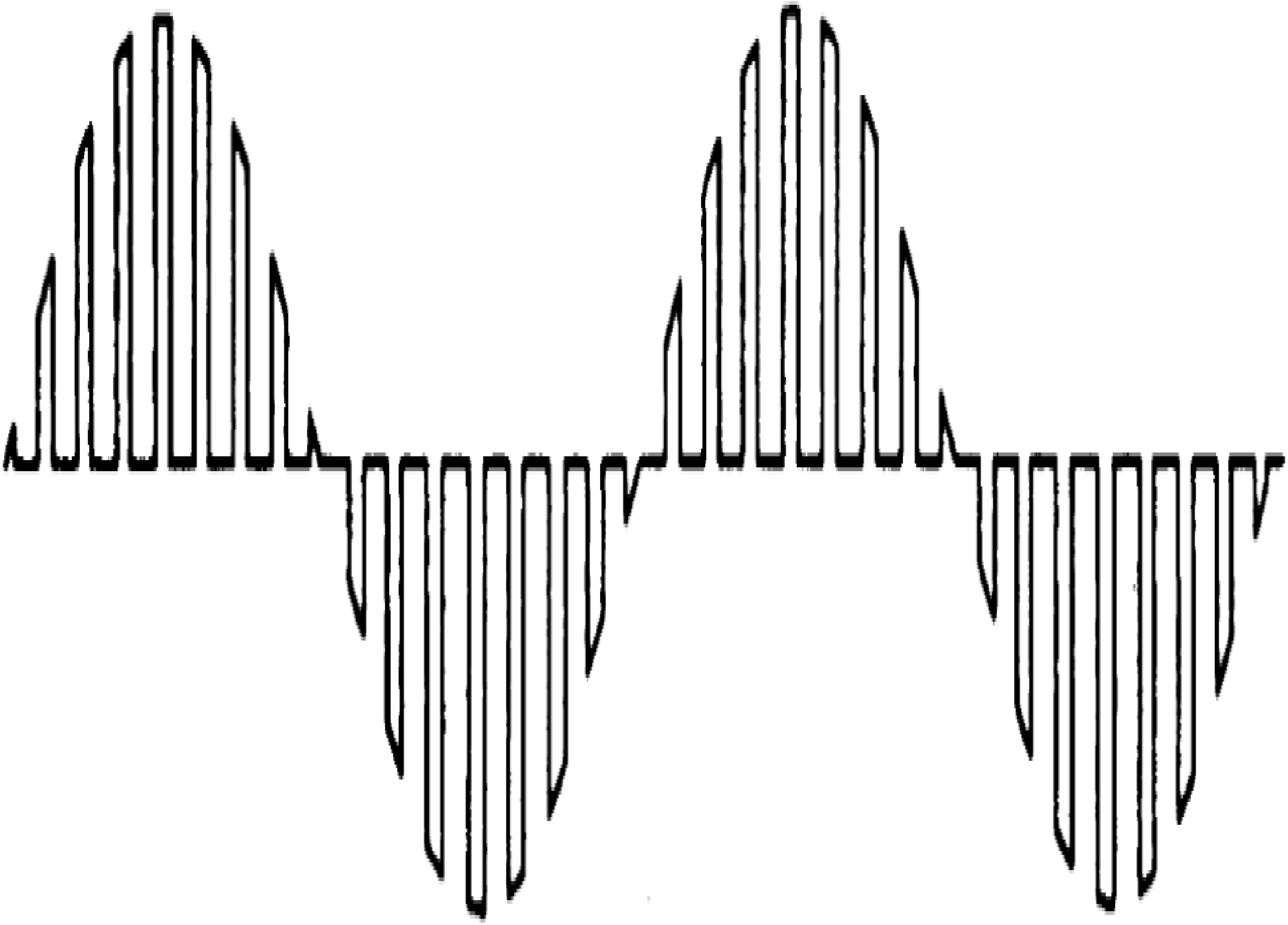
- These signals must still modulate a carrier in order to shift their spectra.

- The scheme of transmitting data by digitizing and then using pulse codes to transmit the digitized data is known as pulse-code modulation (PCM).

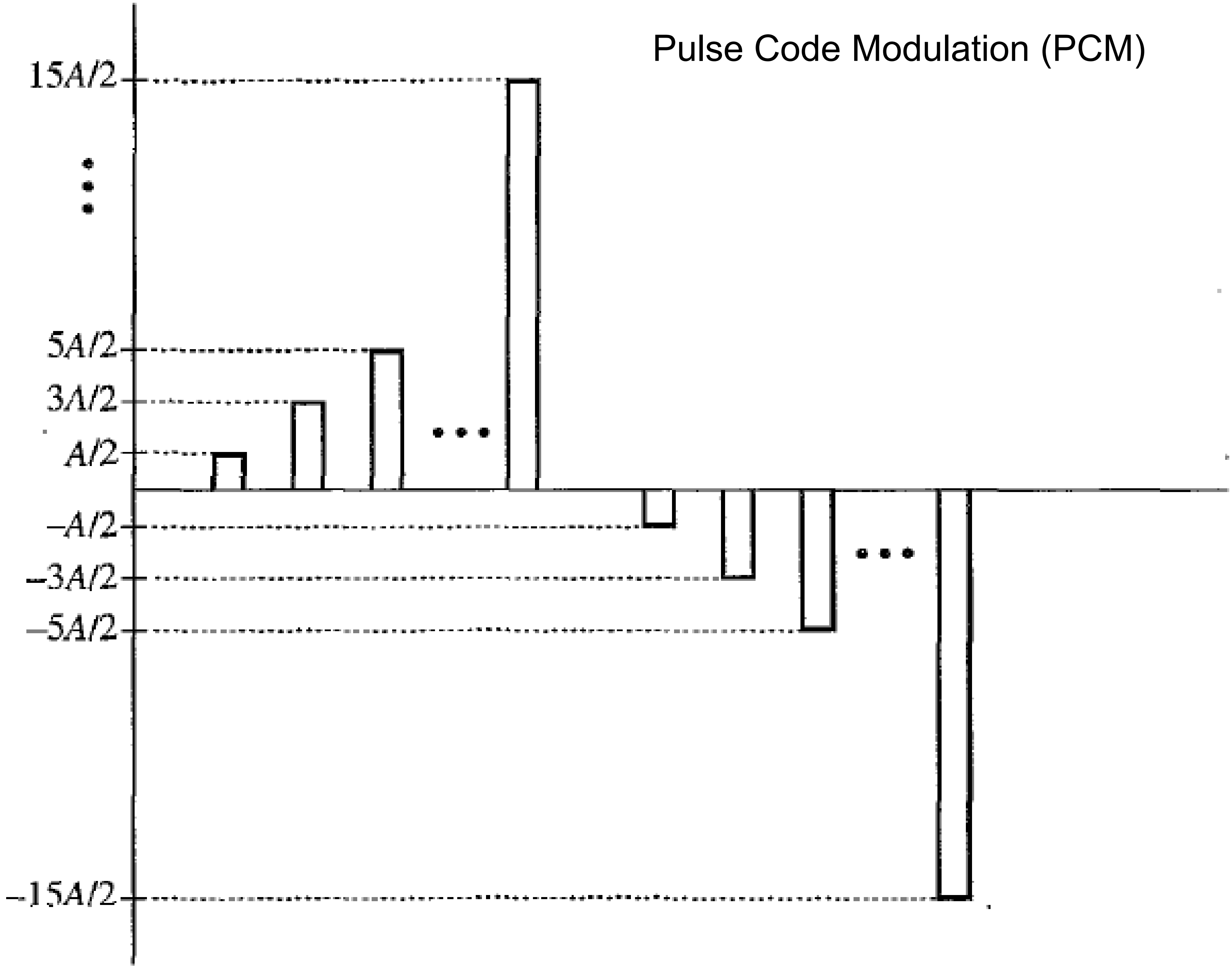
- The binary case is of great practical importance because of its simplicity and ease of detection.

- Virtually all digital communication today is binary.

Pulse Amplitude Modulation (PAM) Wave



Pulse Code Modulation (PCM)



Amplitude Modulation: DSB-SC

- Amplitude modulation is characterized by the fact that the amplitude A of the carrier $A\cos(w_c t + \theta_c)$ is varied in proportion to the baseband (message) signal rn (t), the modulating signal. The frequency w_c and the phase θ_c are constant.

- If the carrier amplitude A is made directly proportional to the modulating signal m(t), the modulated signal is $m(t)\cos w_{c:}$

- The process of modulation shifts the spectrum of the modulating signal to the left and the right by $w_{c.}$

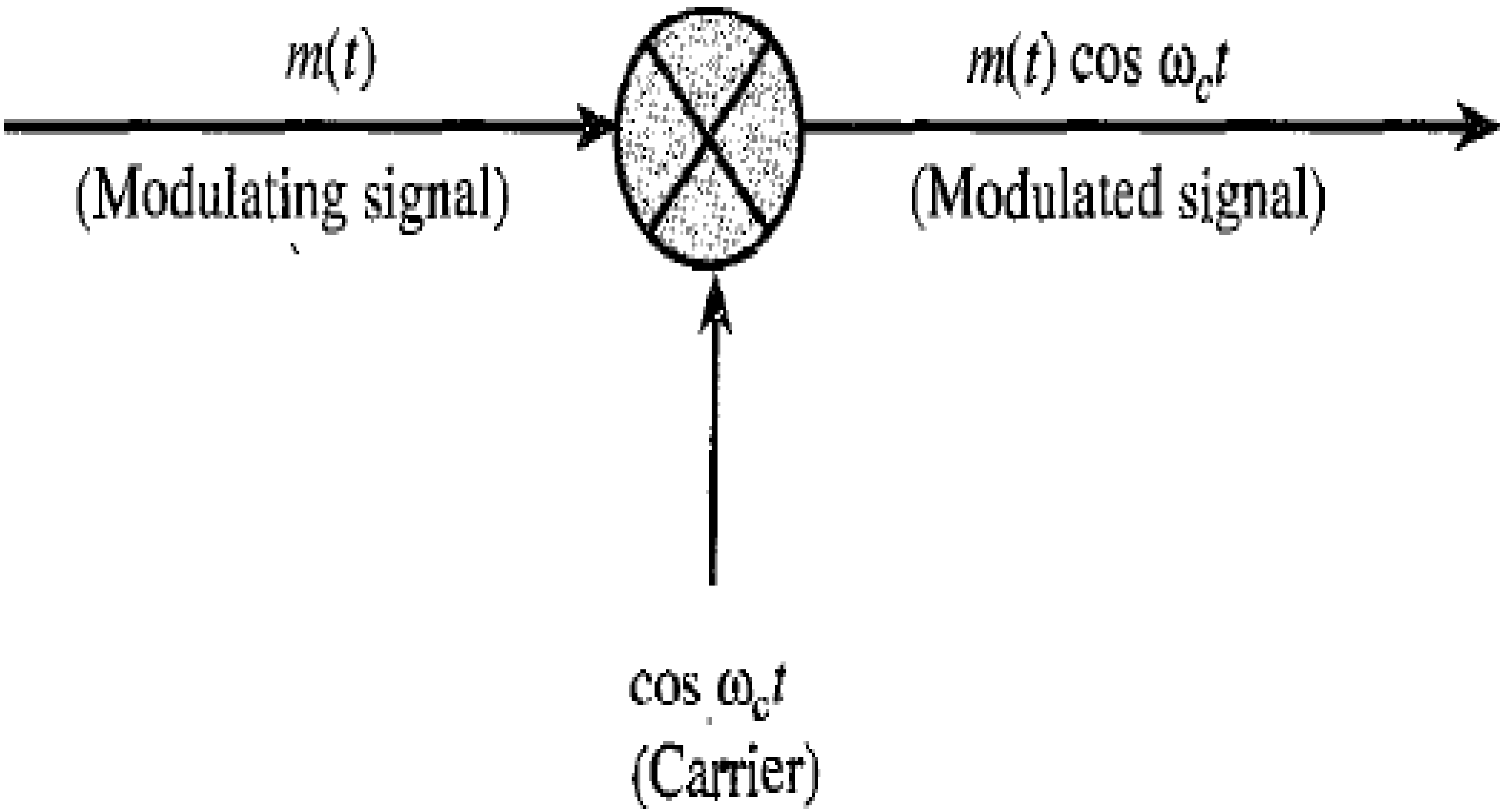
$$m(t)\cos\omega_c t \Longleftrightarrow \frac{1}{2}[M(\omega+\omega_c)+M(\omega-\omega_c)]$$

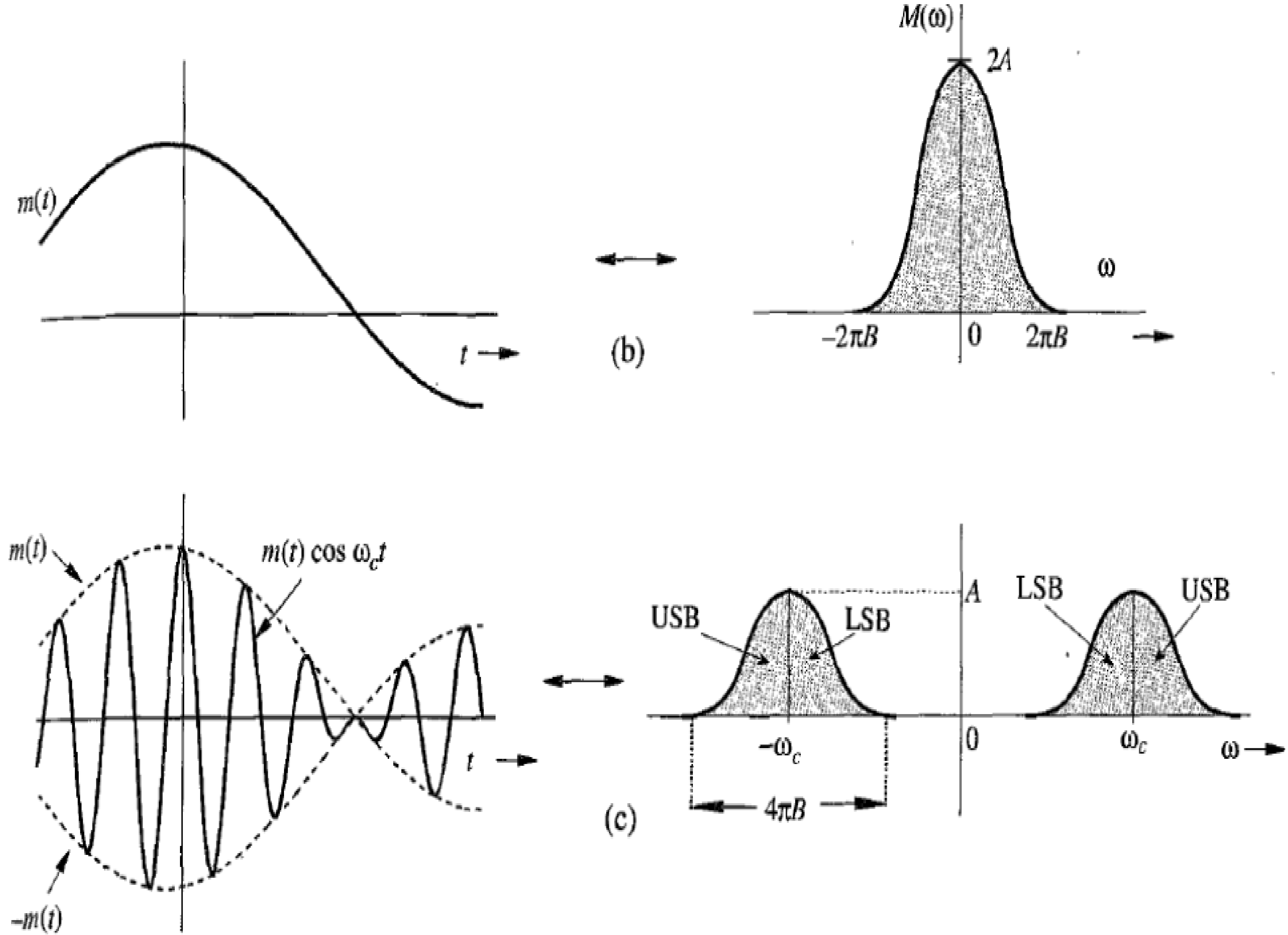
- If the bandwidth of m(t) is B Hz, then the bandwidth of the modulated signal is 2B Hz.

- The modulated signal spectrum centered at w_c is composed of two parts:
  - a portion that lies above w_c, known as the upper sideband (USB), and a portion that lies below w_c, known as the lower sideband (LSB).
  - Similarly, the spectrum centered at $-w_c$ has upper and lower sidebands.
  - Hence, this is a modulation scheme with double sidebands.

- The modulated signal in this scheme does not contain a discrete component of the carrier frequency w_c. For this reason it is called double-sideband suppressed carrier (DSB-SC) modulation.

Double-Sideband Suppressed Carrier (DSB-SC) Modulator




M(ω)
2A
m(t)
ω
t
(b)
−2πB
0
2πB
m(t)
m(t) cos ω_c t
A
LSB
USB
USB
LSB
t
−ω_c
0
ω_c
ω
(c)
4πB
−m(t)

Choice of B and w_c

- The relationship of B to w_c is of great interest. $wc \geq 2\pi B$ in order to avoid the overlap of the spectra centered at w_c and w_c.

- If $w_c < 2\pi B$, these spectra overlap and the information of m(t) is lost in the process of modulation, which makes it impossible to get back m(t) from the modulated signal m(t) cos w_ct.

- Practical factors may impose additional restrictions on w_c.

- For instance, in the case of broadcast applications, a radiating antenna can radiate only a narrow band without distortion.

- This means that to avoid distortion caused by the radiating antenna, $wc \gg 2\pi B$.

- The broadcast band AM radio, for instance, with B =5 kHz and the band of 550 to 1600 kHz for the carrier frequency give a ratio of $wc/2\pi B$ roughly in the range of 100 to 300.

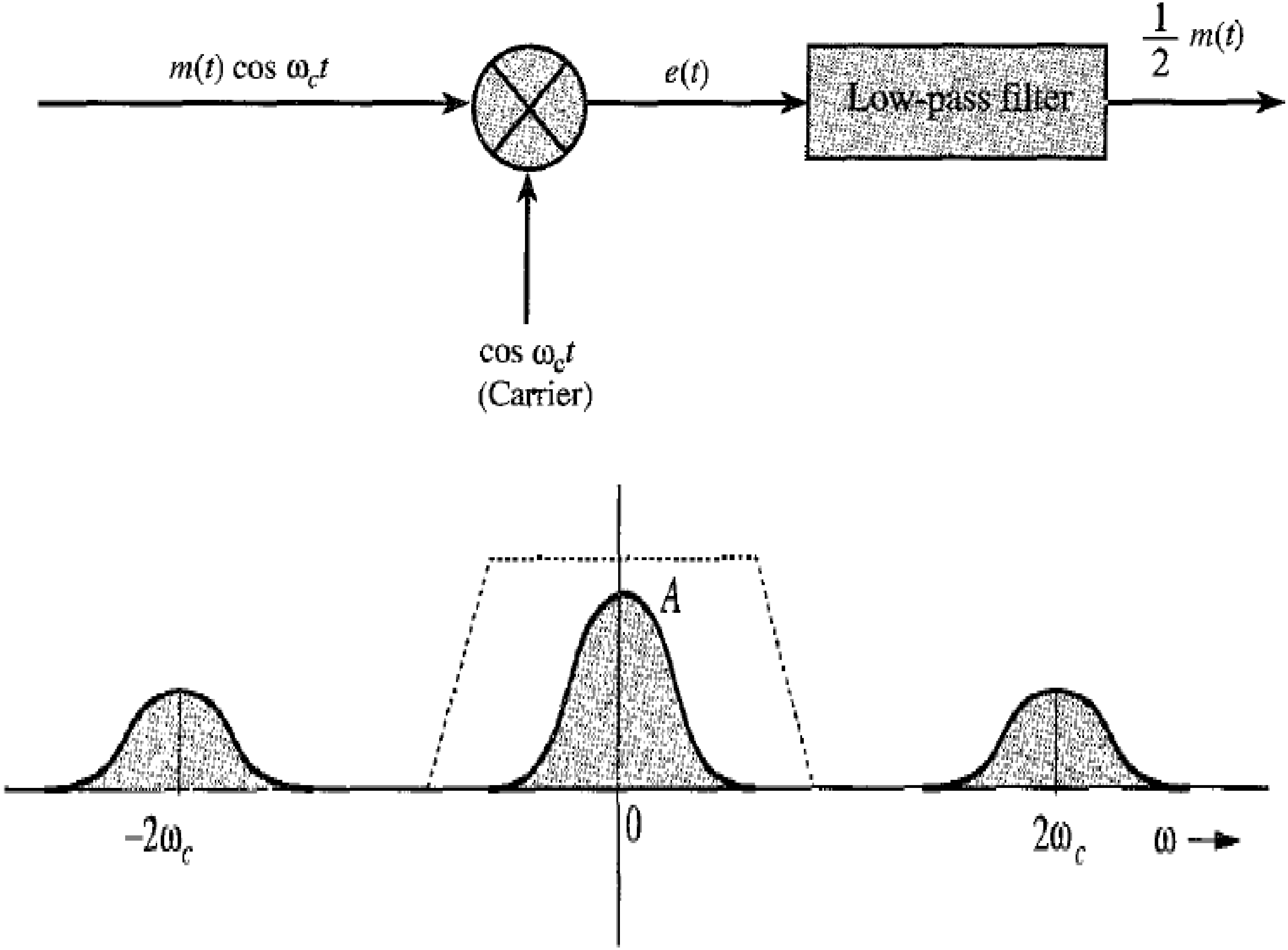
DSC-SC Demodulation

- To recover the original signal m(t) from the modulated signal, it is necessary to retranslate the spectrum to its original position.
- The process of recovering the signal from the modulated signal (retranslating the spectrum to its original position) is referred to as demodulation, or detection.
- The demodulation, which is almost identical to modulation, consists of multiplication of the incoming modulated signal m(t)cosw_ct by a carrier cosw_ct followed by a low pass filter.

$$\begin{aligned} e(t) &= m(t)\cos^2\omega_c t \\ &= \frac{1}{2}[m(t) + m(t)\cos 2\omega_c t] \end{aligned}$$

$$E(\omega) = \frac{1}{2}M(\omega) + \frac{1}{4}[M(\omega + 2\omega_c) + M(\omega - 2\omega_c)]$$

- This method of recovering the baseband signal is called synchronous detection, or coherent detection, where we use a carrier of exactly the same frequency (and phase) as the carrier used for modulation.
- Thus, for demodulation, we need to generate a local carrier at the receiver in frequency and phase coherence (synchronism) with the carrier used at the modulator.



