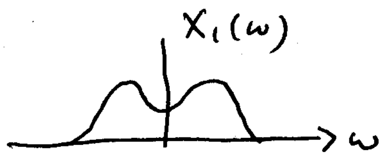
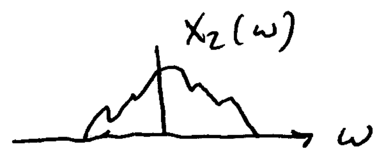
Introduction to Communication Systems

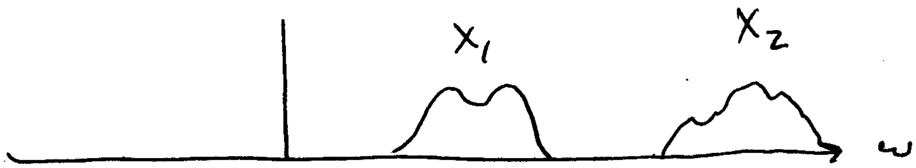
- Communications Engineering refers to all engineering aspects of designing and analyzing systems for communicating information electronically.
- "Signals and systems", or signal processing, plays many important roles in communications engineering.
- EX: radio station 1 wants to transmit signal $x_1(t)$ and radio station 2 wants to transmit signal $x_2(t)$.





→ How can both transmit to the public at the same time?

Answer: The signals $x_1(t)$ and $x_2(t)$ are frequency shifted by different amounts for transmission:



- Listeners can now receive the signal they want, but they must undo the frequency shift before they can listen to the information.

- The frequency shift is accomplished by embedding the information signals $x_1(t)$ and $x_2(t)$ into other signals.

- The act of combining two signals like this is called <u>modulation</u>.

- The act of extracting the information signal at the receiver is called <u>demodulation</u>.

- Information bearing electronic signals are transmitted through a medium.

  <u>Examples:</u>
  - Atmosphere (broadcast radio and TV)
  - Coaxial cable (cable TV)
  - Twisted pair copper wire (plane old telephone system – POTS)
  - Fiber optic (digital telephone, internet trunk)
  - ~~Sattelite~~ Satellite Link (telephone, TV).

- The transmission medium is called the <u>channel</u>.

- Generally, the channel is <u>not</u> an all-pass filter.
  - → it distorts the signal.
  - → prior to transmission, the signal is often passed through the <u>inverse system</u> of the channel.
  - → Then the desired signal is received.
  - → This is called <u>channel equalization</u>.

- We will focus on modulation, which is probably the most important signal processing in communications engineering.

Amplitude Modulation (AM)

- $x(t)$ is the information signal, also called the modulating signal.

- $x(t)$ is embedded in another signal $c(t)$, called the carrier signal.

- $c(t)$ is a complex exponential or a real sinusoid.

- The modulated signal $y(t)$ is

$$y(t) = x(t)\,c(t).$$

→ The frequency shifting property of the Fourier transform tells you that $y(t)$ is a frequency shifted version of $x(t)$.

→ This is one way of accomplishing the frequency shifting discussed on page 8.1.

→ This way of doing it is called amplitude modulation, or AM.

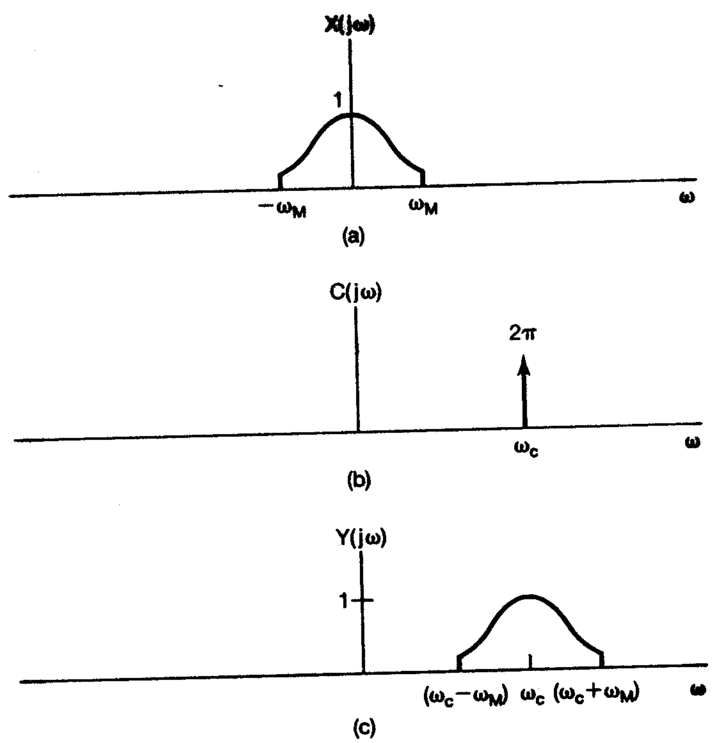
- For a complex carrier, $c(t) = e^{j(\omega_c t + \theta_c)}$
- For a real carrier, $c(t) = \cos(\omega_c t + \theta_c)$