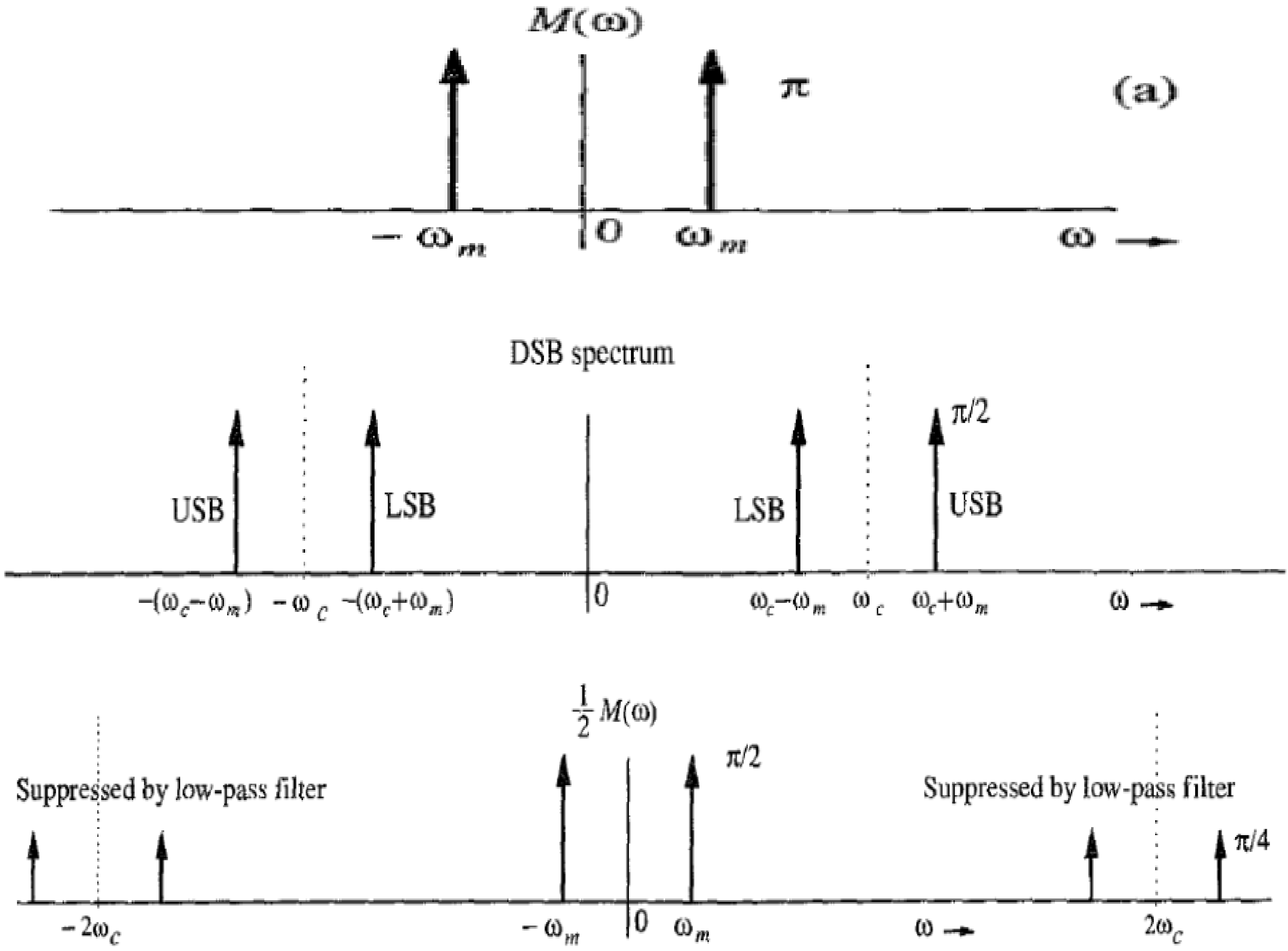
$m(t) \cos \omega_c t$
$e(t)$
Low-pass filter
$\frac{1}{2} m(t)$
$\cos \omega_c t$
(Carrier)
A
$-2\omega_c$
0
$2\omega_c$
$\omega \rightarrow$

EXAMPLE 4.1 For a baseband signal $m(t) = \cos \omega_m t$, find the DSB-SC signal, and sketch its spectrum. Identify the USB and LSB. Verify that the DSB-SC modulated signal can be demodulated by the demodulator

$$M(\omega) = \pi[\delta(\omega - \omega_m) + \delta(\omega + \omega_m)]$$

$$\begin{aligned}\varphi_{\text{DSB-SC}}(t) &= m(t)\cos \omega_c t \\ &= \cos \omega_m t \cos \omega_c t \\ &= \frac{1}{2}[\cos(\omega_c + \omega_m)t + \cos(\omega_c - \omega_m)t]\end{aligned}$$

$$\begin{aligned}e(t) &= \cos \omega_m t \cos^2 \omega_c t \\ &= \frac{1}{2}\cos \omega_m t\,(1 + \cos 2\omega_c t)\end{aligned}$$



M(ω)
π
(a)
− ω_m
O
ω_m
ω →
DSB spectrum
π/2
USB
LSB
LSB
USB
−(ω_c−ω_m)
− ω_c
−(ω_c+ω_m)
0
ω_c−ω_m
ω_c
ω_c+ω_m
ω →
1/2 M(ω)
π/2
Suppressed by low-pass filter
Suppressed by low-pass filter
π/4
− 2ω_c
− ω_m
0
ω_m
ω →
2ω_c

Modulators

- Modulation can be achieved in several ways. Some important categories of modulators are:
  - **Multiplier Modulators**
  - **Nonlinear Modulators (single balanced modulators)**
  - **Switching Modulators**
    - Diode-bridge modulator
    - Series-bridge diode modulator
    - Shunt-bridge diode modulator
    - Ring modulator(a double balanced modulator)

Multiplier Modulators

- The modulation is achieved directly by multiplying m(t) by $\cos w_c t$ using an analog multiplier whose output is proportional to the product of two input signals.

- It is rather difficult to maintain linearity in this kind of amplifier, and they tend to be rather expensive.

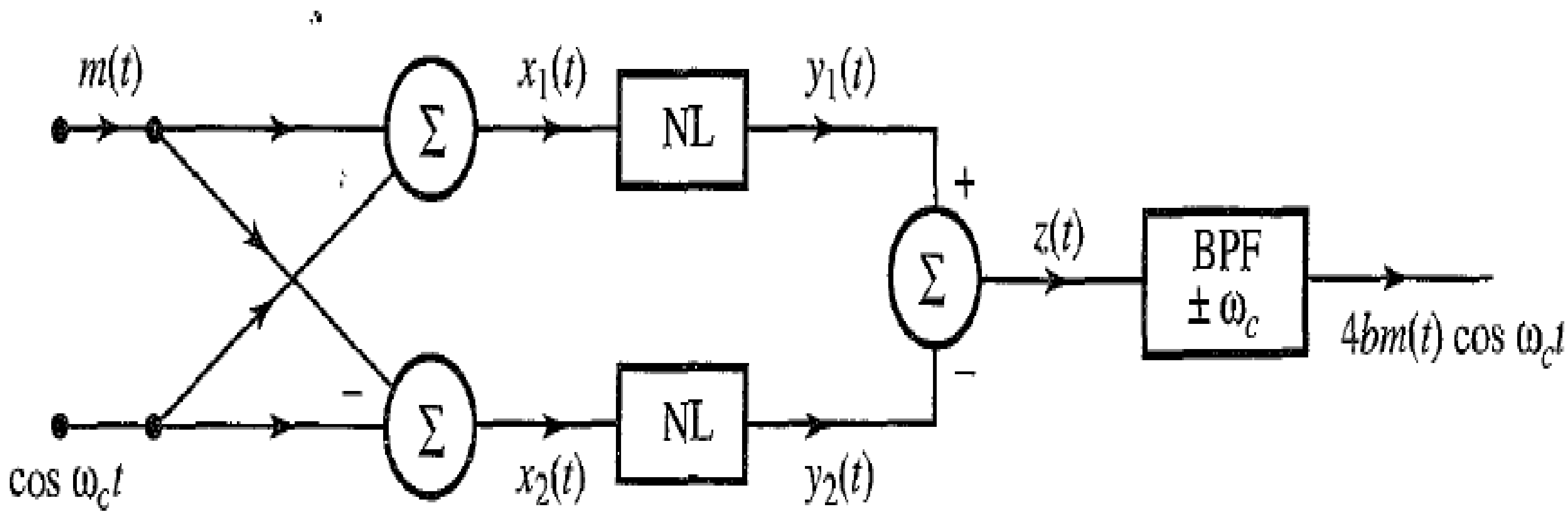
- Such a multiplier may be obtained from a variable-gain amplifier in which the gain parameter is controlled by one of the signals, say m(t).

- When the signal $\cos w_c t$ is applied at the input of this amplifier, the output is proportional to $m(t) \cos w_c t$.

- Another way to multiply two signals is through logarithmic amplifiers.

Nonlinear Modulators (single balanced modulators)

- Modulation can also be achieved by using nonlinear devices, such as a semiconductor diode or a transistor.



$$y(t) = ax(t) + bx^2(t) \qquad z(t) = y_1(t) - y_2(t) = [ax_1(t) + bx_1{}^2(t)] - [ax_2(t) + bx_2{}^2(t)]$$

$$x_1(t) = \cos \omega_c t + m(t) \text{ and } x_2(t) = \cos \omega_c t - m(t)$$

$$z(t) = 2am(t) + 4bm(t) \cos \omega_c t$$

Switching Modulators

- The multiplication operation required for modulation can be replaced by a simpler switching operation if we realize that a modulated signal can be obtained by multiplying m(t) not only by a pure sinusoid but by any periodic signal Φ(t) of the fundamental radian frequency w_c.

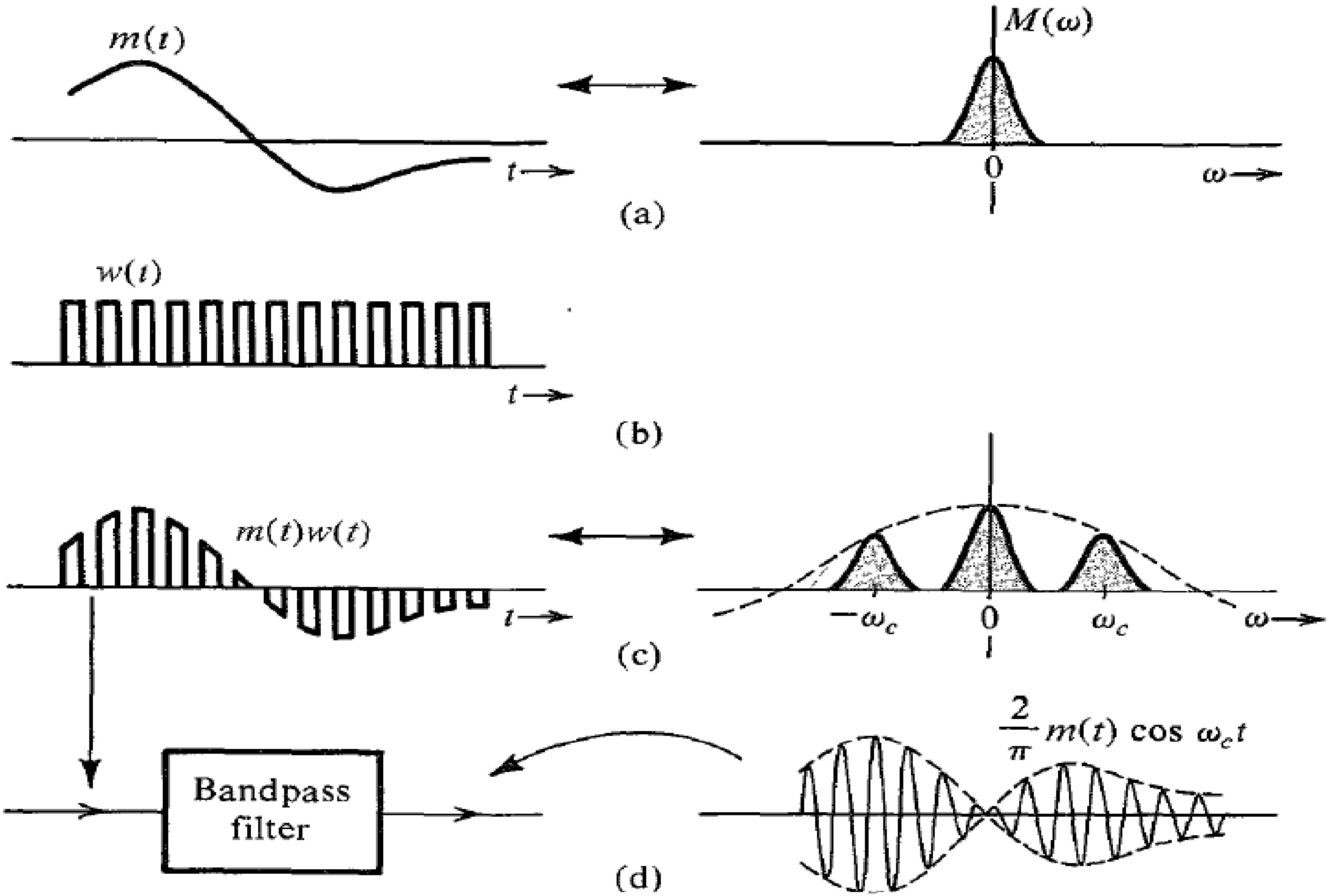
$$\phi(t) = \sum_{n=0}^{\infty} C_n \cos(n\omega_c t + \theta_n)$$

$$m(t)\phi(t) = \sum_{n=0}^{\infty} C_n \, m(t) \cos(n\omega_c t + \theta_n)$$

- The spectrum of the product m(t) Φ(t) is the spectrum M(w) shifted to

$$\pm\omega_c,\ \pm2\omega_c,\ \ldots,\ \pm n\omega_c.$$

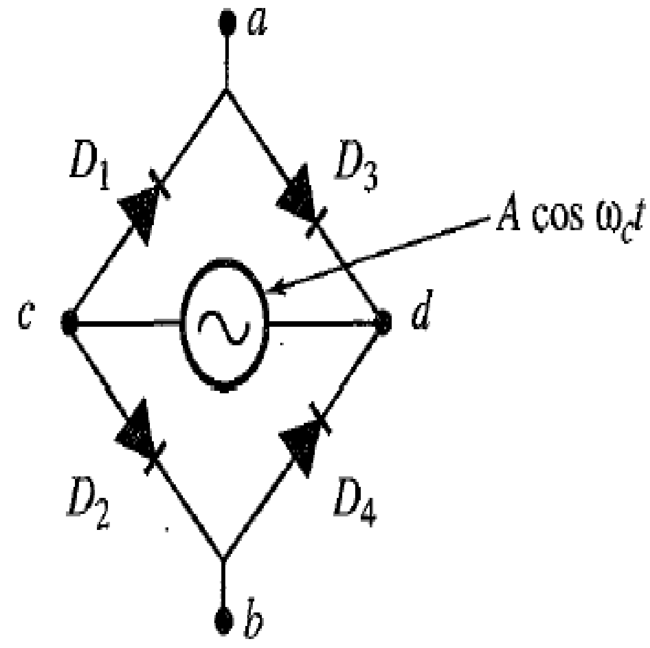
- If this signal is passed through a bandpass filter of bandwidth 2B Hz and tuned to w_o then we get the desired modulated signal.



$$w(t) = \frac{1}{2} + \frac{2}{\pi}\left(\cos\omega_c t - \frac{1}{3}\cos 3\omega_c t + \frac{1}{5}\cos 5\omega_c t - \cdots\right) \qquad m(t)w(t) = \frac{1}{2}m(t) + \frac{2}{\pi}\left[m(t)\cos\omega_c t - \frac{1}{3}m(t)\cos 3\omega_c t + \frac{1}{5}m(t)\cos 5\omega_c t - \cdots\right]$$

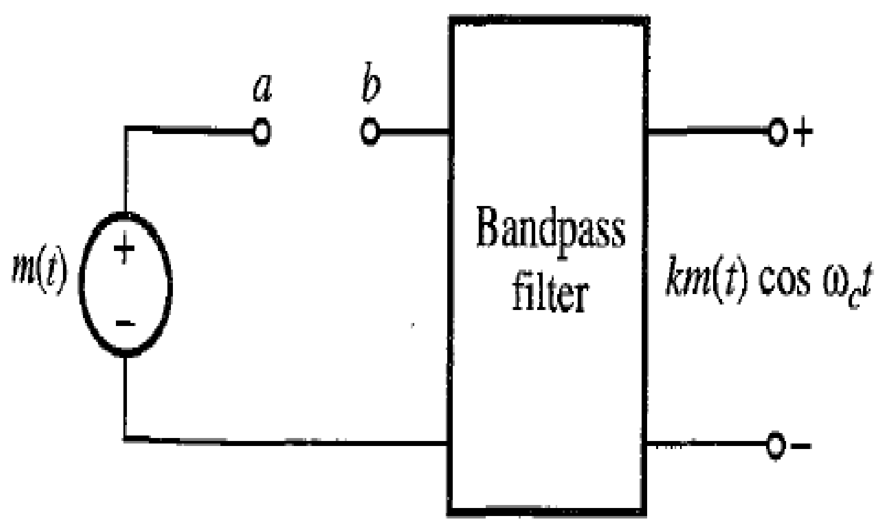
Diode-bridge modulator

- Multiplication of a signal by a square pulse train is in reality a switching operation.
- It involves switching the signal m(t) on and off periodically and can be accomplished by simple switching elements controlled by w(t).
- The diode-bridge modulator is one such electronic switch driven by a sinusoid A $\cos w_c t$ to produce the switching action.
- To obtain the signal m(t)w(t), we may place this electronic switch in series or across (in parallel) m(t).
- These modulators are known as the series-bridge diode modulator and the shunt-bridge diode modulator, respectively.
- This switching on and off of m (t) repeats for each cycle of the carrier, resulting in the switched signal m(t)w(t), which when band pass filtered, yields the desired modulated signal.



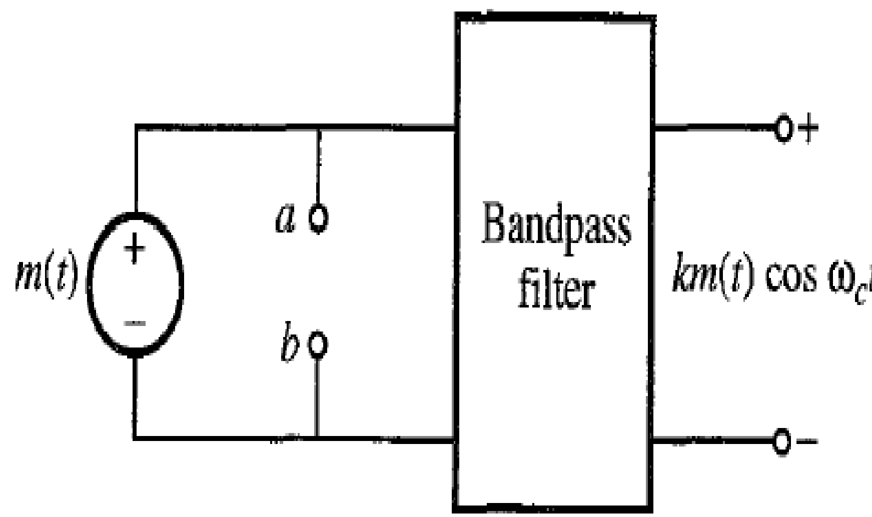
a
D_1
D_3
$A \cos \omega_c t$
c
d
D_2
D_4
b

(a)



a
b
$m(t)$
+
–
Bandpass filter
$km(t) \cos \omega_c t$
+
–

(b)



$m(t)$
+
–
a
b
Bandpass filter
$km(t) \cos \omega_c t$
+
–

(c)

Ring modulator

- In Ring modulator the output is proportional to m(t) during the positive half-cycle and to -m(t) during the negative half-cycle.

- In effect, m(t) is multiplied by a square pulse train $w_o(t)$.

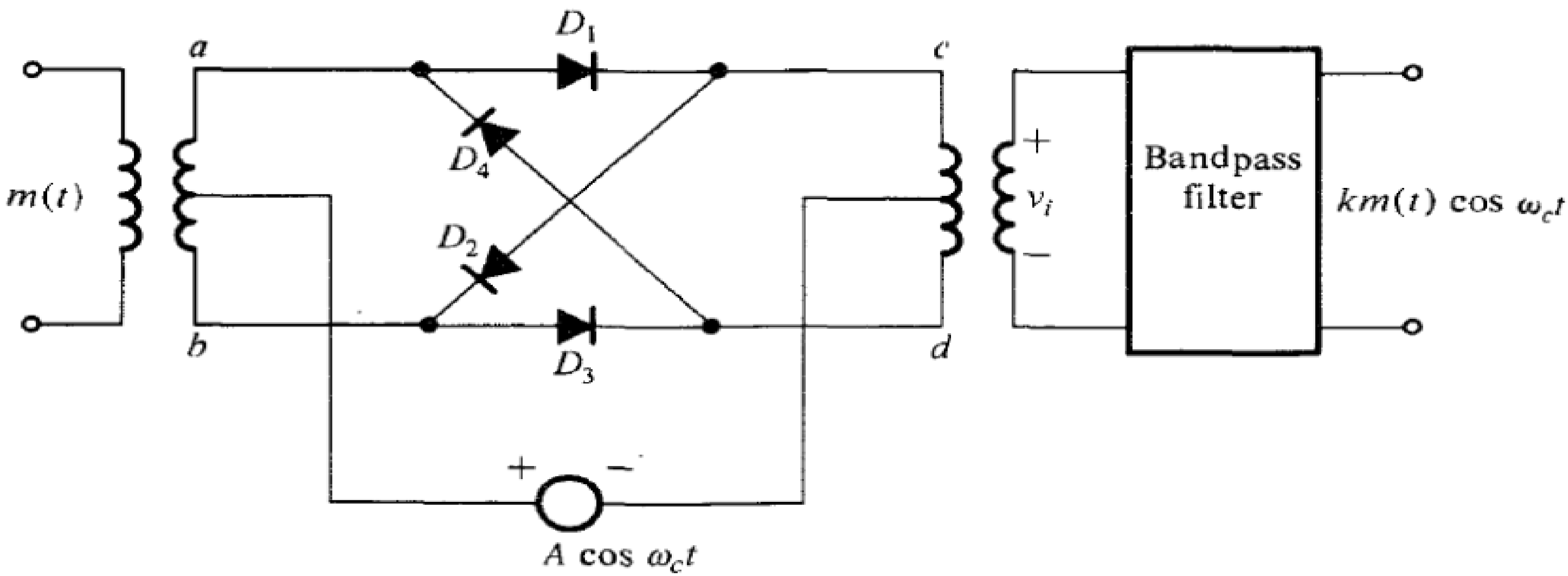
- When this waveform is passed through a bandpass filter tuned to w_c the filter output will be the modulated signal.

$$w_0(t) = \frac{4}{\pi}\left(\cos \omega_c t - \frac{1}{3}\cos 3\omega_c t + \frac{1}{5}\cos 5\omega_c t - \cdots\right)$$

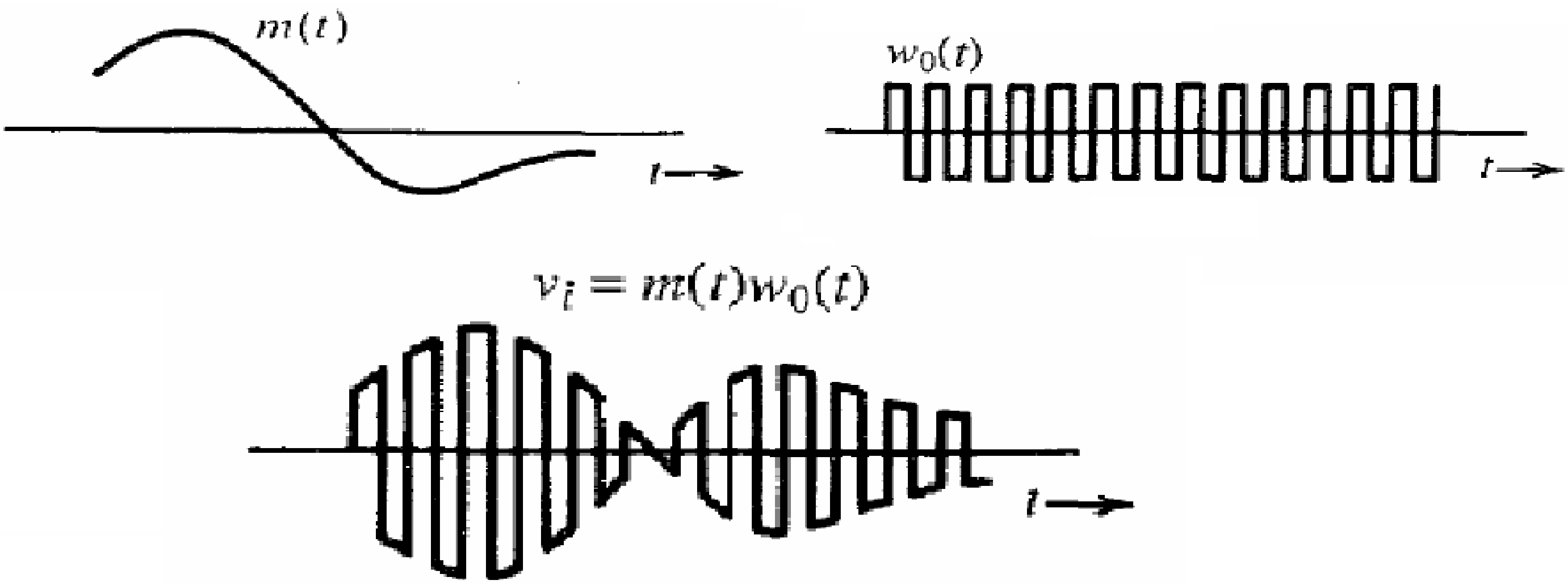
$$v_i(t) = m(t)w_0(t) = \frac{4}{\pi}\left[m(t)\cos \omega_c t - \frac{1}{3}m(t)\cos 3\omega_c t + \frac{1}{5}m(t)\cos 5\omega_c t - \cdots\right]$$

- The input to the final band pass filter does not contain either of these inputs.

- Consequently, this circuit is an example of a double balanced modulator.

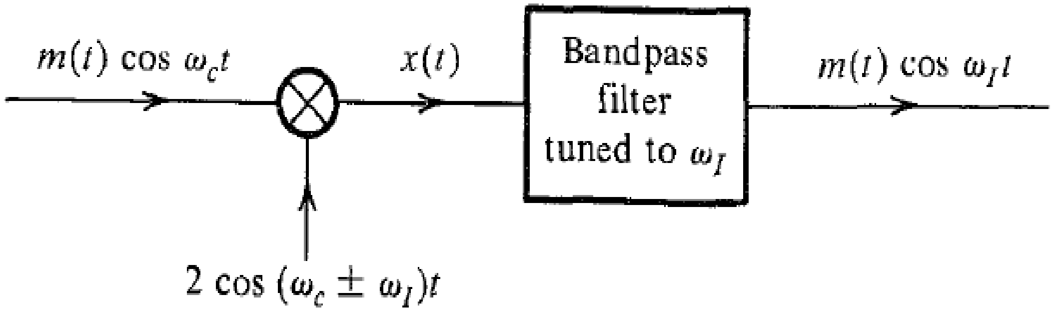


D_1
a
c
D_4
$m(t)$
Bandpass filter
$km(t) \cos \omega_c t$
$+$
v_i
$-$
D_2
b
D_3
d
$+$
$-$
$A \cos \omega_c t$

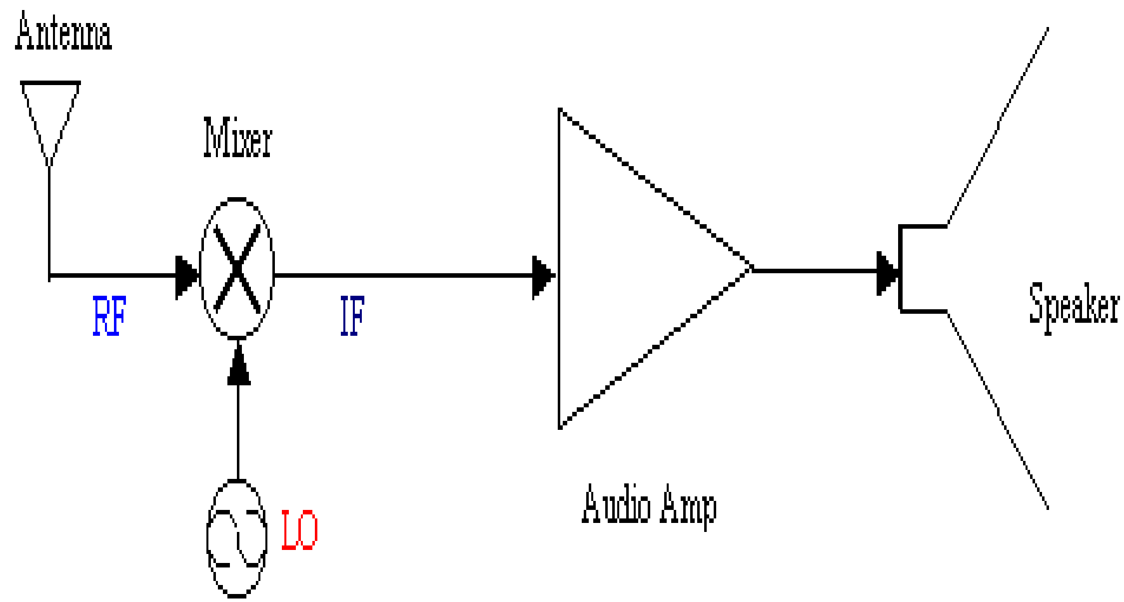


$m(t)$
t
$w_0(t)$
t
$v_i = m(t)w_0(t)$
t

EXAMPLE 4.2 Frequency Mixer or Converter



$$x(t) = 2m(t) \cos \omega_c t \cos \omega_{\text{mix}} t$$
$$= m(t)[\cos (\omega_c - \omega_{\text{mix}})t + \cos (\omega_c + \omega_{\text{mix}})t]$$



Demodulation of DSB-SC Signals

- Demodulation of a DSB-SC signal is identical to modulation.
- At the receiver, we multiply the incoming signal by a local carrier of frequency and phase in synchronism with the carrier used at the modulator.
- The product is then passed through a low- pass filter.
- The only difference between the modulator and the demodulator is the output filter.
- In the modulator, the multiplier output is passed through a band pass filter tuned to w_c, whereas in the demodulator, the multiplier output is passed through a low-pass filter.
- Therefore, all the modulators discussed earlier can also be used as demodulators, provided the bandpass filters at the output are replaced by low-pass filters of bandwidth B.
- For demodulation, the receiver must generate a carrier in phase and frequency synchro- nism with the incoming cartier.
- These demodulators are called synchronous or coherent (also homodyne) demodulators.

EXAMPLE 4.3 Analyze the switching demodulator that uses the electronic switch (diode bridge) as a switch (either in series or in parallel).

$$m(t)\cos\omega_c t \times w(t) = m(t)\cos\omega_c t\left[\frac{1}{2}+\frac{2}{\pi}\left(\cos\omega_c t - \frac{1}{3}\cos 3\omega_c t + \cdots\right)\right]$$

$$= \frac{2}{\pi}m(t)\cos^2\omega_c t + \text{terms of the form } m(t)\cos n\omega_c t$$

$$= \frac{1}{\pi}m(t) + \frac{1}{\pi}m(t)\cos 2\omega_c t + \text{terms of the form } m(t)\cos n\omega_c t$$

Amplitude Modulation (AM):DSB with carrier

- For the suppressed carrier scheme , a receiver must generate a carrier in frequency and phase synchronism with the carrier at the transmitter that may be located hundreds or thousands of miles away.

- This calls for a sophisticated receiver and could be quite costly.

- The other alternative is for the transmitter to transmit a carrier A cos $w_c t$ [along with the modulated signal m(t) cos $w_c t$] so that there is no need to generate a carrier at the receiver.

- In this case the transmitter needs to transmit much larger power, which makes it rather expensive.

- In point-to-point communications, where there is one transmitter for each receiver, substantial complexity in the receiver system can be justified, provided it results in a large enough saving in expensive high-power transmitting equipment.

- On the other hand, for a broadcast system with a multitude of receivers for each transmitter, it is more economical to have one expensive high-power transmitter and simpler, less expensive receivers.

- The second option (transmitting a carrier along with the modulated signal) is the obvious choice for this case. This is the so-called AM (amplitude modulation).

The transmitted signal is given by:

$$\begin{aligned}\varphi_{AM}(t) &= A\cos\omega_c t + m(t)\cos\omega_c t \\ &= [A + m(t)]\cos\omega_c t\end{aligned}$$

$$\varphi_{AM}(t) \iff \frac{1}{2}[M(\omega+\omega_c) + M(\omega-\omega_c)] + \pi A[\delta(\omega+\omega_c) + \delta(\omega-\omega_c)]$$

- The condition for envelope detection of an AM signal is:

$$A + m(t) \geq 0 \qquad \text{for all } t \qquad A \geq m_p$$

- The modulation index μ is defined as:

$$\mu = \frac{m_p}{A}$$

- So the required condition for the viability of demodulation of AM by an envelope detector is:

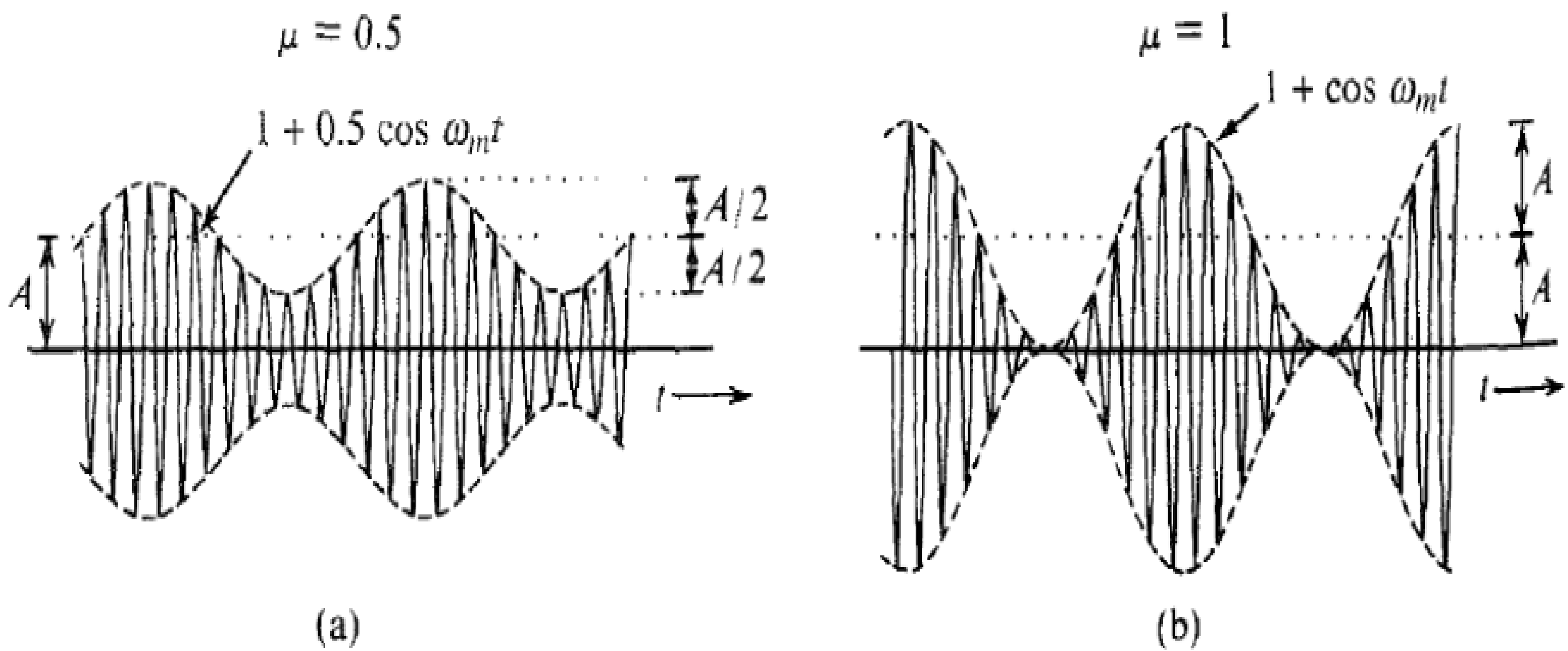
$$0 \leq \mu \leq 1$$

- When $A < m_p$ then $\mu > 1$ (overmodulation). In this case, the option of envelope detection is no longer viable. We then need to use synchronous demodulation.

- The synchronous demodulation can be used for any value of μ.

- The envelope detector, which is considerably simpler and less expensive than the synchronous detector, can be used only for $\mu \leq 1$.

EXAMPLE 4.4 Sketch $\varphi_{AM}(t)$ for modulation indices of $\mu = 0.5$ and $\mu = 1$, when $m(t) = B\cos\omega_m t$. This case is referred to as **tone modulation** because the modulating signal is a pure sinusoid (or tone).



$$\mu = \frac{B}{A} \qquad m(t) = B\cos\omega_m t = \mu A\cos\omega_m t$$

$$\varphi_{AM}(t) = [A + m(t)]\cos\omega_c t = A[1 + \mu\cos\omega_m t]\cos\omega_c t$$

Sideband and Carrier Power

- The advantage of envelope detection in AM has its price. In AM, the carrier term does not carry any information, and hence, the carrier power is wasted.

$$\varphi_{\mathrm{AM}}(t) = \underbrace{A \cos \omega_c t}_{\text{carrier}} + \underbrace{m(t) \cos \omega_c t}_{\text{sidebands}}$$

$$P_c = \frac{A^2}{2} \qquad \text{and} \qquad P_s = \frac{1}{2}\widetilde{m^2(t)}$$

- The total power is the sum of the carrier (wasted) power and the sideband (useful) power.

- Hence η the power efficiency is:

$$\eta = \frac{\text{useful power}}{\text{total power}} = \frac{P_s}{P_c + P_s} = \frac{\widetilde{m^2(t)}}{A^2 + \widetilde{m^2(t)}} 100\%$$

Special case of tone modulation

- For the special case of tone modulation:

$$m(t) = \mu A \cos \omega_m t \qquad \text{and} \qquad \widetilde{m^2(t)} = \frac{(\mu A)^2}{2}$$

$$\eta = \frac{\mu^2}{2 + \mu^2} 100\%$$

$$\eta_{\max} = 33\%$$

- Thus, for tone modulation, under best conditions (μ = 1), only one-third of the transmitted power is used for carrying message.

- For practical signals, the efficiency is even worse-----on the order of 25% or lower as compared to that of the DSB-SC case.

- The best condition implies μ = 1. Smaller values of μ degrade efficiency further.

- For this reason volume compression and peak limiting are commonly used in AM to ensure that full modulation (μ = 1) is maintained most of the time.

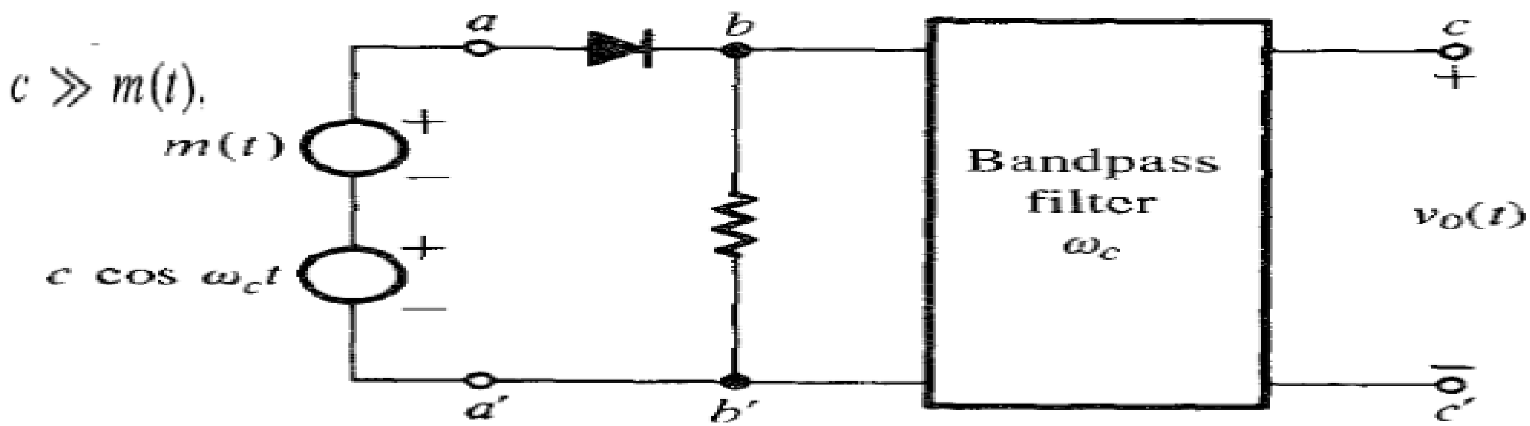
EXAMPLE 4.5 Determine η and the percentage of the total power carried by the sidebands of the AM wave for tone modulation when (**a**) $\mu = 0.5$ and (**b**) $\mu = 0.3$.

$$\eta = \frac{\mu^2}{2+\mu^2}100\% = \frac{(0.5)^2}{2+(0.5)^2}100\% = 11.11\%$$

$$\eta = \frac{(0.3)^2}{2+(0.3)^2}100\% = 4.3\%$$

Generation of AM Signals

- AM signals can be generated by any DSB-SC modulators if the modulating signal is A + m(t) instead of just m(t).
- But because there is no need to suppress the carrier in the output, the modulating circuits do not have to be balanced.
- This results in considerably simpler modulators for AM.



$$
\begin{aligned}
v_{bb'}(t) &= [c\cos\omega_c t + m(t)]w(t) \\
&= [c\cos\omega_c t + m(t)]\left[\frac{1}{2} + \frac{2}{\pi}\left(\cos\omega_c t - \frac{1}{3}\cos 3\omega_c t + \frac{1}{5}\cos 5\omega_c t - \cdots\right)\right] \\
&= \underbrace{\frac{c}{2}\cos\omega_c t + \frac{2}{\pi}m(t)\cos\omega_c t}_{\text{AM}} + \underbrace{\text{other terms}}_{\substack{\text{suppressed by} \\ \text{bandpass filter}}}
\end{aligned}
$$

Demodulation of AM Signals

- The AM signal can be demodulated coherently by a locally generated carrier.

- However coherent or synchronous demodulation of AM will defeat the very purpose of AM and hence is rarely used in practice.

- The two noncoherent methods of AM demodulation are:
  - **Rectifier detection**
  - **Envelope detection.**

Rectifier detector

- If an AM signal is applied to a diode and a resistor circuit the negative part of the AM wave will be suppressed.

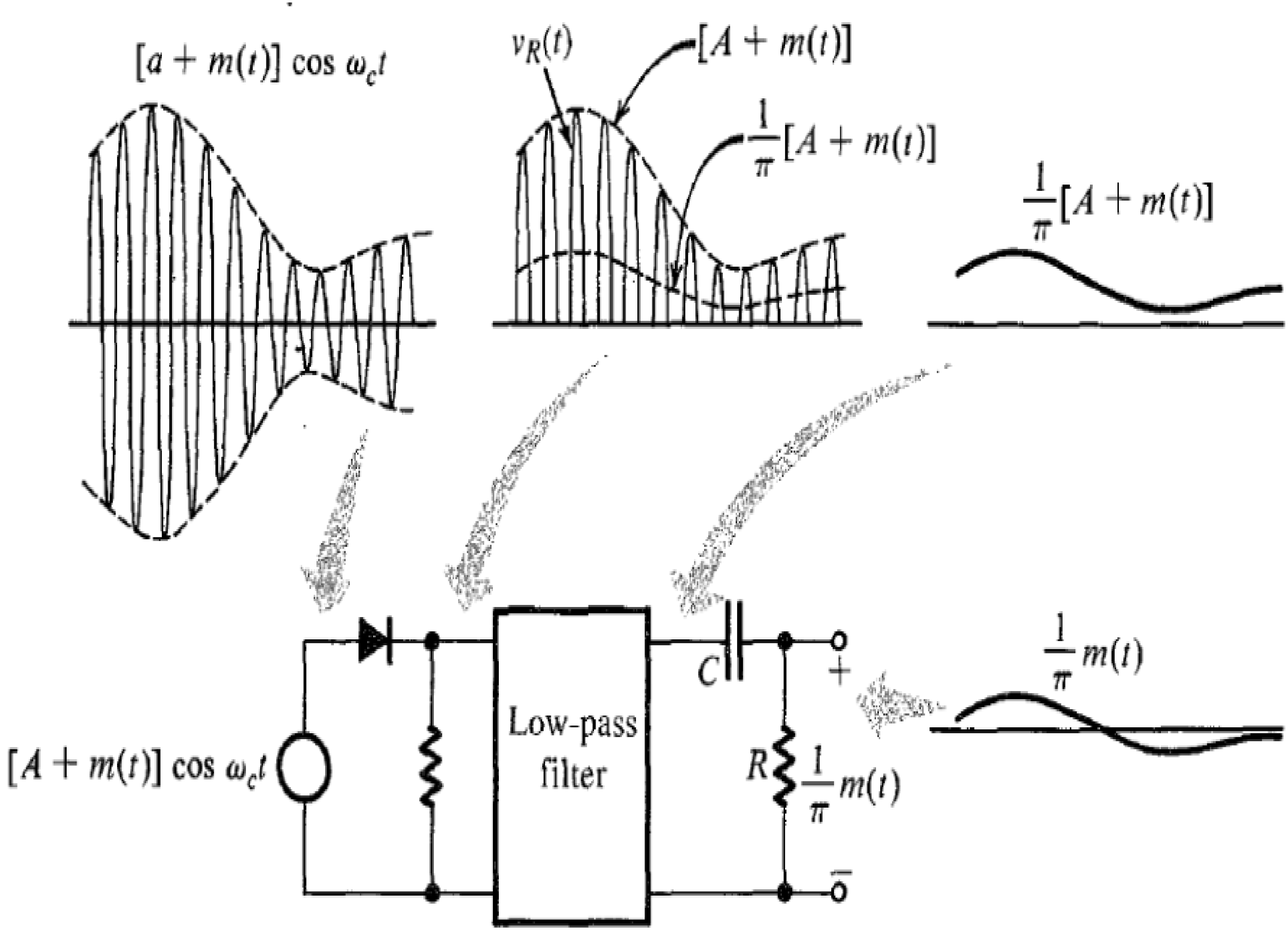
- The output across the resistor is a half-wave rectified version of the AM signal.

- In essence, the AM signal is multiplied by w(t):

$$\begin{aligned}
v_R &= \{[A + m(t)]\cos\omega_c t\}\, w(t) \\
&= [A + m(t)]\cos\omega_c t \left[\frac{1}{2} + \frac{2}{\pi}\left(\cos\omega_c t - \frac{1}{3}\cos 3\omega_c t + \frac{1}{5}\cos 5\omega_c t - \cdots\right)\right] \\
&= \frac{1}{\pi}[A + m(t)] + \text{other terms of higher frequencies}
\end{aligned}$$

- The output can be doubled by using a full wave rectifier.