$\rightarrow$ ω_c is called the <u>carrier frequency</u>.

- For simplicity, suppose the carrier phase θ_c is zero.
- Then $y(t) = x(t)e^{j\omega_c t}$ for a complex carrier.
- Then $Y(\omega) = \frac{1}{2\pi} X(\omega) * C(\omega)$.

$\rightarrow$ but $C(\omega) = 2\pi\delta(\omega - \omega_c)$.

$\rightarrow$ so $Y(\omega) = X(\omega - \omega_c)$ (frequency shift)



- Amplitude modulation shifts the frequency band of the information signal $x(t)$.

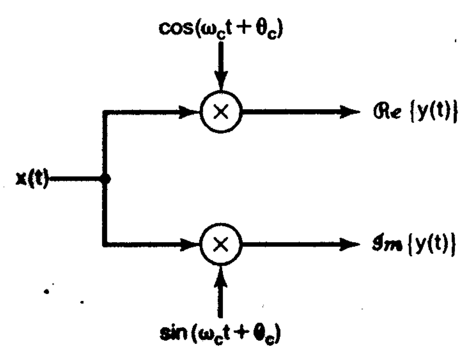
- The information signal $x(t)$ is also called the baseband signal.

- To recover $x(t)$ from $y(t)$ at the receiver, we simply multiply by $e^{-j\omega_c t}$, the conjugate of the carrier signal:

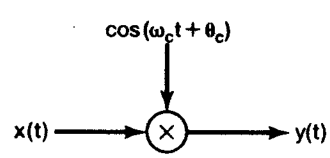
$$x(t) = y(t)\, e^{-j\omega_c t}$$

→ This is called demodulation.

- Block diagram of complex carrier AM transmitter:



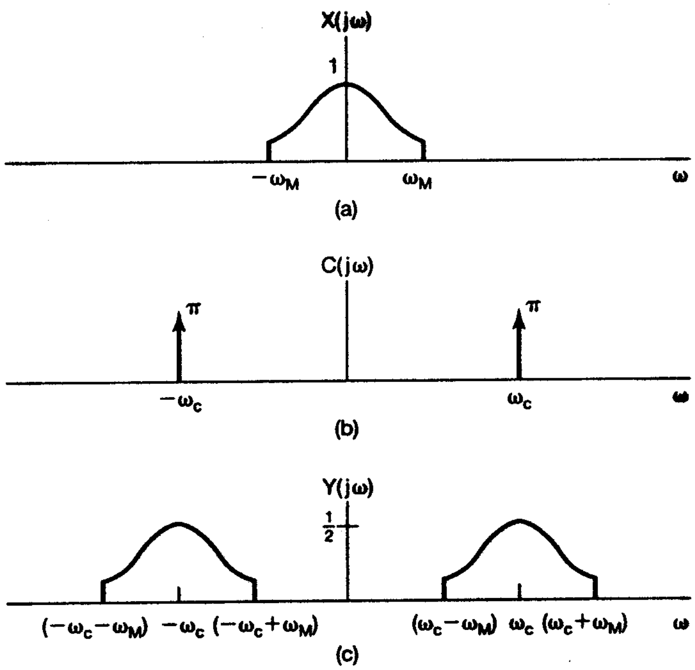
- For a real carrier, the block diagram becomes:



- with a real carrier $c(t) = \cos(\omega_c t + \theta_c)$, we have

$$C(\omega) = \pi[\delta(\omega - \omega_c) + \delta(\omega + \omega_c)].$$

- So, $Y(\omega) = \frac{1}{2}[X(\omega - \omega_c) + X(\omega + \omega_c)]$ $\quad (\theta_c = 0)$



- For demodulation to work, the two copies of $X(\omega)$ cannot overlap.