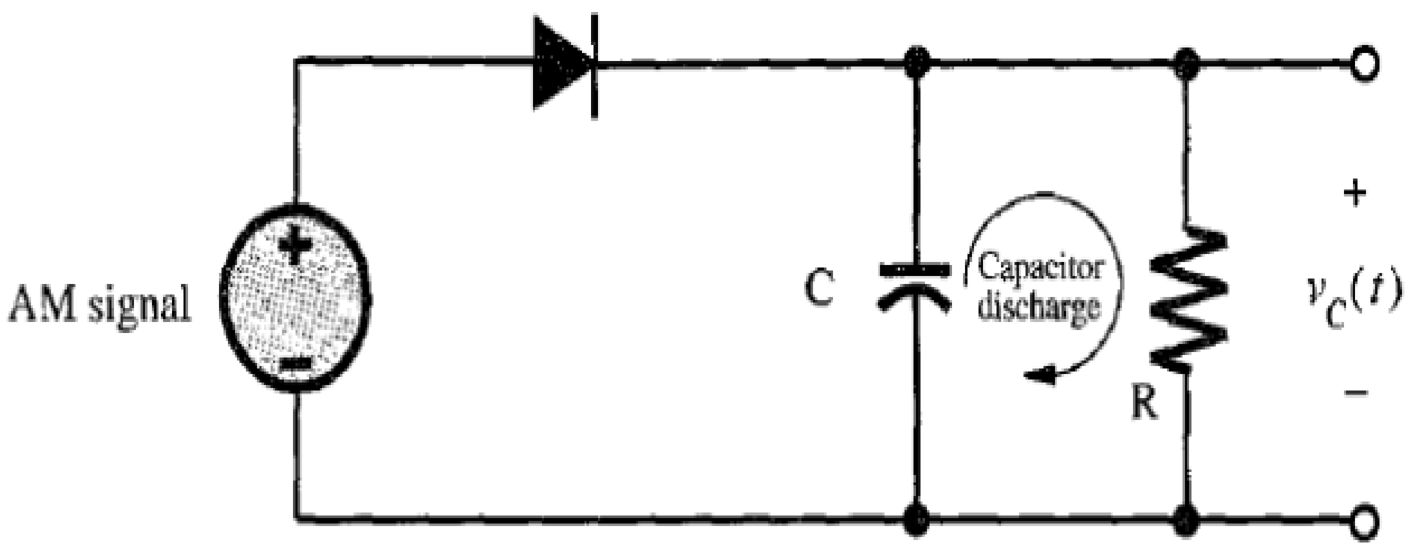
- The rectifier detection is in effect synchronous detection performed without using a local carrier.

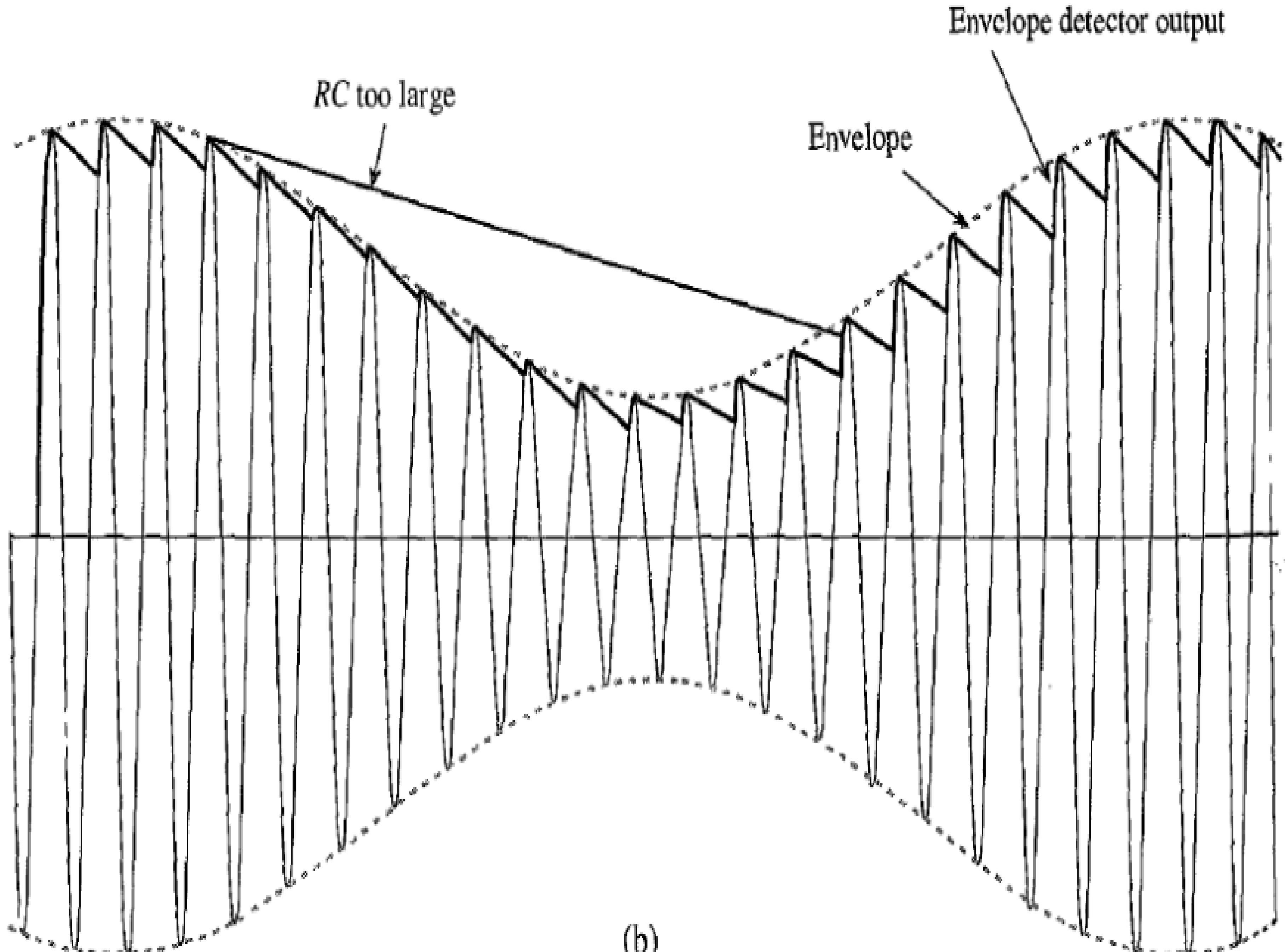


$[a + m(t)] \cos \omega_c t$
$v_R(t)$
$[A + m(t)]$
$\frac{1}{\pi}[A + m(t)]$
$\frac{1}{\pi}[A + m(t)]$
Low-pass filter
C
R
$[A + m(t)] \cos \omega_c t$
$\frac{1}{\pi} m(t)$
$\frac{1}{\pi} m(t)$

Envelope detector

- In an envelope detector, the output of the detector follows the envelope of the modulated signal.
- The rectifier detector is basically a synchronous demodulator but the envelope detection is a nonlinear operation.
- The low pass filter used in rectifier detector does not depend on the value μ.
- On the other hand, the time constant RC of the low-pass filter for the envelope detector does depend on the value of μ.





(b)

EXAMPLE 4.6 For tone modulation , determine the upper limit of RC to ensure that the capacitor voltage follows the envelope.

Quadrature Amplitude Modulation (QAM)

- The DSB signals occupy twice the bandwidth required for the baseband.

- This disadvantage can be overcome by transmitting two DSB signals using carriers of the same frequency but in phase quadrature.

- If the two baseband signals to be transmitted are m1(t) and m2(t), the corresponding QAM signal is the sum of the two DSB-modulated signals:

$$\varphi_{\text{QAM}}(t) = m_1(t)\cos\omega_c t + m_2(t)\sin\omega_c t$$

- Thus, two baseband signals, each of bandwidth B Hz, can be transmitted simultaneously over a bandwidth 2B by using DSB transmission and quadrature multiplexing.

- The upper channel is also known as the in-phase (I) channel and the lower channel is the quadrature (Q) channel.

- Both modulated signals occupy the same band. Yet two baseband signals can be separated at the receiver by synchronous detection using two local carriers in phase quadrature.

$$\begin{aligned} x_1(t) &= 2\varphi_{\text{QAM}}(t)\cos\omega_c t = 2[m_1(t)\cos\omega_c t + m_2(t)\sin\omega_c t]\cos\omega_c t \\ &= m_1(t) + m_1(t)\cos 2\omega_c t + m_2(t)\sin 2\omega_c t \end{aligned}$$

- QAM is somewhat of an exacting scheme. A slight error in the phase or the frequency of the carrier at th.e demodulator in QAM will not only result in loss and distortion of signals, but will also lead to interference between the two channels.

$$x_1(t) = 2[m_1(t)\cos\omega_c t + m_2(t)\sin\omega_c t]\cos(\omega_c t + \theta)$$
$$= m_1(t)\cos\theta + m_1(t)\cos(2\omega_c t + \theta) - m_2(t)\sin\theta + m_2(t)\sin(2\omega_c t + \theta)$$

- The low-pass filter suppresses the two signals with frequency $2w_c$, resulting in the output m1(t) cosθ – m2(t) sin θ.

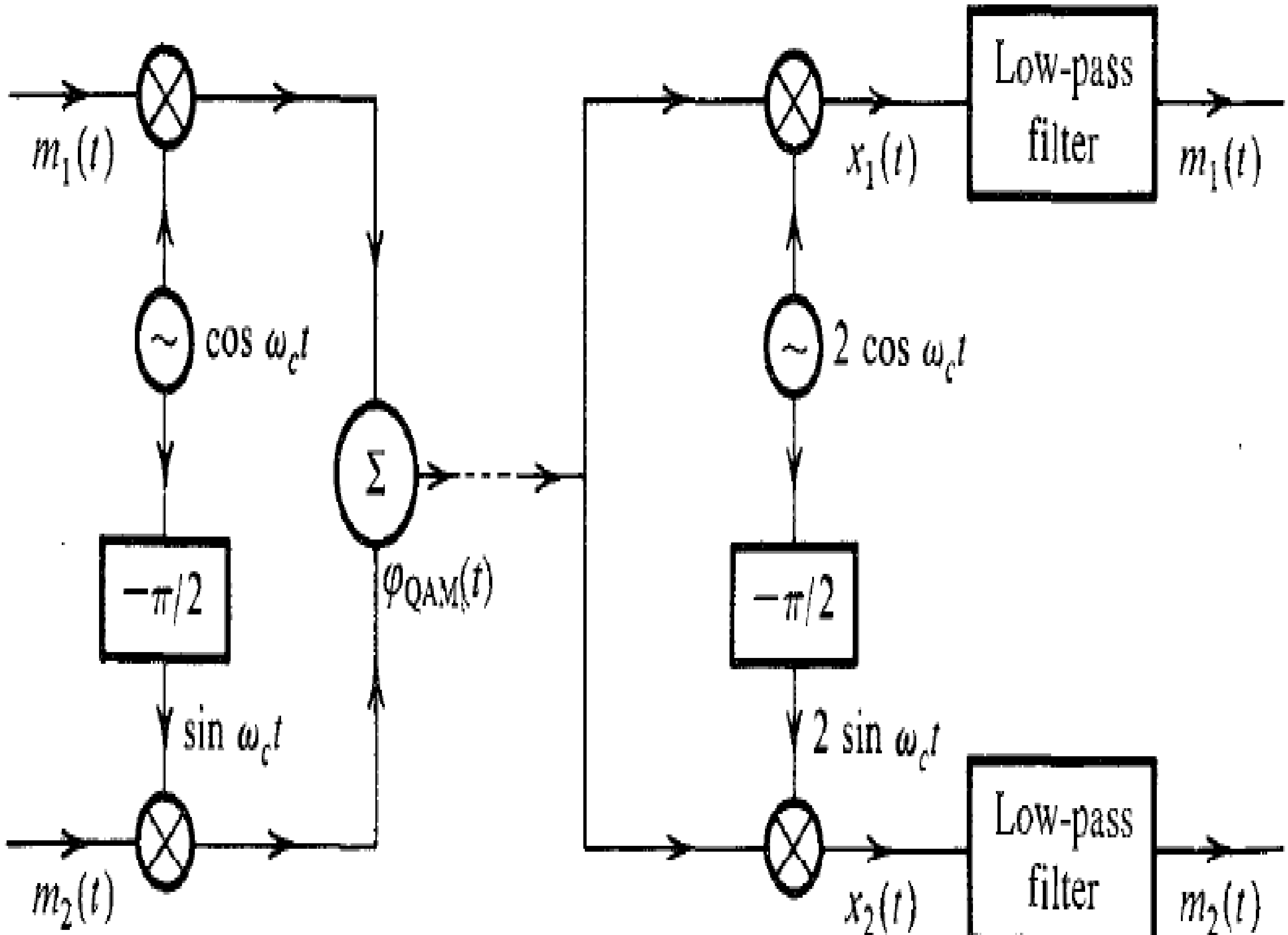
- Thus, in addition to the desired signal rn1(t), we also receive signal m2 (t) in the upper branch.

- Similar argument shows that in addition to the desired signal m2(t), we receive signal m1(t) in the lower branch. This cochannel interference is undesirable.Similar difficulties arise when the local frequency is in error.

- In addition, unequal attenuation of the USB and the LSB during transmission also leads to crosstalk or cochannel interference.

- Quadrature multiplexing is used in Color television to multiplex the so-called chrominance signals, which carry the information about colors.

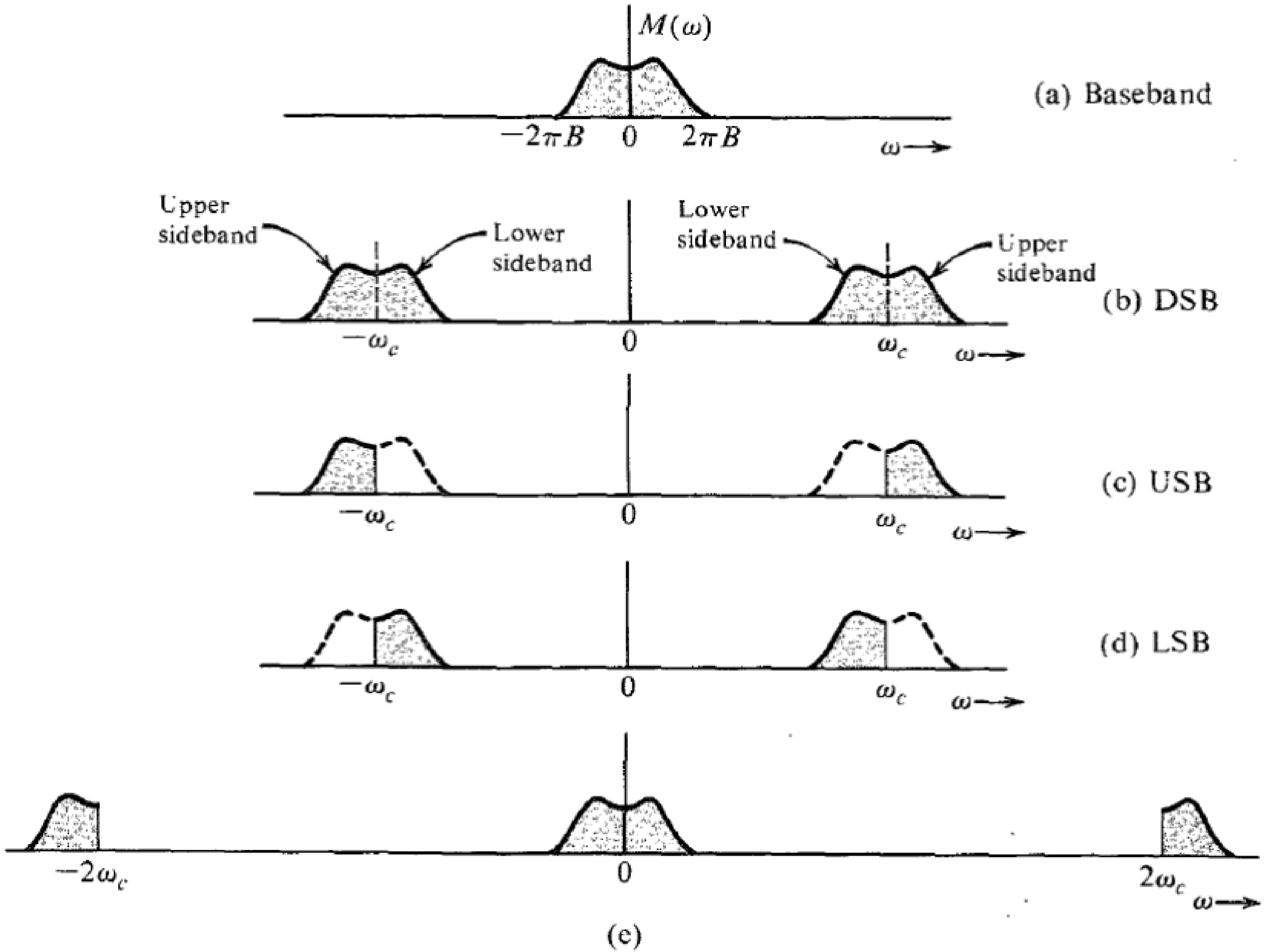
- There the synchronization is achieved by periodic insertion of a short burst of carrier signal (called color burst in the transmitted signal.



$m_1(t)$
$\cos \omega_c t$
$-\pi/2$
$\sin \omega_c t$
$m_2(t)$
Σ
$\varphi_{QAM}(t)$
$x_1(t)$
$2 \cos \omega_c t$
$-\pi/2$
$2 \sin \omega_c t$
$x_2(t)$
Low-pass filter
$m_1(t)$
Low-pass filter
$m_2(t)$

AM-Single Sideband (SSB)

- The DSB spectrum has two sidebands:
  - The upper sideband (USB) and the lower sideband (LSB), both containing the complete information of the baseband signal .
- A scheme in which only one sideband is transmitted is known as single-sideband (SSB) transmission, which requires only one-half the bandwidth of the DSB signal.
- An SSB signal can be coherently (synchronously) demodulated.
- The demodulation of SSB signals is identical to that of DSB-SC signals.
- We will discuss only SSB signals without an additional carrier and hence they are suppressed carrier signals (SSB-SC).



$M(\omega)$
(a) Baseband
$-2\pi B$
0
$2\pi B$
$\omega \rightarrow$
Upper sideband
Lower sideband
Lower sideband
Upper sideband
(b) DSB
$-\omega_c$
0
ω_c
$\omega \rightarrow$
(c) USB
$-\omega_c$
0
ω_c
$\omega \rightarrow$
(d) LSB
$-\omega_c$
0
ω_c
$\omega \rightarrow$
$-2\omega_c$
0
$2\omega_c$
$\omega \rightarrow$
(e)

Time-Domain Representation of SSB Signals

$$M_+(\omega) = M(\omega)u(\omega) \text{ and } M_-(\omega) = M(\omega)u(-\omega).$$

- Let $m_+(t)$ and $m_-(t)$ be the inverse Fourier transforms of $M_+(w)$ and $M\_(w)$ respectively.

- Because the amplitude spectra $M_+(w)$ and $M\_(w)$ are not even functions of w, the signals $m_+(t)$ and $m\_(t)$ cannot be real; they are complex. Moreover, M+(w) and M_(w) are the two halves of M(w).

$$m_+(t) = \frac{1}{2}[m(t) + jm_h(t)]$$

$$m_-(t) = \frac{1}{2}[m(t) - jm_h(t)]$$

$$\begin{aligned} M_+(\omega) &= M(\omega)u(\omega) \\ &= \tfrac{1}{2}M(\omega)[1 + \operatorname{sgn}(\omega)] \\ &= \tfrac{1}{2}M(\omega) + \tfrac{1}{2}M(\omega)\operatorname{sgn}(\omega) \end{aligned}$$

$$M_h(\omega) = -jM(\omega)\operatorname{sgn}(\omega) \qquad 1/\pi t \Longleftrightarrow -j\operatorname{sgn}(\omega)$$

$$m_h(t) = \frac{1}{\pi}\int_{-\infty}^{\infty} \frac{m(\alpha)}{t-\alpha}\,d\alpha$$

Hints

Conjugate Symmetry Property

- If g(t) is a real function of t, then G(w) and G(-w) are complex conjugates:

$$G(-\omega) = G^*(\omega)$$

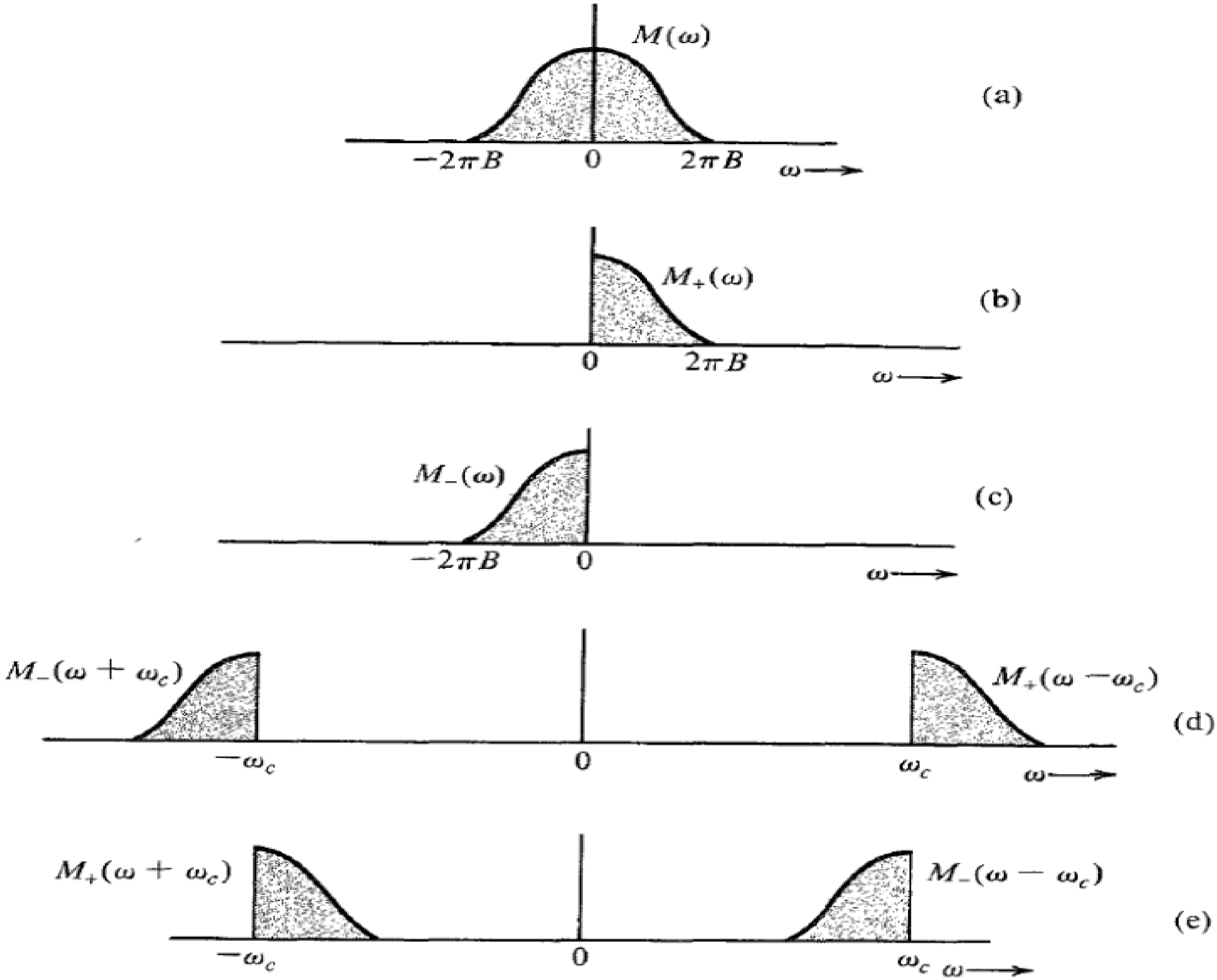
$$|G(-\omega)| = |G(\omega)|$$
$$\theta_g(-\omega) = -\theta_g(\omega)$$

- For real g (t), the amplitude spectrum is an even function, and the phase spectrum is an odd function of w.
- This property (the conjugate symmetry property) is valid only for real g(t).

Symmetry Property

if $g(t) \Longleftrightarrow G(\omega)$

then $G(t) \Longleftrightarrow 2\pi g(-\omega)$



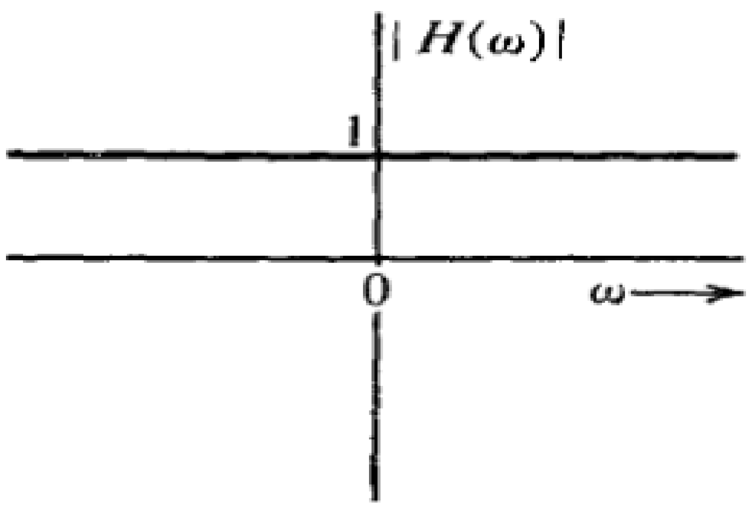
M(ω)
(a)
−2πB
0
2πB
ω
M₊(ω)
(b)
0
2πB
ω
M₋(ω)
(c)
−2πB
0
ω
M₋(ω + ωc)
M₊(ω − ωc)
(d)
−ωc
0
ωc
ω
M₊(ω + ωc)
M₋(ω − ωc)
(e)
−ωc
0
ωc
ω

- If m(t) is passed through a transfer function H (w) = -j sgn (w), then the output is $m_h(t)$, the Hilbert transform of m(t).

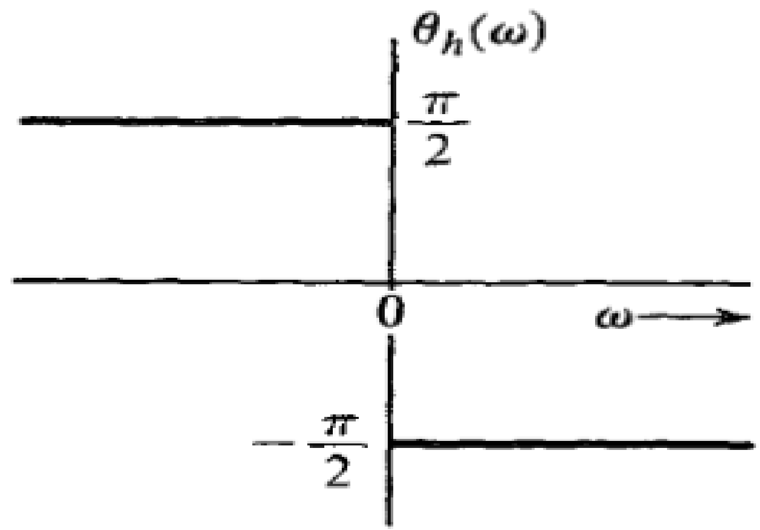
$$H(\omega) = -j \text{ sgn } (\omega)$$
$$= \begin{cases} -j = 1e^{-j\pi/2} & \omega > 0 \\ j = 1e^{j\pi/2} & \omega < 0 \end{cases}$$

- Thus, if we delay the phase of every component of m (t) by pi/2 (without changing its amplitude), the resulting signal is $m_h(t)$, the Hilbert transform of m(t).

- Therefore, a Hilbert transformer is an ideal phase shifter that shifts the phase of every spectral component by –pi/2.



(a)

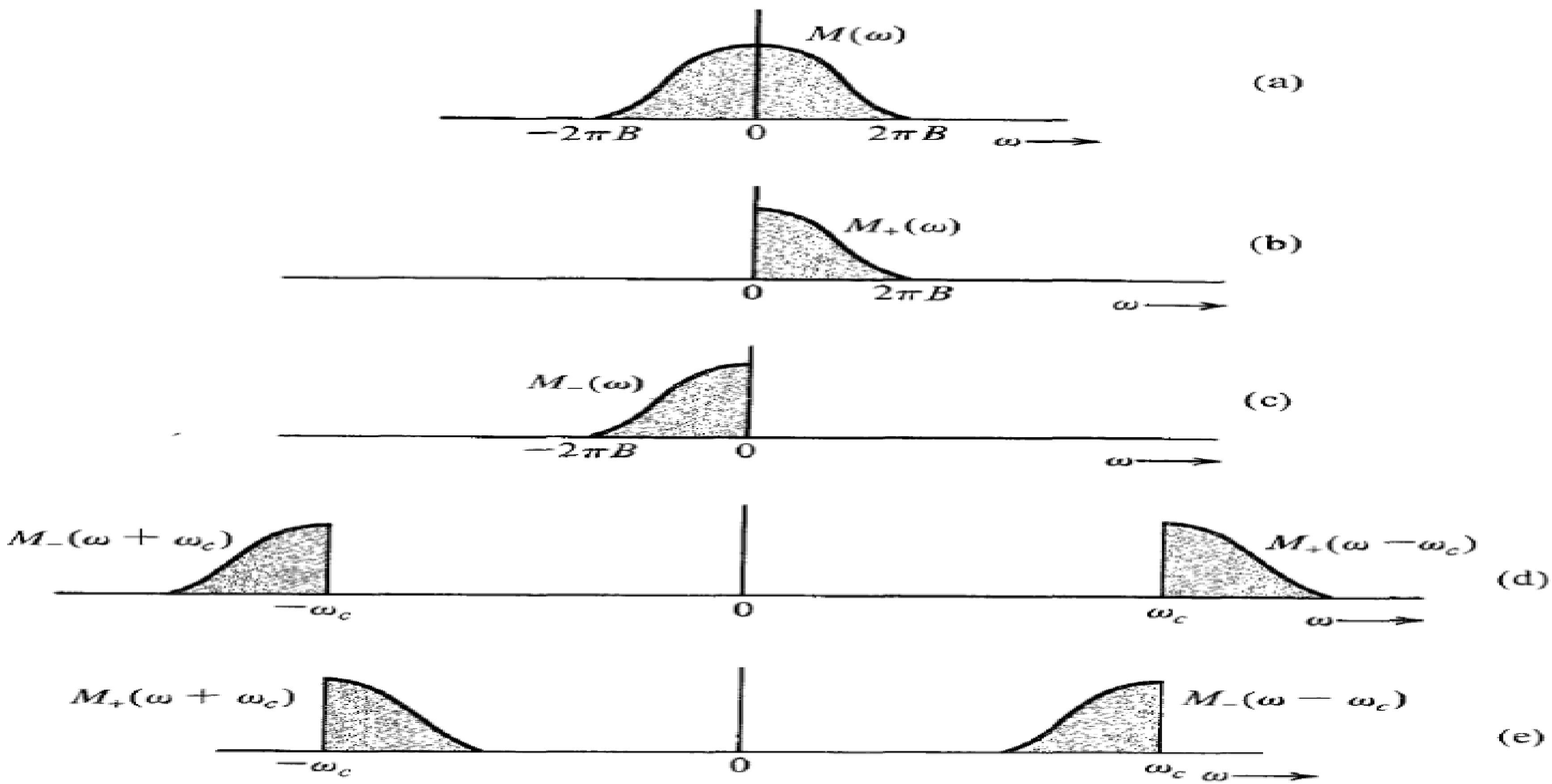


(b)

$$\Phi_{USB}(\omega) = M_+(\omega - \omega_c) + M_-(\omega + \omega_c) \qquad \varphi_{USB}(t) = m_+(t)e^{j\omega_c t} + m_-(t)e^{-j\omega_c t}$$

$$\varphi_{USB}(t) = m(t)\cos\omega_c t - m_h(t)\sin\omega_c t \qquad \varphi_{LSB}(t) = m(t)\cos\omega_c t + m_h(t)\sin\omega_c t$$

$$\varphi_{SSB}(t) = m(t)\cos\omega_c t \mp m_h(t)\sin\omega_c t$$



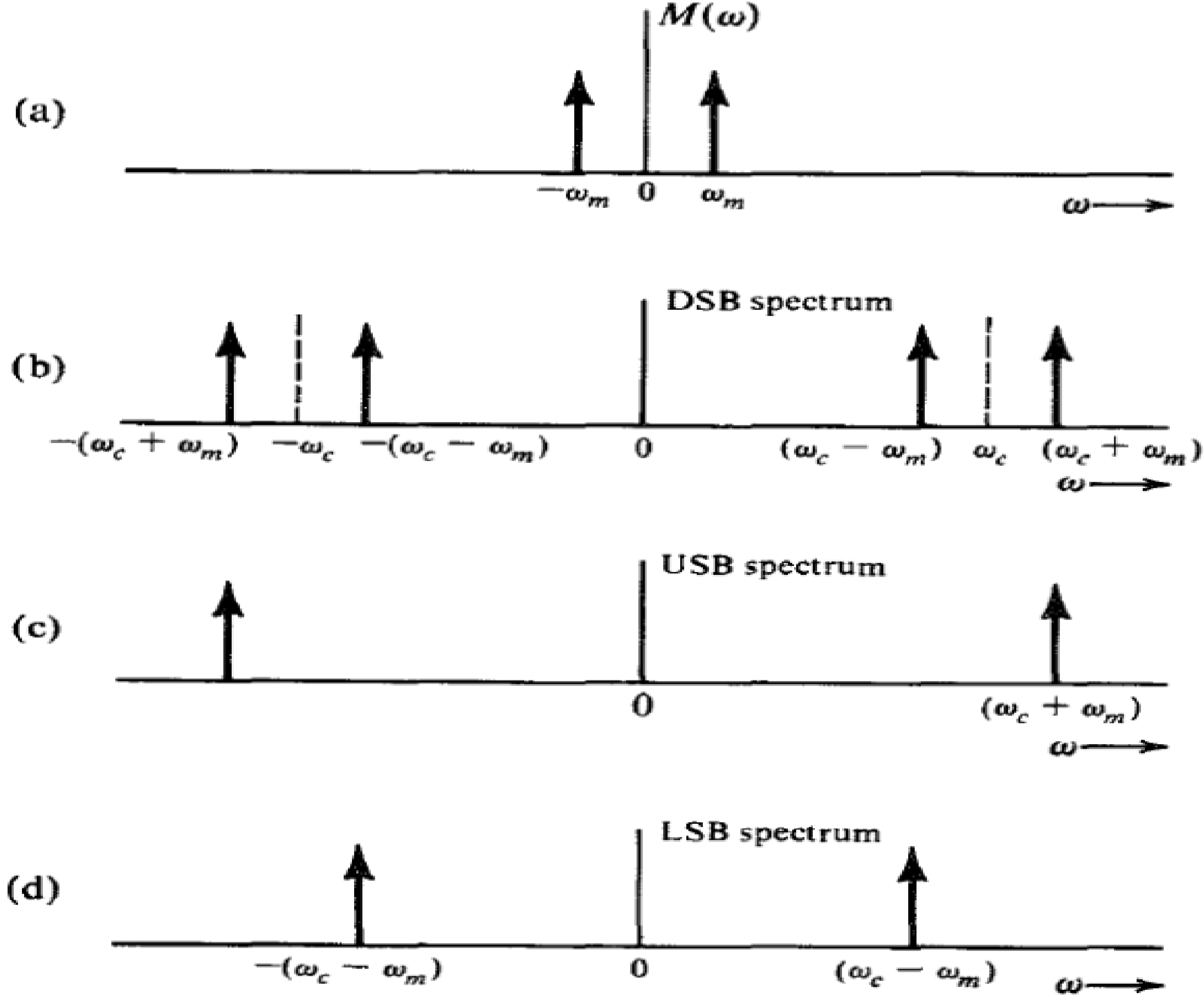
EXAMPLE 4.7 Tone Modulation: SSB

Find $\varphi_{\text{SSB}}(t)$ for a simple case of a tone modulation, that is, when the modulating signal is a sinusoid $m(t) = \cos \omega_m t$.

$$m_h(t) = \cos\left(\omega_m t - \frac{\pi}{2}\right) = \sin \omega_m t$$

$$\begin{aligned}\varphi_{\text{SSB}}(t) &= \cos \omega_m t \cos \omega_c t \mp \sin \omega_m t \sin \omega_c t \\ &= \cos (\omega_c \pm \omega_m)t\end{aligned}$$

$$\varphi_{\text{USB}}(t) = \cos (\omega_c + \omega_m)t \qquad \varphi_{\text{LSB}}(t) = \cos (\omega_c - \omega_m)t$$



(a)
M(ω)
−ωm
0
ωm
ω
(b)
DSB spectrum
−(ωc + ωm)
−ωc
−(ωc − ωm)
0
(ωc − ωm)
ωc
(ωc + ωm)
ω
(c)
USB spectrum
0
(ωc + ωm)
ω
(d)
LSB spectrum
−(ωc − ωm)
0
(ωc − ωm)
ω

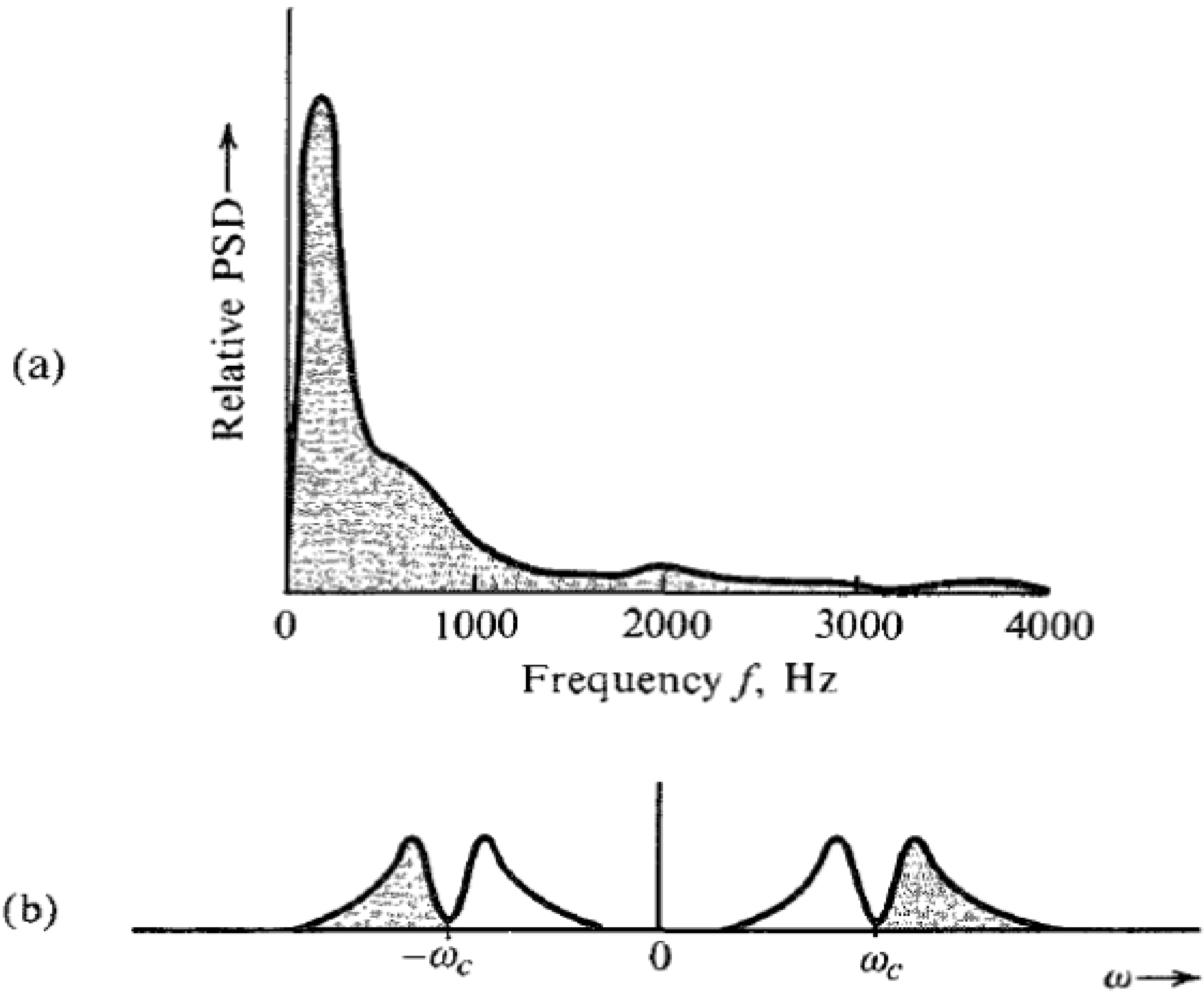
Generation of SSB Signals

- Two methods are commonly used to generate SSB signals are:
  - Selective-Filtering Method
  - Phase-Shift Method

- The first method uses sharp cutoff filters to eliminate the undesired sideband, and the second method uses phase-shifting networks.

Selective-Filtering Method

- This is the most commonly used method of generating SSB signals.
- In this method a DSB-SC signal is passed through a sharp cutoff filter to eliminate the undesired sideband.
- To obtain the USB, the filter should pass all components above w_c unattenuated and completely suppress all components below w_c. Such an operation requires an ideal filter, which is unrealizable.
- Such filter can however, be realized closely if there is some separation between the pass band and the stop band.
- Fortunately, the voice signal provides this condition, because its spectrum shows little power content at the origin in addition articulation tests have shown that for speech signals, frequency components below 300 Hz are not important.
- In other words, we may suppress "all speech components below 300 Hz without affecting the intelligibility appreciably.
- Thus, filtering of the unwanted sideband becomes relatively easy for speech signals because we have a 600-Hz transition region around the cutoff frequency $w_{c.}$ To minimize adjacent channel interference, the undesired sideband should be attenuated at least 40 dB.
- For very high carrier frequencies, the ratio of the gap band (600 Hz) to the carrier frequency may be too small, and, thus, a transition of 40 dB in amplitude over 600 Hz may pose a problem.
- In such a case, the modulation is carried out using a smaller carrier frequency first(w_{c1}). The resulting SSB signal effectively widens the gap to $2w_{c1}$. Now, treating this signal as the new baseband signal, it is possible to SSB-modulate the high-frequency carrier.


(a)
Relative PSD →
0
1000
2000
3000
4000
Frequency f, Hz
(b)
$-\omega_c$
0
ω_c
$\omega \rightarrow$

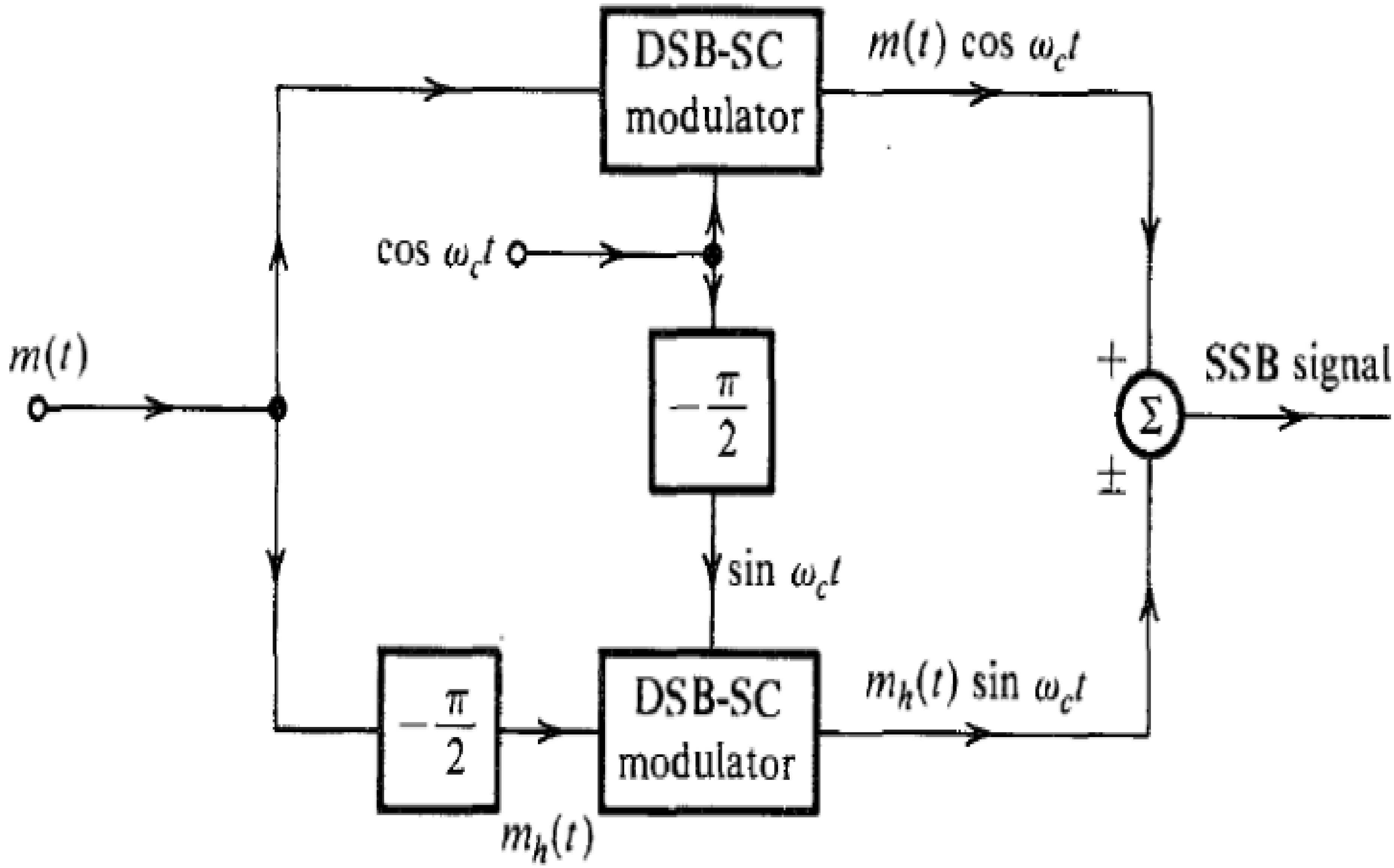
Phase-Shift Method

- The implementation of this method is based on:

$$\varphi_{\text{SSB}}(t) = m(t)\cos\omega_c t \mp m_h(t)\sin\omega_c t$$

- An ideal phase shifter is unrealizable. We can at most approximate it over a finite band.
- However, it is possible to realize a filter with two outputs such that both outputs have the same (constant) amplitude spectrum, but their phase spectra differ by pi/2 rad over a given band of frequencies.
- In terms of bandwidth requirement, SSB is similar to QAM but less exacting in terms of the carrier frequency and phase or the requirement of a distortionless transmission medium.
- However, SSB is difficult to generate if the baseband signal has no dc null in its spectrum.
- It is easy to build a circuit to shift the phase of a single frequency component by pi/2 rad, but a device to achieve a pi/2 phase shift of all the spectral components over a band of frequencies is unrealizable.

$$\varphi_{\text{SSB}}(t) = m(t)\cos\omega_c t \mp m_h(t)\sin\omega_c t$$



Demodulation of SSB-SC Signals

- SSB-SC signals can be coherently demodulated.

$$\varphi_{\text{SSB}}(t) = m(t)\cos\omega_c t \mp m_h(t)\sin\omega_c t$$

$$\begin{aligned}\varphi_{\text{SSB}}(t)\cos\omega_c t &= \tfrac{1}{2}m(t)[1+\cos 2\omega_c t] \mp \tfrac{1}{2}m_h(t)\sin 2\omega_c t \\ &= \tfrac{1}{2}m(t) + \tfrac{1}{2}[m(t)\cos 2\omega_c t \mp m_h(t)\sin 2\omega_c t]\end{aligned}$$

- So SSB-SC demodulator is identical to the synchronous demodulator used for DSB-SC.