- This means we must have $\omega_c > \omega_M$, which is almost always the case

EX: AM radio: $\omega_M \sim$ KHz
$\omega_c \sim$ MHz

Synchronous AM Demodulation

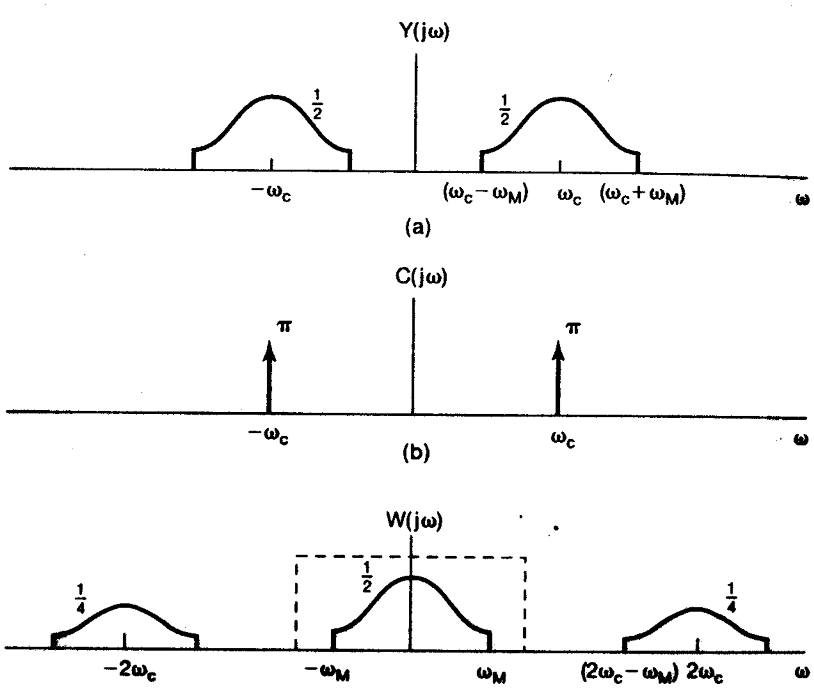
- The received signal is $y(t) = x(t)\cos\omega_c t$ (assuming a lossless channel).
- The first thing the receiver does is multiply $y(t)$ by the carrier signal to get

$$w(t) = y(t)\cos\omega_c t = y(t)c(t)$$

→ $Y(\omega)$ contains two frequency shifted copies of $X(\omega)$.

→ Each of these copies is frequency shifted a second time in $W(\omega)$. Two of them add together to make a new copy of $X(\omega)$.

- As shown below, $X(\omega)$, and thus $x(t)$, can be recovered by applying an ideal low-pass filter **(dashed line)**:



- The filter must have a gain of 2.

- Another way to see this is to realize that

$$w(t) = y(t)\cos\omega_c t$$
$$= x(t)\cos^2\omega_c t$$

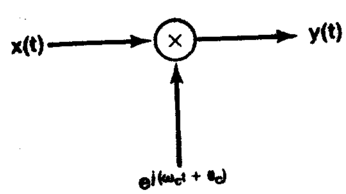
→ use identity $\cos^2\omega_c t = \frac{1}{2} + \frac{1}{2}\cos 2\omega_c t$

$$\Rightarrow w(t) = \frac{1}{2}x(t) + \underbrace{\frac{1}{2}x(t)\cos 2\omega_c t}$$

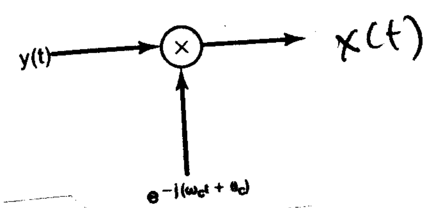
This term is removed by the low-pass filter

- Recall that no low-pass filtering is required when a complex carrier is used, since we can simply multiply $y(t)$ by the conjugate of the carrier.