- Thus, any one of the synchronous DSB-SC demodulators can be used to demodulate an SSB-SC signal.

Envelope Detection of SSB Signals with a Carrier (SSB+C)

$$\varphi_{\mathrm{SSB+C}} = A\cos\omega_c t + [m(t)\cos\omega_c t + m_h(t)\sin\omega_c t]$$

$$\begin{aligned}\varphi_{\mathrm{SSB+C}} &= [A + m(t)]\cos\omega_c t + m_h(t)\sin\omega_c t \\ &= E(t)\cos(\omega_c t + \theta)\end{aligned}$$

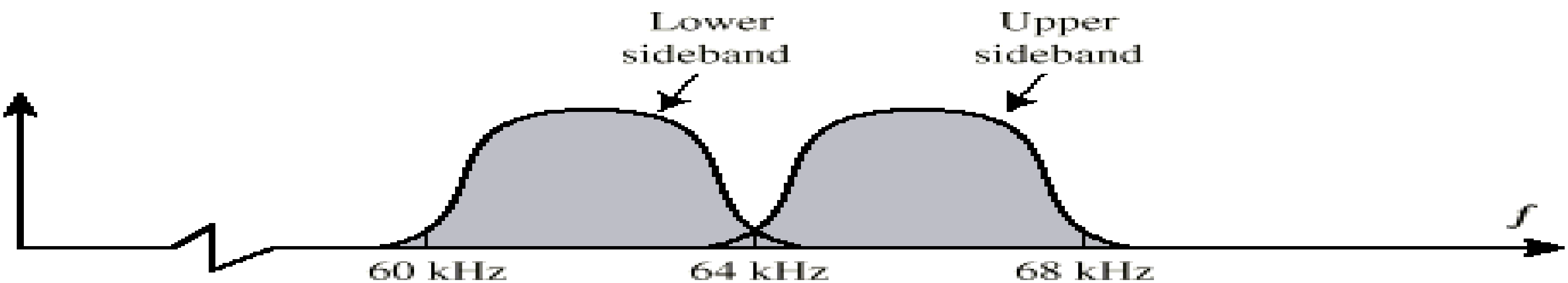
$$\begin{aligned}E(t) &= \{[A + m(t)]^2 + m_h^2(t)\}^{1/2} \\ &= A\left[1 + \frac{2m(t)}{A} + \frac{m^2(t)}{A^2} + \frac{m_h^2(t)}{A^2}\right]^{1/2}\end{aligned}$$

$$E(t) \simeq A\left[1 + \frac{2m(t)}{A}\right]^{1/2}$$

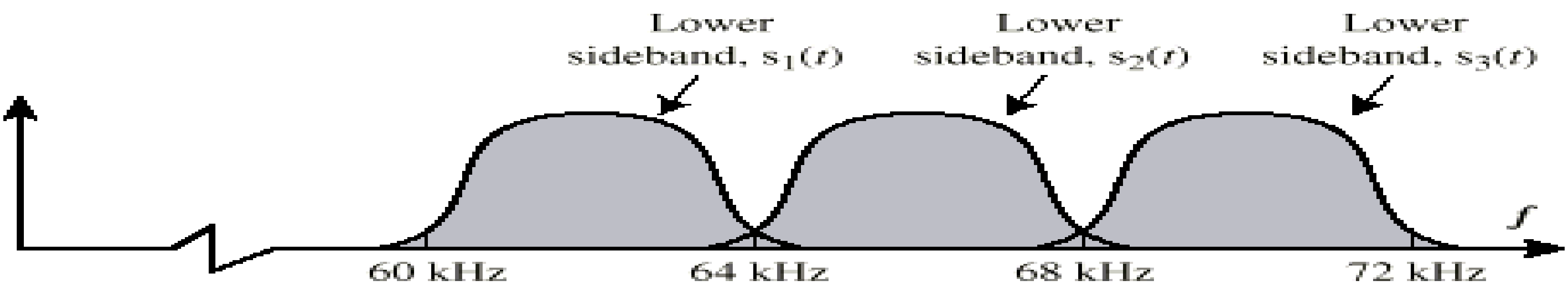
$$\begin{aligned}E(t) &\simeq A\left[1 + \frac{m(t)}{A}\right] \\ &= A + m(t)\end{aligned}$$

Telephone-Channel Multiplexing

(a) Spectrum of $m_1(t)$, positive f



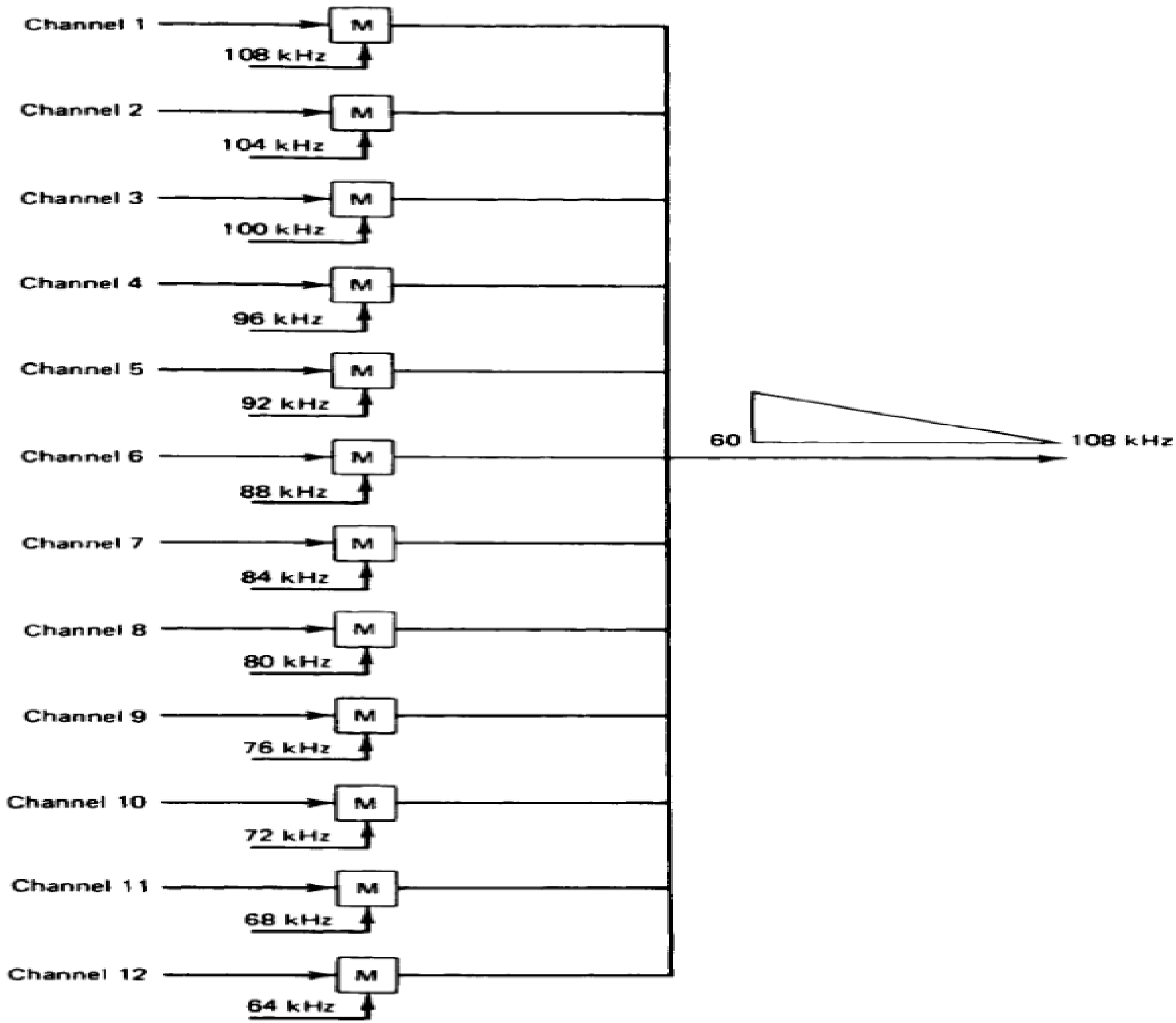
(b) Spectrum of $s_1(t)$ for $f_1 = 64$ kHz



(c) Spectrum of composite signal using subcarriers at 64 kHz, 68 kHz, and 72 kHz

Standard/Basic Group

- Multiplexing of 12 Voice channels to form a standard group.
- Frequencies assigned to standard group are from 60kHz to 108 kHz.
- The LSB are selected while USB are rejected after mixing process.
- The mixing process causes frequencies inversion.



Channel 1
108 kHz
M
Channel 2
104 kHz
M
Channel 3
100 kHz
M
Channel 4
96 kHz
M
Channel 5
92 kHz
M
Channel 6
88 kHz
M
Channel 7
84 kHz
M
Channel 8
80 kHz
M
Channel 9
76 kHz
M
Channel 10
72 kHz
M
Channel 11
68 kHz
M
Channel 12
64 kHz
M
60
108 kHz

Super Group

- Multiplexing of 5 standard groups to form a Super group.

- Frequencies assigned to Super group are from 312kHz to 552 kHz.

- The carrier frequencies are assigned as

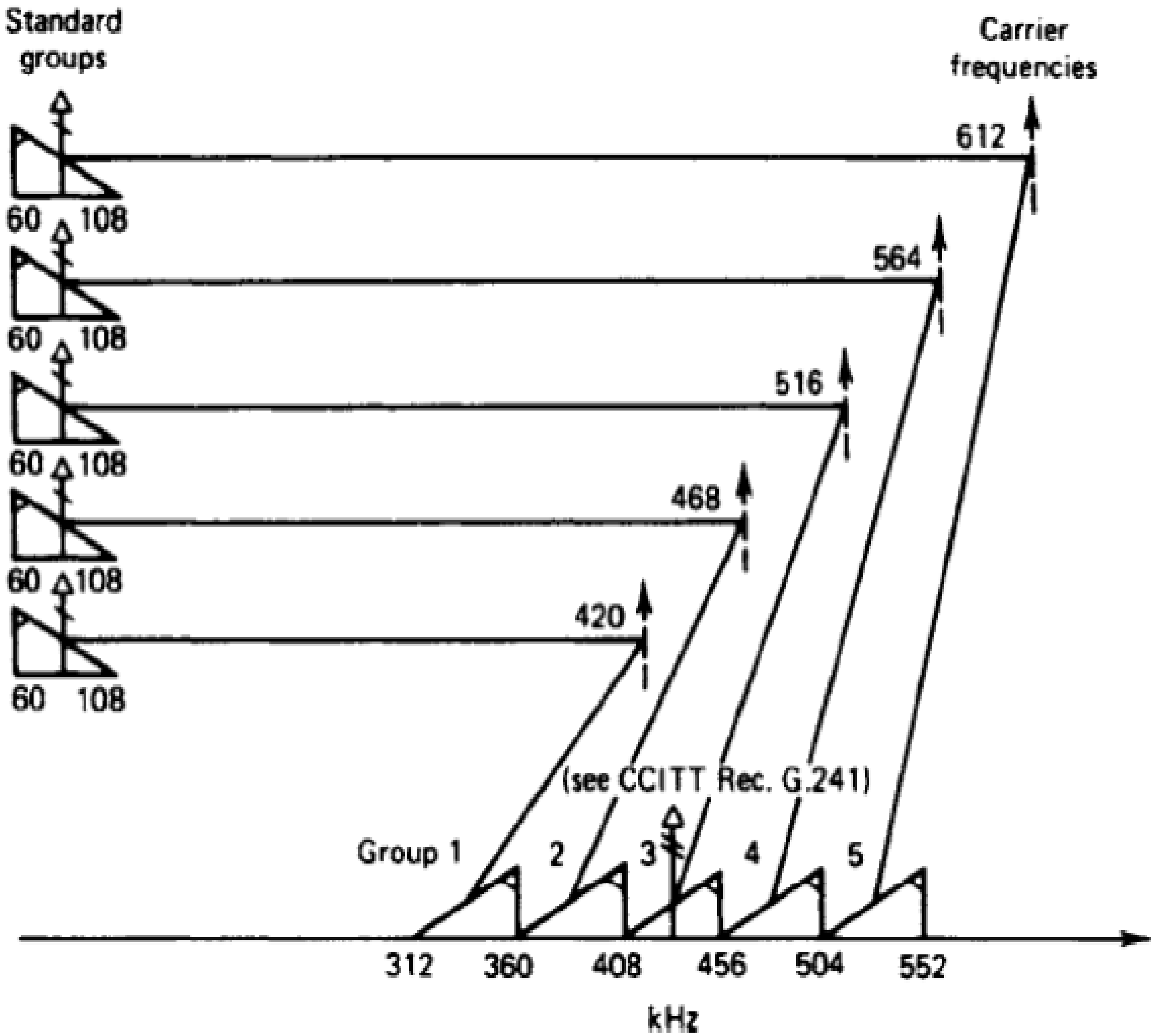
  For Standard group1:420KHz
  For Standard group1:468KHz
  For Standard group1:516KHz
  For Standard group1:564KHz
  For Standard group1:612KHz

- The LSB are selected while USB are rejected after mixing process. The mixing process causes frequencies inversion.

- A basic master group of 600 channels is formed by multiplexing 10 super groups in North American Hierarchy.

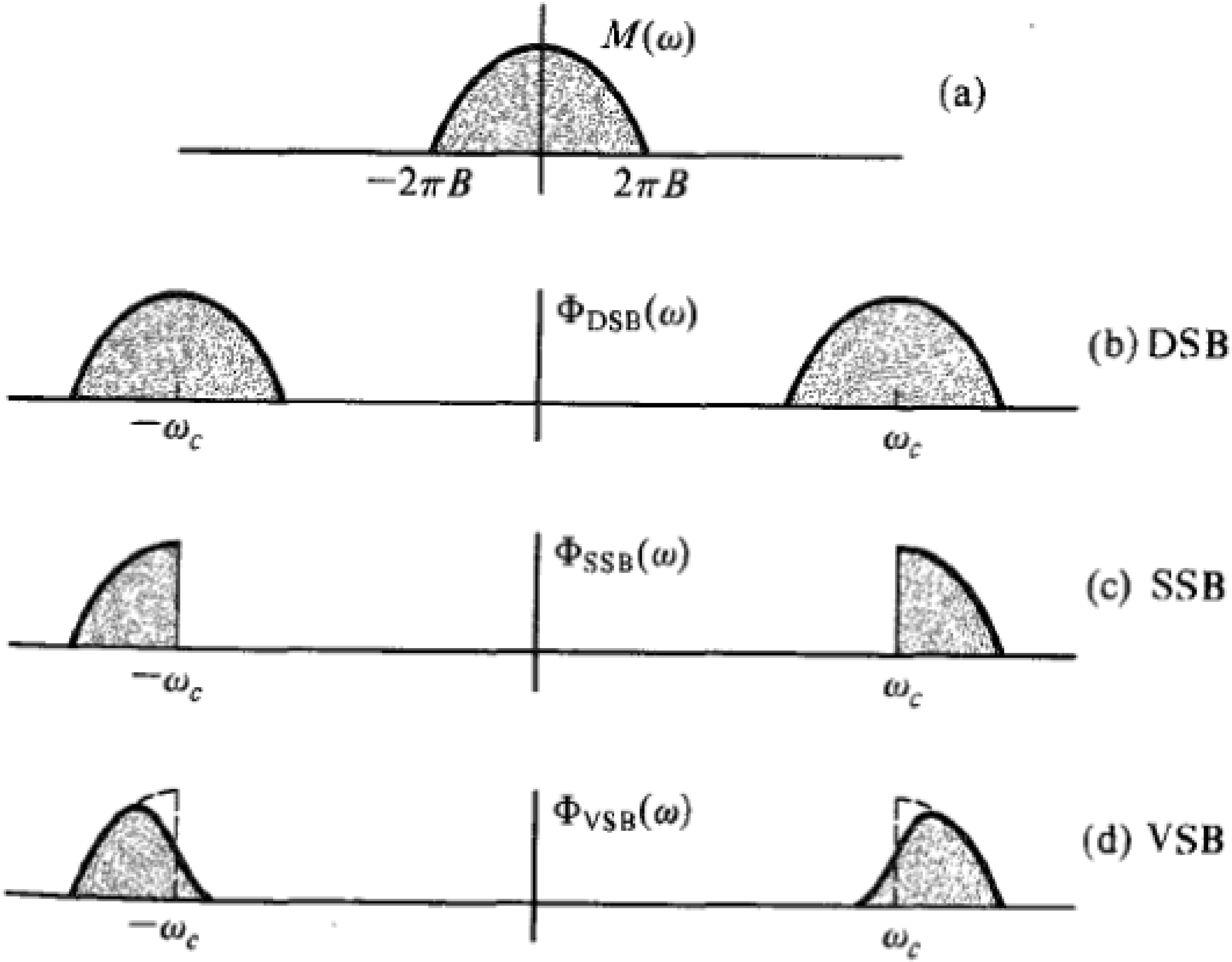
- In the CCITT hierarchy, a basic master group is formed by multiplexing five super groups (300 voice channels).

- There are two standard master group configurations: the L600 and the U600.

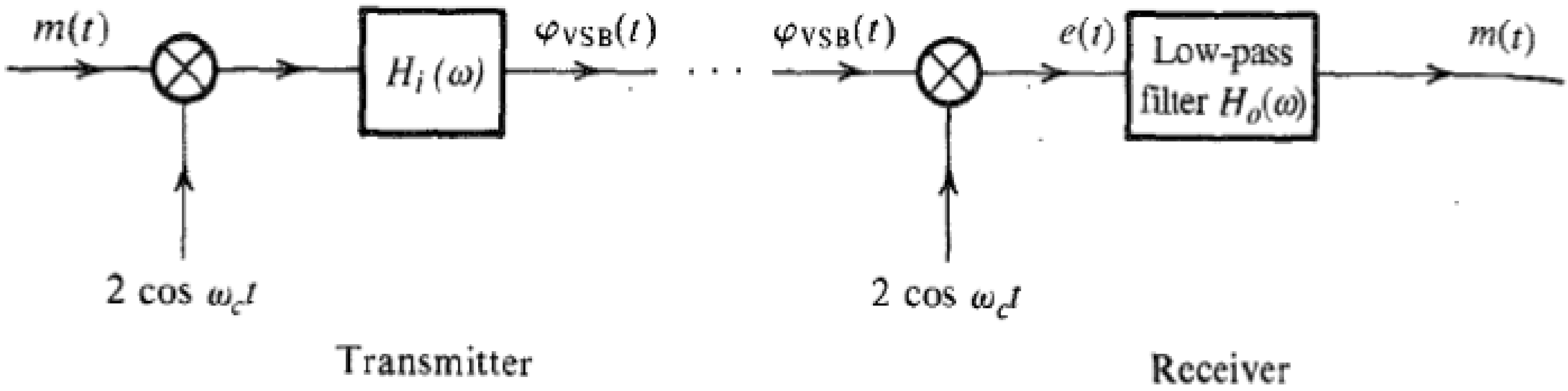

Standard groups
Carrier frequencies
612
564
516
468
420
60
108
(see CCITT Rec. G.241)
Group 1
2
3
4
5
312
360
408
456
504
552
kHz

AM-Vestigial Sideband (VSB)

- The generation of SSB signals is rather difficult:
  - The selective-filtering method demands dc null in the modulating signal spectrum.
  - A phase shifter required in the phase- shift method is unrealizable, or realizable only approximately.
- The generation of DSB signals is much simpler, but requires twice the signal bandwidth.
- A vestigial-sideband (VSB), also called asymmetric sideband system is a compromise between DSB and SSB.
- It inherits the advantages of DSB and SSB but avoids their disadvantages at a small cost.
- VSB signals are relatively easy to generate and at the same time, their bandwidth is only (typically 25%) greater than that of SSB signals.
- In VSB, instead of rejecting one sideband completely (as in SSB), a gradual cutoff of one sideband is accepted.
- The baseband signal can be recovered exactly by a synchronous detector in conjunction with an appropriate equalizer filter $H_o(w)$ at the receiver output.
- If a large carrier is transmitted along with the VSB signal, the baseband signal can be recovered by an envelope (or a rectifier) detector.



M(ω)
(a)
−2πB
2πB
Φ_DSB(ω)
(b) DSB
−ω_c
ω_c
Φ_SSB(ω)
(c) SSB
−ω_c
ω_c
Φ_VSB(ω)
(d) VSB
−ω_c
ω_c



- If the vestigial shaping filter that produces VSB from DSB is $H_i(w)$ then the resulting VSB signal spectrum is:

$$\Phi_{\mathrm{VSB}}(\omega) = [M(\omega+\omega_c) + M(\omega-\omega_c)]H_i(\omega)$$

- VSB shaping filter $H_i(w)$ allows the transmission of one sideband, but suppresses the other sideband not completely but gradually.

- This makes it easy to realize such a filter, but the transmission bandwidth is now somewhat higher than that of the SSB.

- We require that m(t) be recoverable from VSB modulated signal using synchronous demodulation at the receiver. This is done by multiplying the incoming VSB signal with carrier. The product e(t) is given by:

$$e(t) = 2\varphi_{\text{VSB}}(t)\cos\omega_c t \Longleftrightarrow [\Phi_{\text{VSB}}(\omega+\omega_c) + \Phi_{\text{VSB}}(\omega-\omega_c)]$$

- The signal e (t) is further passed through the low-pass equalizer filter of transfer function $H_o(w)$.

- The output of the equalizer filter is required to be m(t). Hence, the output signal spectrum is given by:

$$M(\omega) = [\Phi_{\text{VSB}}(\omega+\omega_c) + \Phi_{\text{VSB}}(\omega-\omega_c)]H_o(\omega)$$

but

$$\Phi_{\text{VSB}}(\omega) = [M(\omega+\omega_c) + M(\omega-\omega_c)]H_i(\omega)$$

so

$$M(\omega) = M(\omega)[H_i(\omega+\omega_c) + H_i(\omega-\omega_c)]H_o(\omega)$$

$$H_o(\omega) = \frac{1}{H_i(\omega+\omega_c) + H_i(\omega-\omega_c)} \qquad |\omega| \le 2\pi B$$

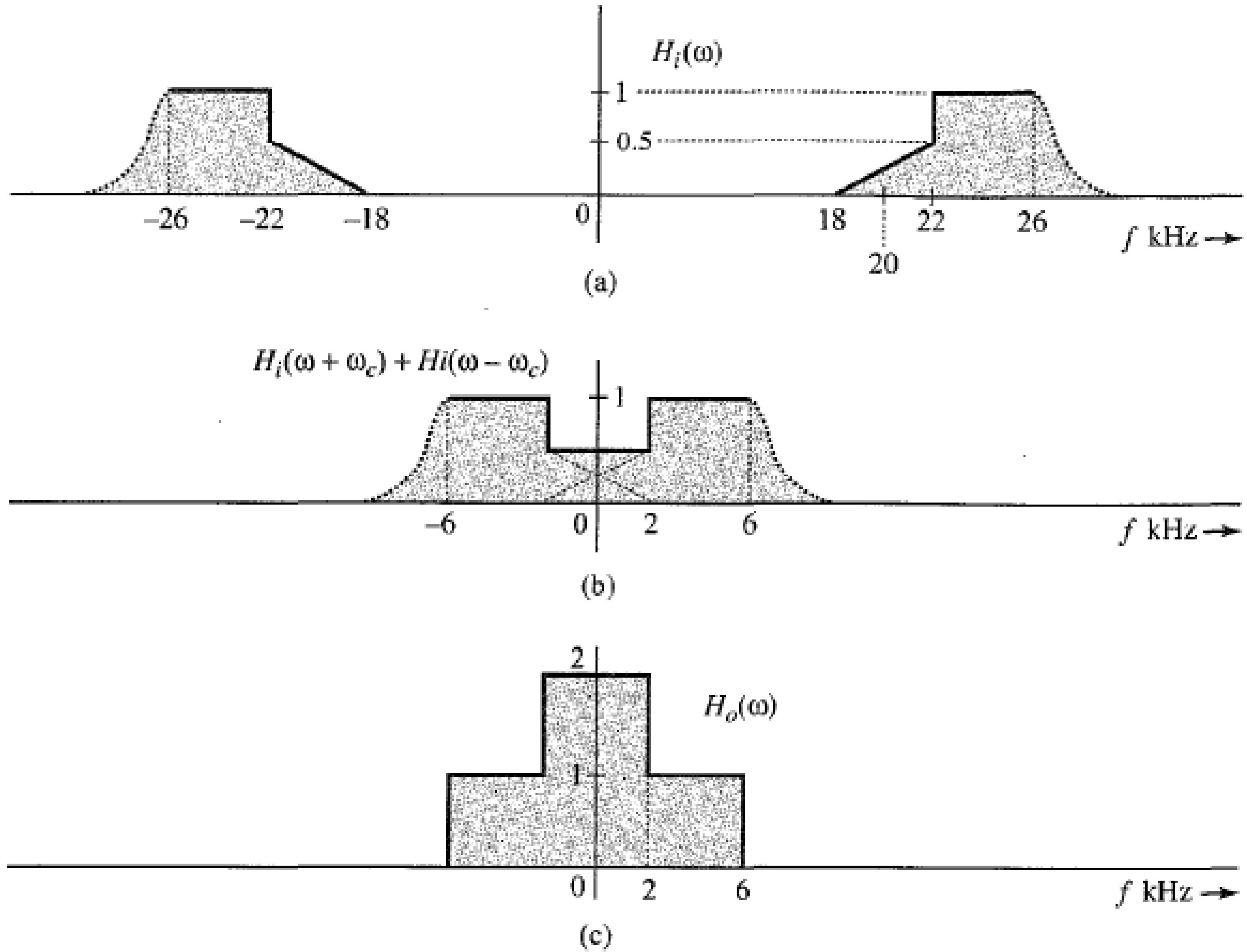
because $H_i(w)$ is a band pass filter, the terms $H_i(w \pm wc)$ contain low-pass components.

If we choose $H_i(\omega)$ such that

$$H_i(\omega+\omega_c) + H_i(\omega-\omega_c) = 1 \qquad |\omega| \le 2\pi B$$

The output filter is just a simple low-pass filter with transfer function $H_o(\omega) = 1$ over the baseband $|\omega| = 2\pi B$,

EXAMPLE 4.8 The carrier frequency of a certain VSB signal is $\omega_c = 20$ kHz, and the baseband signal bandwidth is 6 kHz. The VSB shaping filter $H_i(\omega)$ at the input, which cuts off the lower sideband gradually over 2 kHz, is shown in Fig. 4.23a. Find the output filter $H_o(\omega)$ required for distortionless reception.



Envelope Detection of VSB+C Signals

- VSB+C signals can be envelope detected in the same way as that for SSB+C signals.

- Both the SSB and the VSB modulated signals have the same form, with $m_h(t)$ in SSB replaced by some other signal $m_s(t)$ in VSB.

- SSB+C requires a much larger carrier than DSB+C (AM) for envelope detection.

- Because VSB+C is an in-between case, the added carrier required in VSB is larger than that in AM, but smaller than that in SSB+C.

Use of VSB in Broadcast Television

- VSB is a clever compromise between SSB and DSB, which makes it very attractive for television broadcast systems.

- The baseband video signal of television occupies an enormous bandwidth of 4.5 MHz, and a DSB signal needs a bandwidth of 9 MHz.

- It would seem desirable to use SSB in order to conserve the bandwidth. Unfortunately, this creates several problems.

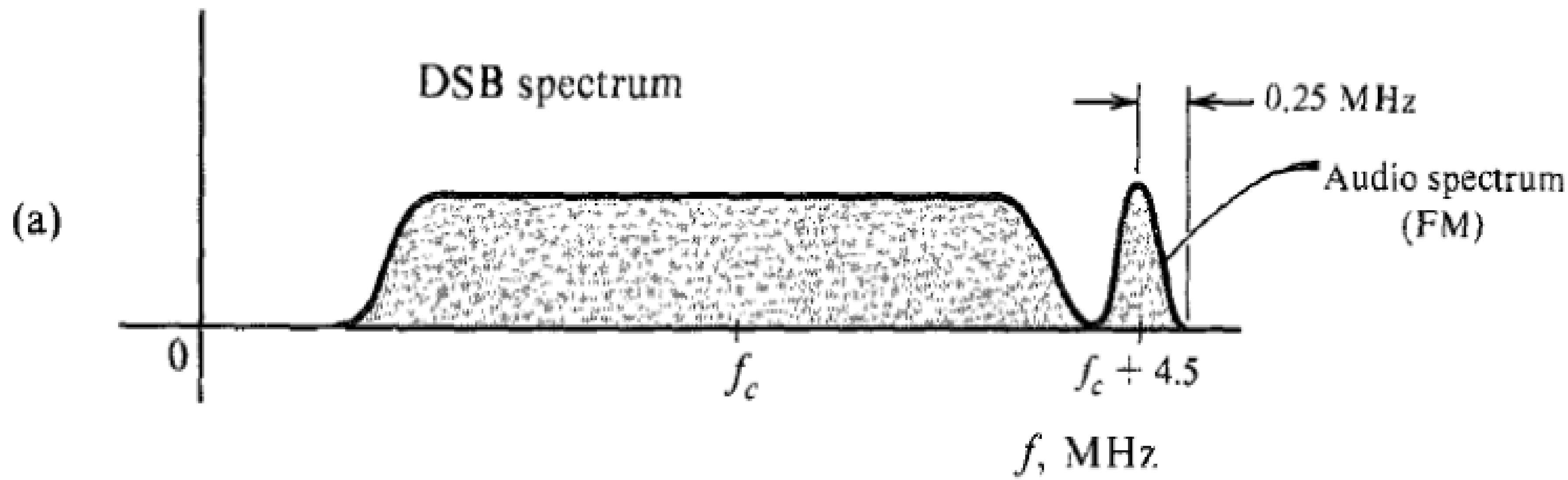
  - First, the baseband video signal has sizable power in the low-frequency region, and consequently it is difficult to suppress one sideband completely.

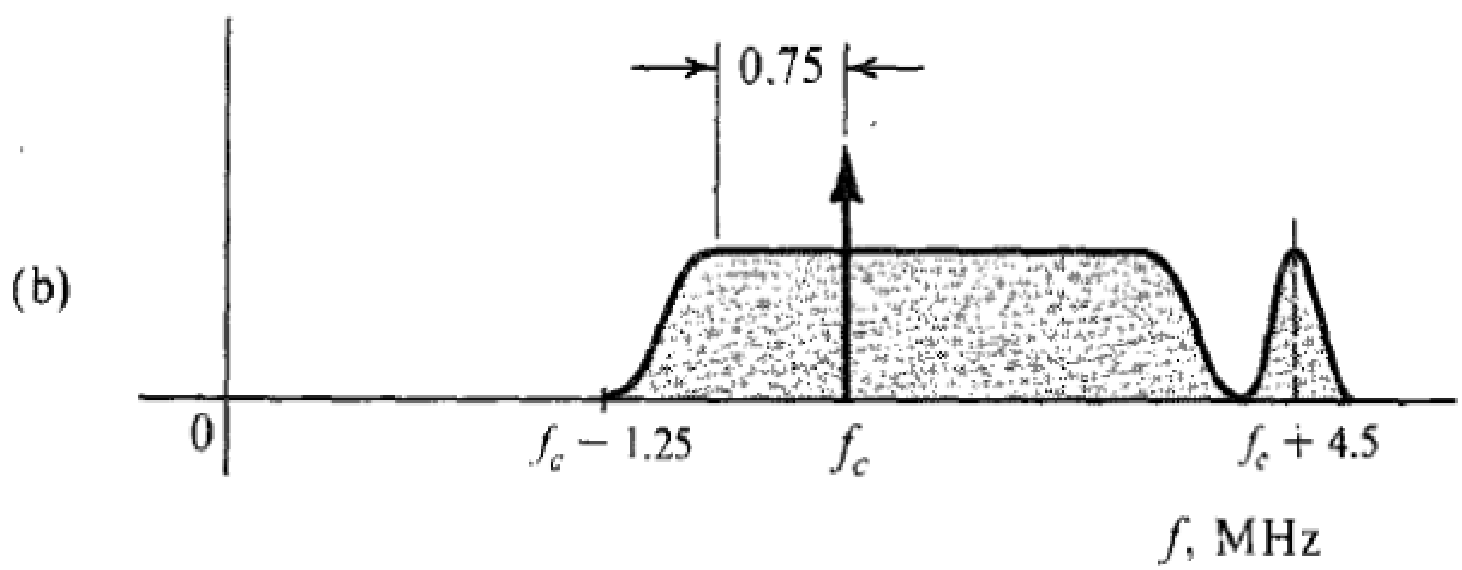
  - Second, for a broadcast receiver, an envelope detector is preferred over a synchronous one in order to reduce the receiver cost but SSB+C has a very low power efficiency and use of SSB will increase the receiver cost.

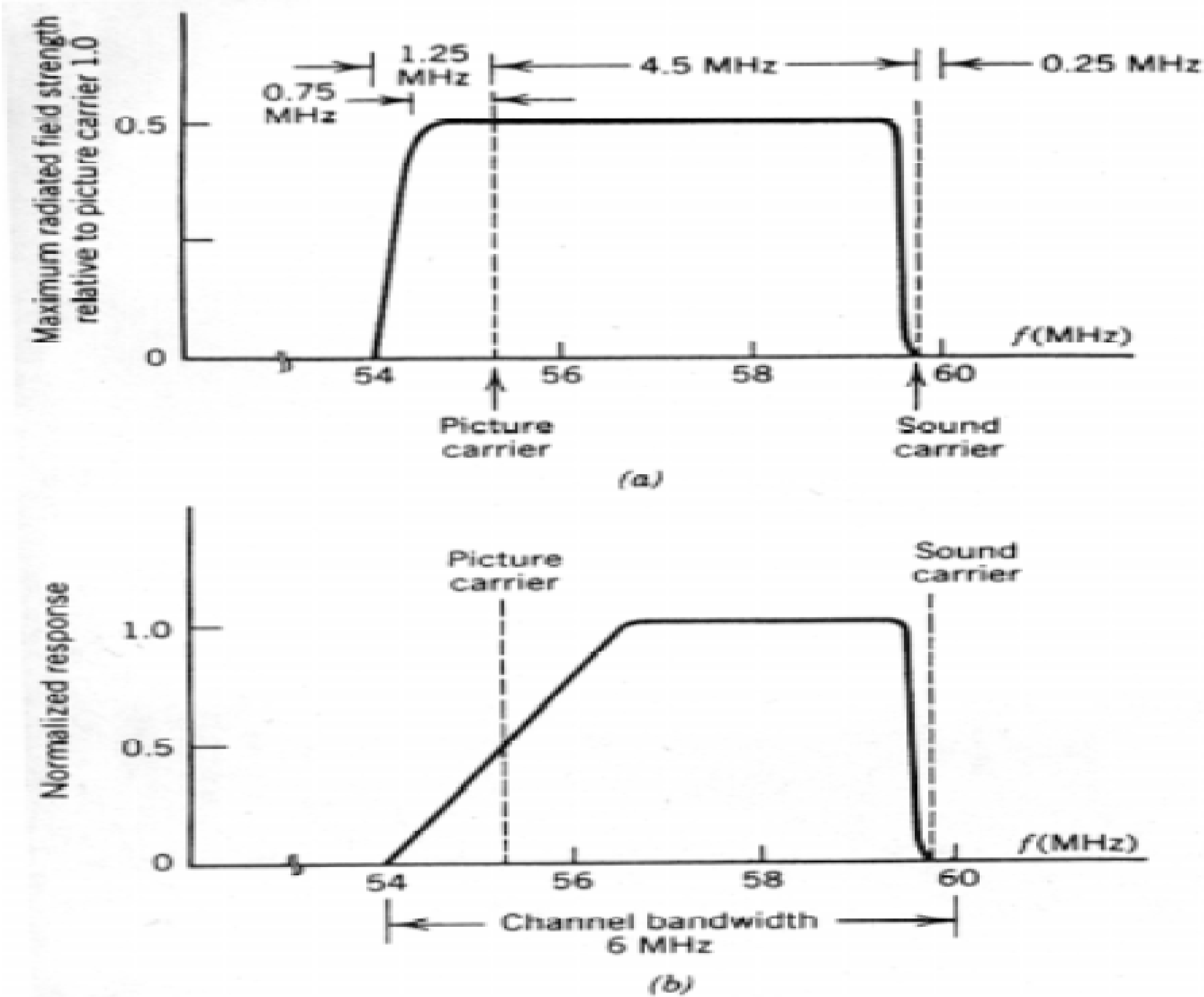
- The vestigial shaping filter $H_i(w)$ cuts off the lower sideband spectrum gradually starting at 0.75 MHz to 1.25 MHz below the carrier frequency fc.

- The receiver output filter $H_o(w)$ is designed accordingly and the resulting VSB spectrum bandwidth is 6 MHz as compared to DSB bandwidth of 9 MHz and the SSB bandwidth of 4.5 MHz.

Use of VSB in Broadcast Television






Maximum radiated field strength relative to picture carrier 1.0
0.75 MHz
1.25 MHz
4.5 MHz
0.25 MHz
0.5
0
54
56
58
60
f (MHz)
Picture carrier
Sound carrier
(a)
Normalized response
1.0
0.5
0
Picture carrier
Sound carrier
54
56
58
60
f (MHz)
Channel bandwidth 6 MHz
(b)

Linearity of Amplitude Modulation

- In all the types of modulation (DSB, SSB, AM, and VSB) the modulated signal satisfies the principles of superposition.

- For example, if modulating signals $m_1(t)$ and $m_2(t)$ produce modulated signals φ1(t) and φ2(t) respectively, then the modulating $k1m_1(t) + k2m_2(t)$ produces the modulated signal $k1\varphi_1(t) + k2\,\varphi_2(t)$.

- Because any signal can be expressed as a sum (discrete or in continuum) of sinusoids, the complete description of the modulation system can be expressed in terms of tone modulation.

$$\cos \omega_m t \cos \omega_c t = \frac{1}{2}[\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t]$$

- We can generalize this result to any non-sinusoidal modulating signal m(t).

Carrier Acquisition

Carrier Acquisition

- In the suppressed-carrier amplitude-modulated system (DSB-SC, SSB-SC, and VSB-SC), one must generate a local carrier at the receiver for the purpose of synchronous demodulation.

- Ideally, the local carrier must be in frequency and phase synchronism with the incoming carrier.

- Any discrepancy in the frequency or phase of the local carrier gives rise to distortion in the detector output.

- Consider a DSB-SC case where a received signal is $m(t)\cos w_c t$ and the local carrier is in error. The product of the received signal and the local carrier is e(t), given by:

$$
\begin{aligned}
e(t) &= 2m(t)\cos\omega_c t\cos[(\omega_c + \Delta\omega)t + \delta] \\
&= m(t)\{\cos[(\Delta\omega)t + \delta] + \cos[(2\omega_c + \Delta\omega)t + \delta]\}
\end{aligned}
$$

$$e_o(t) = m(t)\cos[(\Delta\omega)t + \delta]$$

- If there is no frequency or phase error in local carrier then

$$e_o(t) = m(t)$$

Two special cases

Case 1: If Δw = 0 then

$$e_o(t) = m(t)\cos\delta$$

- This output is proportional to m(t) when δ is a constant. The output is maximum when δ = 0 and minimum (zero) when δ=±π/2.
- Thus, the phase error in the local carrier causes the attenuation of the output signal without causing any distortion, as long as δ is constant.
- Unfortunately, the phase error δ may vary randomly with time. This may occur, for example, because of variations in the propagation path.
- This causes the gain factor cosδ at the receiver to vary randomly and is undesirable.

Case 2: If Δδ=0 then

$$e_o(t) = m(t)\cos(\Delta\omega)t$$

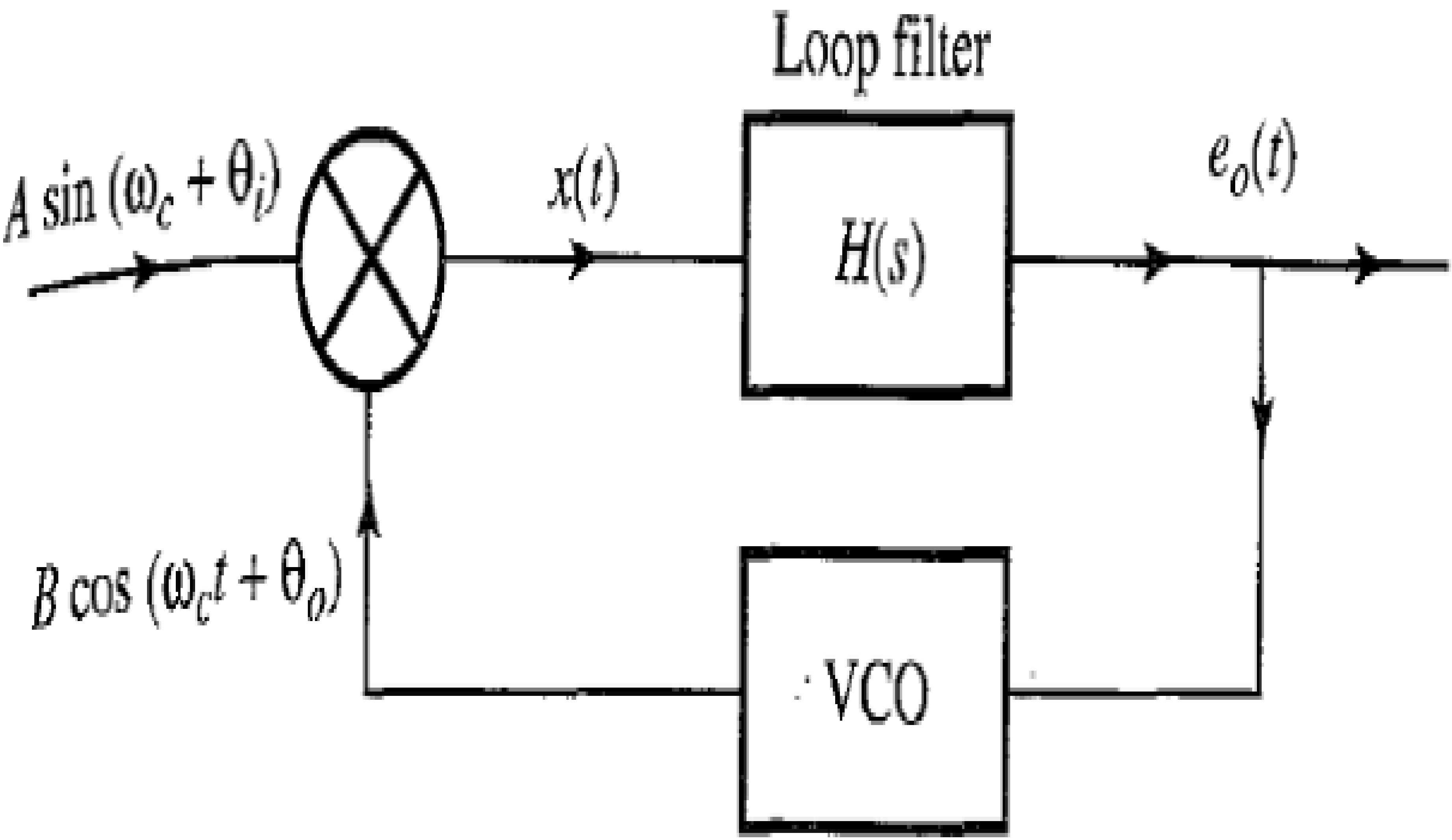
- The output here is not merely an attenuated replica of the original signal but is also distorted.
- Because Δw is usually small, the output is the signal m(t) multiplied by a low-frequency sinusoid.
- This causes the amplitude of the desired signal rn (t) to vary from maximum to zero periodically at twice the period of the beat frequency Δw. This "beating" effect is catastrophic even for a small frequency difference.

Carrier Acquisition Techniques

- To ensure identical carrier frequencies at the transmitter and the receiver, we can use quartz crystal oscillators, which generally are very stable.
- Identical crystals are cut to yield the same frequency at the transmitter and the receiver.
- At very high carrier frequencies, the crystal dimensions become too small to match exactly, quartz-crystal performance may not be adequate.
- In such a case, a carrier or pilot is transmitted at a reduced level (usually about -20 dB) along with the sidebands.
- The pilot is separated at the receiver by a very narrow-band filter tuned to the pilot frequency. It is amplified and used to synchronize the local oscillator.
- The phase-locked loop (PLL), plays an important role in carrier synchronization/acquisition.
- The nature of the distortion caused by asynchronous carrier in SSB-SC is somewhat different than that in DSB-SC.

Phase-Locked Loop (PLL)

- The phase-locked loop (PLL) can be used to track the phase and the frequency of the carrier component of an incoming signal.
- It is, therefore, a useful device for synchronous demodulation of AM signals with suppressed carrier or with a little carrier (the pilot).
- It can also be used for the demodulation of angle-modulated signals, especially under low SNR conditions.
- For this reason, the PLL is used in such applications as space-vehicle-to-earth data links, where there is a premium on transmitter weight, or where the loss along the transmission path is very large.
- PLL is also used in commercial FM receivers.
- A PLL has three basic components:
  - voltage-controlled oscillator (VCO)
  - A multiplier serving as a phase detector (PD) or a phase comparator
  - A loop filter H(s)


Loop filter
$A \sin(\omega_c + \theta_i)$
$x(t)$
$H(s)$
$e_o(t)$
$B \cos(\omega_c t + \theta_o)$
VCO
(a)

Voltage-Controlled Oscillator (VCO)

- An oscillator whose frequency can be controlled by an external voltage is a voltage controlled oscillator (VCO).