- Block diagram for complex carrier:

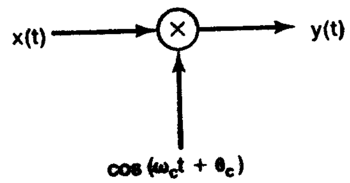


Modulation

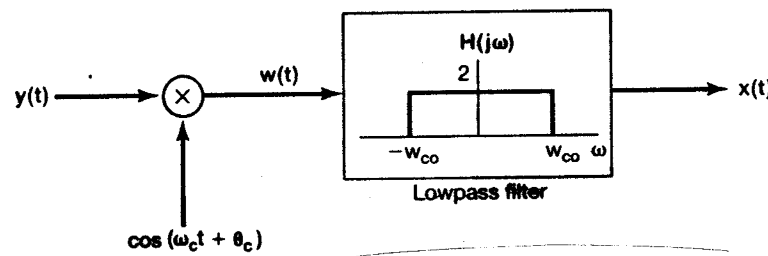


Demodulation

- Block Diagram for real carrier:



Modulation



Demodulation

- In the preceeding block diagrams, note that the transmitter and receiver must <u>both</u> know the carrier signal $c(t) = e^{j\omega_c t}$ or $c(t) = \cos\omega_c t$.
- In particular, the demodulation does not work correctly if there is a phase offset between the carrier signals used in the transmitter and receiver.
- Since The receiver must know the phase of the carrier signal used at the transmitter, this type of demodulation is called <u>synchronous demodulation</u>.
- Suppose now that there is a phase error in the carrier signal used at the receiver (as compared to the one used at the transmitter).

- This type of error is called a synchronization error.

- Specifically, for the case of a complex carrier, suppose that

$$c_T(t) = e^{j(w_c t + \theta_c)} \quad \text{at the transmitter}$$

$$c_R(t) = e^{-j(w_c t + \phi_c)} \quad \text{at the receiver.}$$

- we have

$$\begin{aligned} w(t) &= c_R(t)\, y(t) \\ &= c_R(t)\left[c_T(t)\, x(t)\right] \\ &= e^{-j(w_c t + \phi_c)}\, e^{j(w_c t + \theta_c)}\, x(t) \\ &= e^{j(\theta_c - \phi_c)}\, x(t). \end{aligned}$$

- The demodulated signal is off by a multiplicative complex factor

- For the case of the real-valued carrier, suppose that

$$c_T(t) = \cos(w_c t + \theta_c)$$

$$c_R(t) = \cos(w_c t + \phi_c).$$

- The demodulated signal is

$$w(t) = x(t)\cos(\omega_c t + \theta_c)\cos(\omega_c t + \phi_c)$$

→ Apply the trig identity

$$\cos(\omega_c t + \theta_c)\cos(\omega_c t + \phi_c) = \frac{1}{2}\cos(\theta_c - \phi_c) + \frac{1}{2}\cos(2\omega_c t + \theta_c + \phi_c)$$

- The demodulated signal is

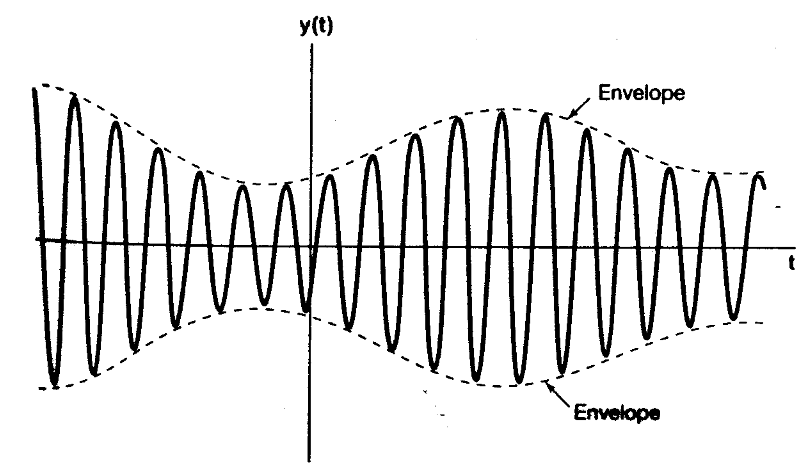
$$w(t) = \frac{1}{2}\cos(\theta_c - \phi_c)\,x(t) + \underbrace{\frac{1}{2}\cos(2\omega_c t + \theta_c + \phi_c)}_{\text{Discarded by low-pass filter.}}$$

→ if $\theta_c = \phi_c$, the output of the low-pass filter is $x(t)$.