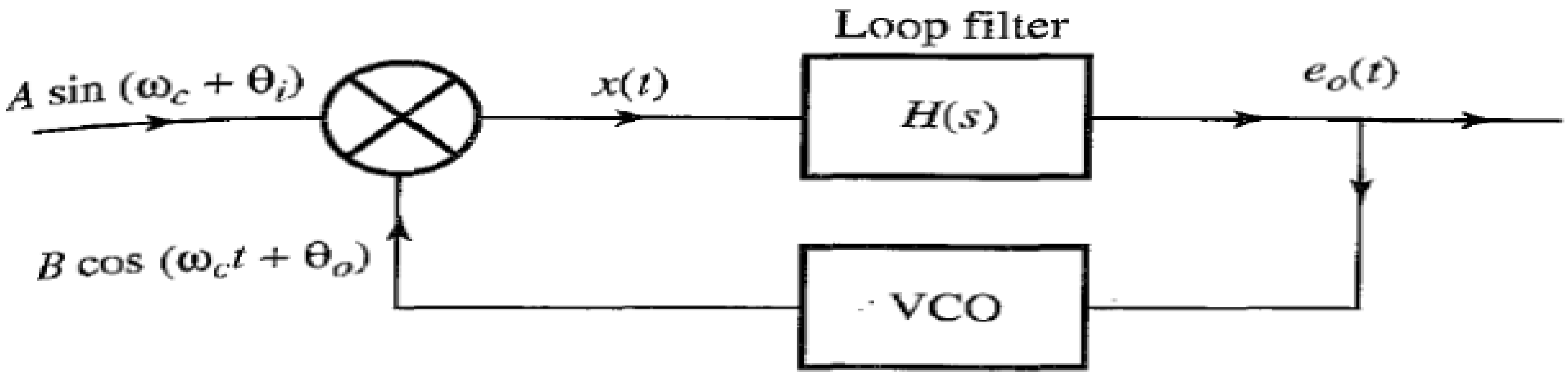
- In a VCO, the oscillation frequency varies linearly with the input voltage.

- If a VCO input voltage is $e_o(t)$, its output is a sinusoid of frequency w given by:

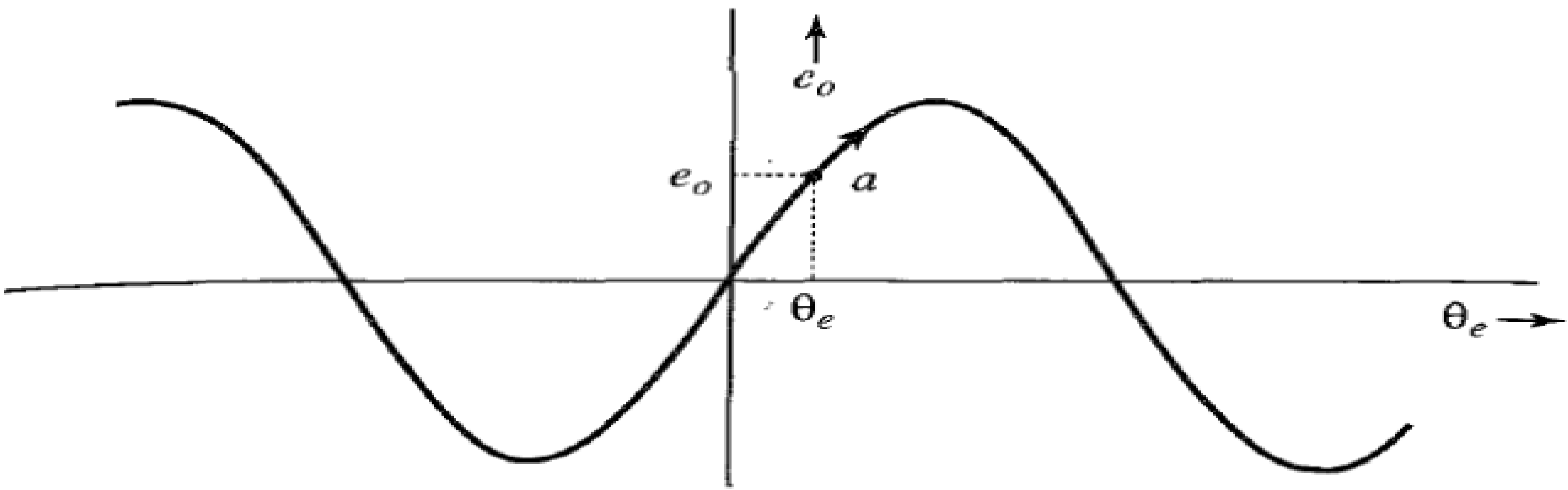
$$\omega(t) = \omega_c + ce_o(t)$$

where c is a constant of the VCO and w is the free-running frequency of the VCO

PLL Working



(a)



(b)

$$x(t) = AB\sin(\omega_c t + \theta_i)\cos(\omega_c t + \theta_o) = \frac{AB}{2}[\sin(\theta_i - \theta_o) + \sin(2\omega_c t + \theta_i + \theta_o)]$$

$$e_o = \frac{AB}{2}\sin\theta_e \qquad \theta_e = \theta_i - \theta_o$$

- The two signals are said to be mutually phase coherent or in phase lock.
- The VCO thus tracks the frequency and the phase of the incoming signal.
- A PLL can track the incoming frequency only over a finite range of frequency shift. This range is called the hold-in or lock range.
- Moreover, if initially the input and output frequencies are not close enough, the loop may not acquire lock.
- The frequency range over which the input will cause the loop to lock is called the pull-in or capture range.
- Also if the input frequency changes too rapidly, the loop may not lock.

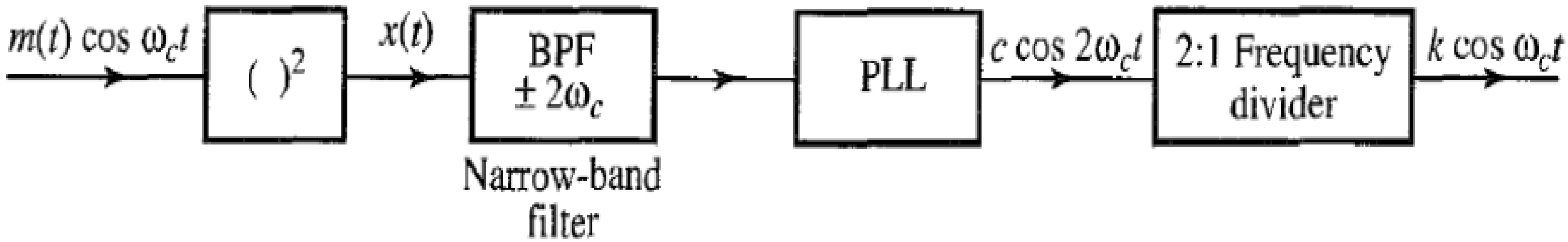
Applications of PLL

- If the input sinusoid is noisy, the PLL not only tracks the sinusoid, but also cleans it up.
- The PLL can also be used as an FM demodulator and frequency synthesizer.
- Frequency multipliers and dividers can also be built using PLL.
- The PLL, being a relatively inexpensive integrated circuit, has become one of the most frequently used communication circuits.
- In space vehicles, because of the Doppler shift and the oscillator drift, the frequency of the received signal has a lot of uncertainty.
- The Doppler shift of the carrier itself could be high whereas the desired modulated signal band may be very low.
- To receive such a signal by conventional receivers would require a filter of bandwidth of twice of Doppler shift
- This would cause an undesirable increase in the noise received because the noise power is proportional to the bandwidth.
- The PLL proves convenient here because it tracks the received frequency continuously, and the filter bandwidth required is same as that of original signal.

Carrier Acquisition in DSB-SC

- Signal squaring Technique
- Costas loop Technique

Signal squaring Technique

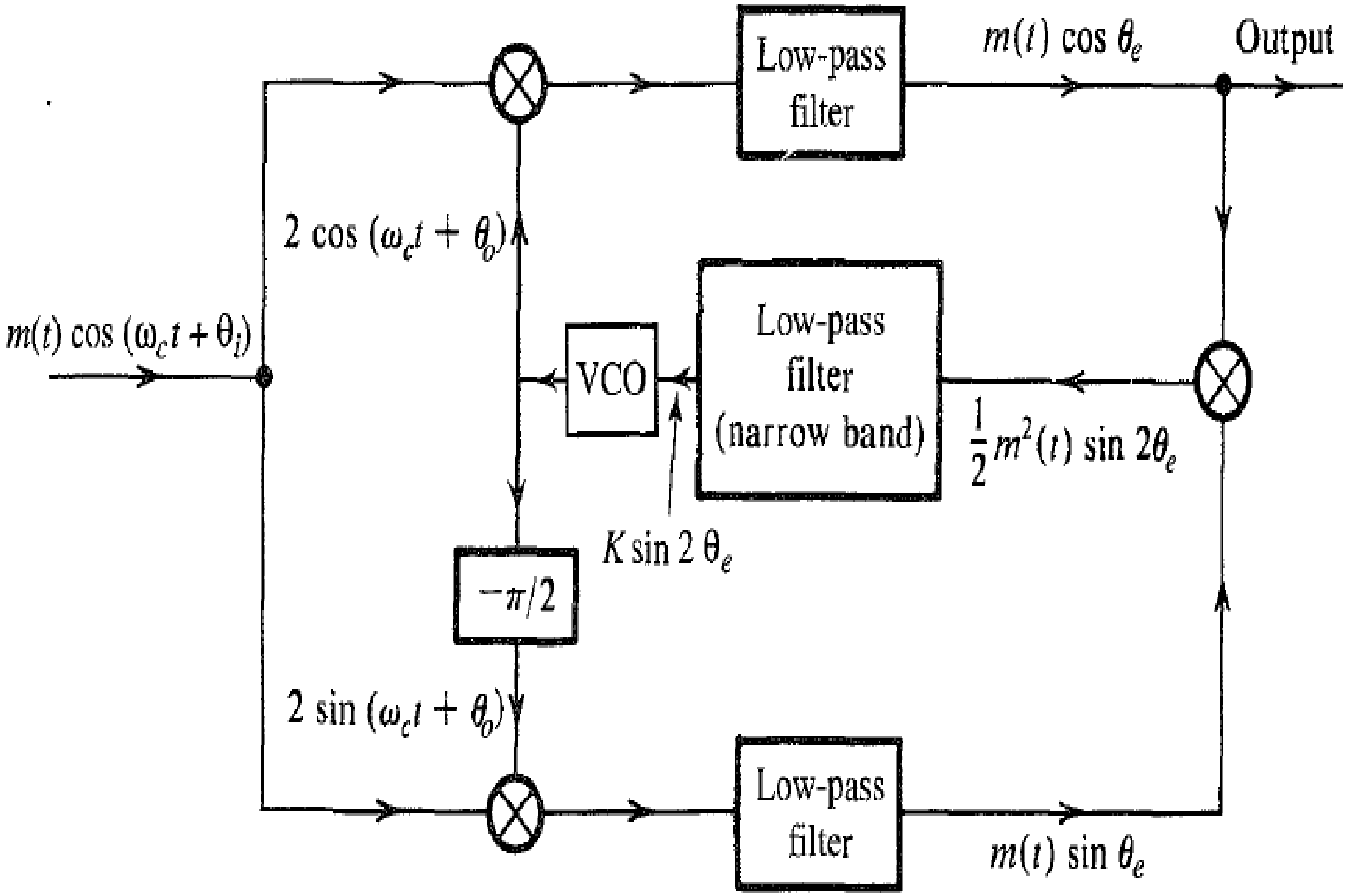


$$x(t) = [m(t)\cos\omega_c t]^2 = \tfrac{1}{2}m^2(t) + \tfrac{1}{2}m^2(t)\cos 2\omega_c t$$

$$\frac{1}{2}m^2(t) = k + \phi(t)$$

$$\begin{aligned} x(t) &= \frac{1}{2}m^2(t) + \frac{1}{2}m^2(t)\cos 2\omega_c t \\ &= \frac{1}{2}m^2(t) + k\ \cos 2\omega_c t + \phi(t)\cos 2\omega_c t \end{aligned}$$

Costas loop Technique



Carrier Acquisition in SSB-SC

- For the purpose of synchronization at the SSB receiver, one may use highly stable crystal oscillators, with crystals cut for the same frequency at the transmitter and the receiver.

- At very high frequencies, where even quartz crystals may have inadequate performance, a pilot carrier may be transmitted.

- However signal squaring technique as well as the Costas loop used in DSB-SC cannot be used for SSB-SC.

$$\begin{aligned}\varphi_{\text{SSB}}(t) &= m(t)\cos\omega_c t \mp m_h(t)\sin\omega_c t \\ &= E(t)\cos[\omega_c t + \theta(t)]\end{aligned}$$

$$\begin{aligned}\varphi_{\text{SSB}}^2(t) &= E^2(t)\cos^2[\omega_c t + \theta(t)] \\ &= \frac{E^2(t)}{2}\{1 + \cos[2\omega_c t + 2\theta(t)]\}\end{aligned}$$

- There is nothing we can do to remove the time-varying phase 2θ(t) from this sinusoid. Hence for SSB, the squaring technique does not work.

Frequency-Division Multiplexing (FDM)

- Signal multiplexing allows the transmission of several signals on the same channel.

- In FDM, several signals share the band of a channel. Each signal is modulated by a different carrier frequency.

- The various carriers are adequately separated to avoid overlap (or interference) between the spectra of various modulated signals.

- These carriers are referred to as subcarriers. Each signal may use a different kind of modulation (for example, DSB-SC, AM, SSB-SC, VSB-SC, or even FM or PM).

- The modulated-signal spectra may be separated by a small guard band to avoid interference and facilitate signal separation at the receiver.

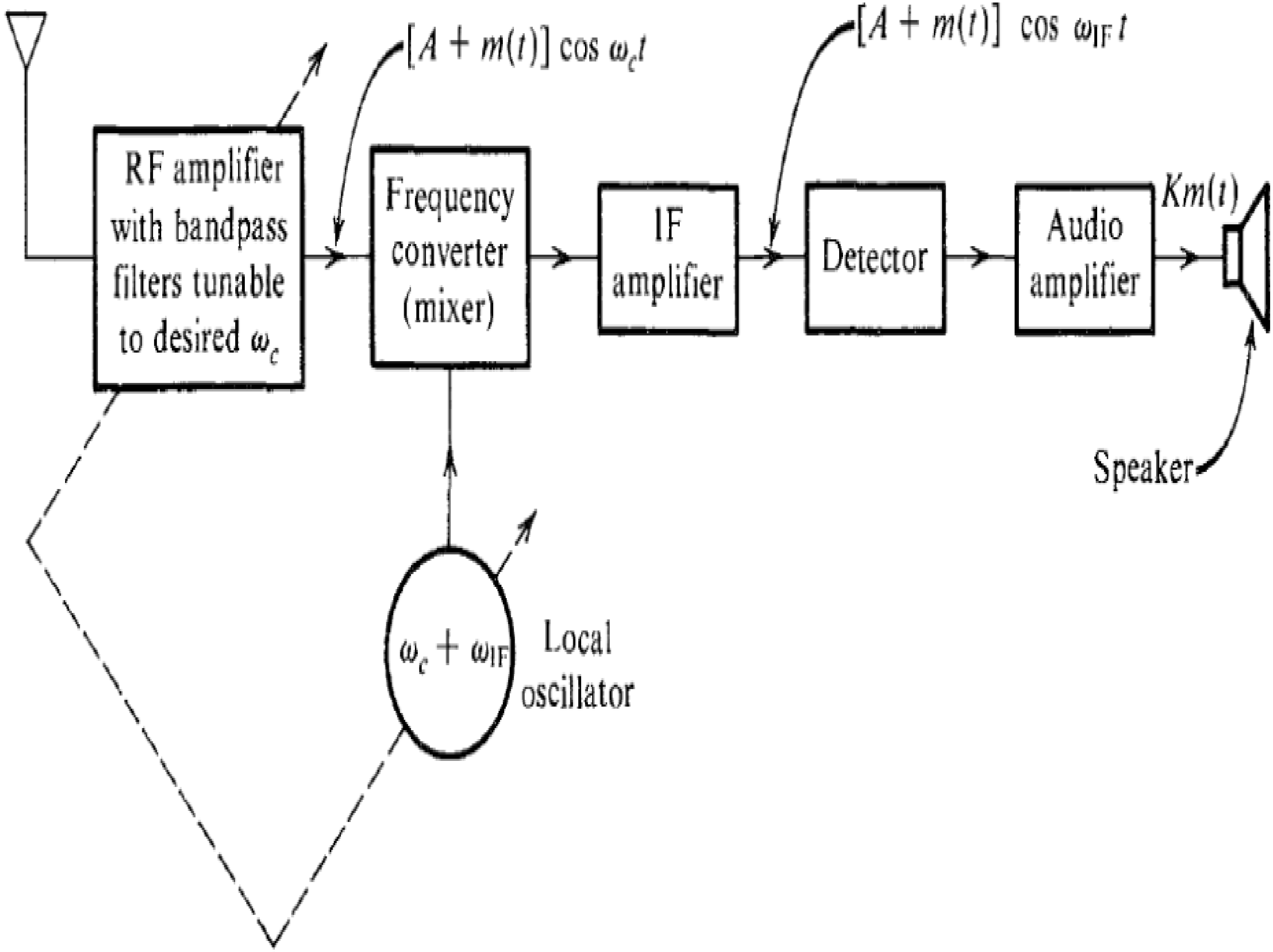
- When all of the modulated spectra are added, we have a composite signal that may be considered as a baseband signal to further modulate a high-frequency [radio frequency (RF)] carrier for the purpose of transmission.

- At the receiver, the incoming signal is first demodulated by the RF carder to retrieve the composite baseband, which is then bandpass filtered to separate each modulated signal.

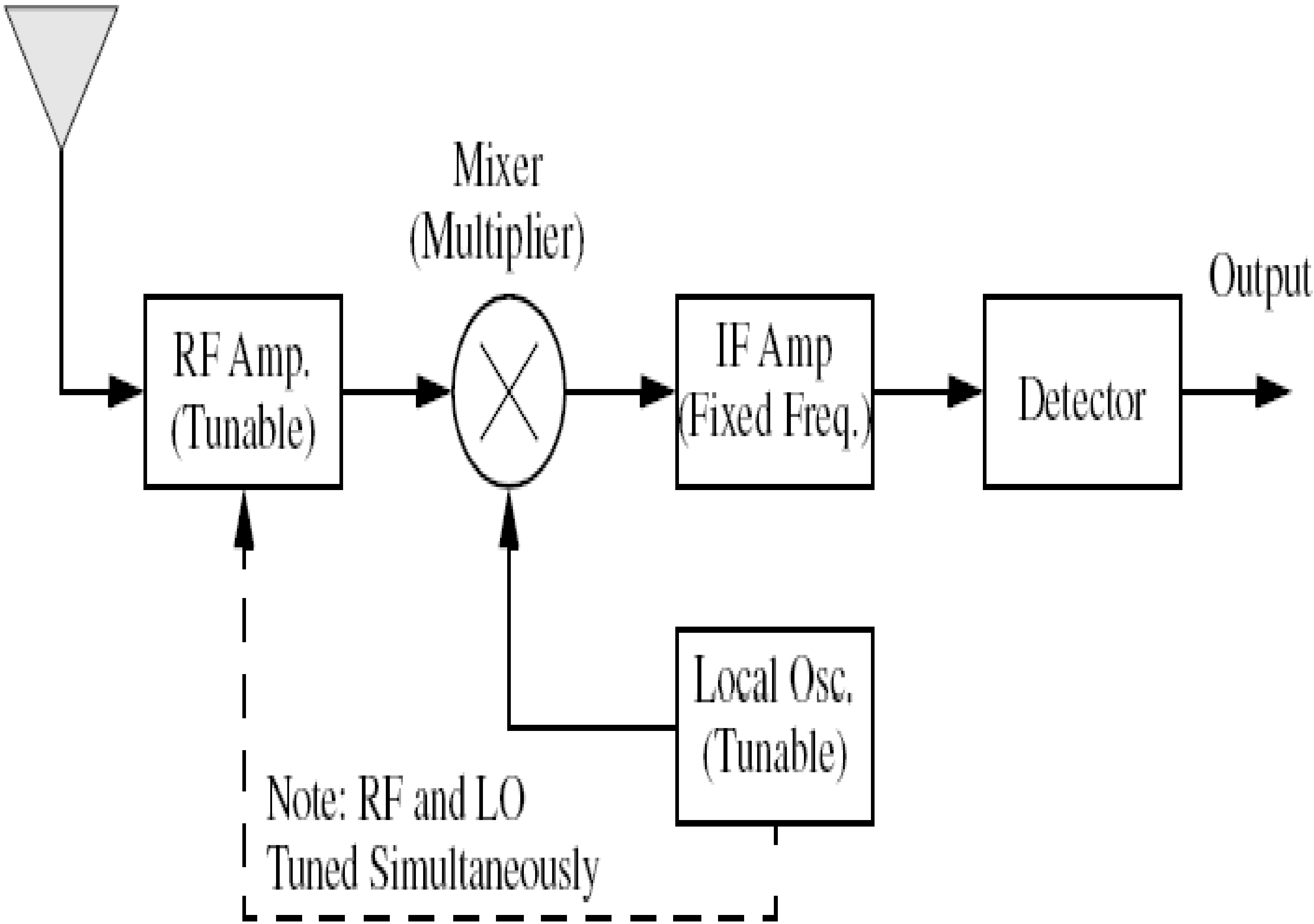
- Then each modulated signal is demodulated individually by an appropriate subcarrier to obtain all the basic baseband signals.

Superheterodyne AM Receiver

- The radio receiver used in an AM system is called the superheterodyne AM receiver.
- It consists of :
  - An RF (radio-frequency) section
  - a frequency converter
  - an intermediate-frequency (IF) amplifier
  - an envelope detector
  - an audio amplifier.
- The RF section is basically a tunable filter and an amplifier that picks up the desired station by tuning the filter to the right frequency band.
- The frequency mixer (converter), translates the carrier from RF to a fixed IF frequency of 455 kHz.
- For this purpose, it uses a local oscillator whose frequency is exactly 455 kHz above the incoming carrier frequency .
- The tuning of the local oscillator and the RF tunable filter is done by one knob.
- Tuning capacitors in both circuits are ganged together and are designed so that the tuning frequency of the LO is always 455 kHz above the tuning frequency of the RF filter.



[A + m(t)] cos ωct
[A + m(t)] cos ωIF t
RF amplifier with bandpass filters tunable to desired ωc
Frequency converter (mixer)
IF amplifier
Detector
Audio amplifier
Km(t)
Speaker
ωc + ωIF
Local oscillator


Mixer
(Multiplier)
Output
RF Amp.
(Tunable)
IF Amp
(Fixed Freq.)
Detector
Local Osc.
(Tunable)
Note: RF and LO
Tuned Simultaneously

- This means every tuned station is translated to a fixed carder frequency of 455 kHz by the frequency converter.
- The reason for translating all stations to a fixed carrier frequency of 455 kHz is to obtain adequate selectivity.
- It is difficult to design sharp bandpass filters of bandwidth 10 kHz (the modulated audio spectrum) if the center frequency fc is very high. This is particularly true if this filter is tunable.
- Hence, the RF filter cannot provide adequate selectivity against adjacent channels.
- But when this signal is translated to an IF frequency by a converter, it is further amplified by an IF amplifier (usually a three-stage amplifier), which does have good selectivity.
- This is because the IF frequency is reasonably low and secondly its center frequency is fixed.
- Hence, the IF section can effectively suppress adjacent-channel interference because of its high selectivity. It also amplifies the signal for envelope detection.
- In reality, practically all of the selectivity is realized in the IF section; the RF section plays a negligible role. The main function of the RF section is image frequency suppression.