→ if $\theta_c - \phi_c = \frac{\pi}{2}$, the output of the low-pass filter is <u>zero</u>.

Asynchronous AM Demodulation

- This approach does not require synchronization between signals at the transmitter and receiver.
  - suppose that $X(\omega)$ is band limited to $-\omega_M \le \omega \le \omega_M$, so that $X(\omega) = 0$ for $|\omega| > \omega_M$.
  - Suppose further that the carrier frequency $\omega_c >> \omega_M$.
  - Finally, suppose that $x(t) \geq 0$ for all t.
  - In this case, $x(t)$ is approximately equal to the envelope of $y(t)$:



  - A demodulator that tracks the envelope of $y(t)$ is called an envelope detector.

- Example of a simple circuit for envelope detection (half-wave rectifier):

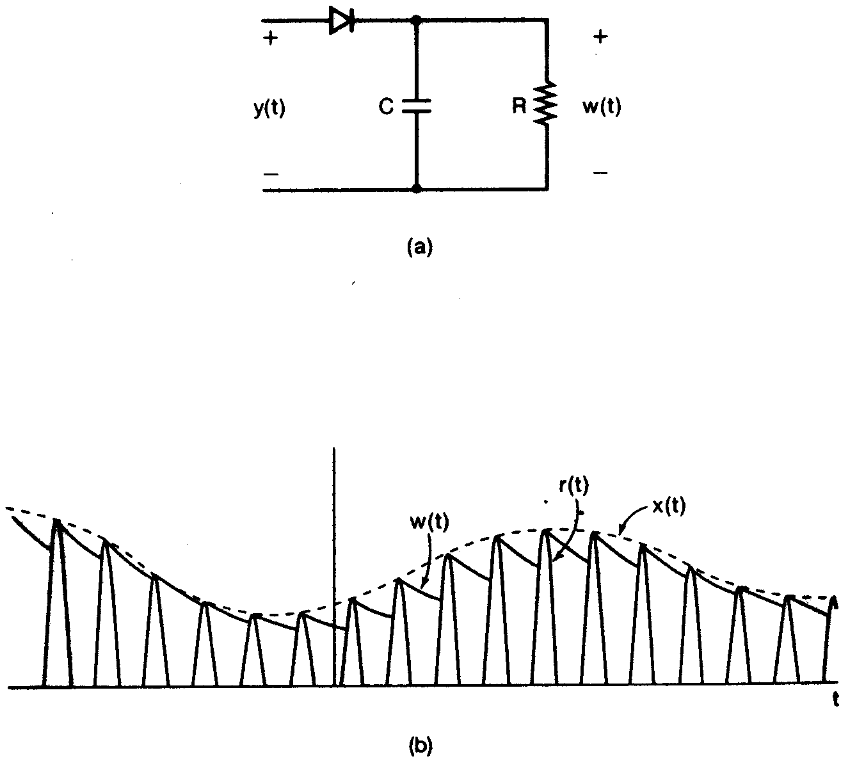


Figure 8.11 Demodulation by envelope detection: (a) circuit for envelope detection using half-wave rectification; (b) waveforms associated with the envelope detector in (a): $r(t)$ is the half-wave rectified signal, $x(t)$ is the true envelope, and $w(t)$ is the envelope obtained from the circuit in (a). The relationship between $x(t)$ and $w(t)$ has been exaggerated in (b) for purposes of illustration. In a practical asynchronous demodulation system, $w(t)$ would typically be a much closer approximation to $x(t)$ than depicted here.

- usually, the output of a demodulation circuit of this type is then processed with a low-pass filter to smooth out the differences between $w(t)$ and $x(t)$.

- The two main requirements for asynchronous demodulation to work are:

    1. $x(t) \geq 0$

    2. $\omega_c \gg \omega_M$

- The second condition is often satisfied automatically.

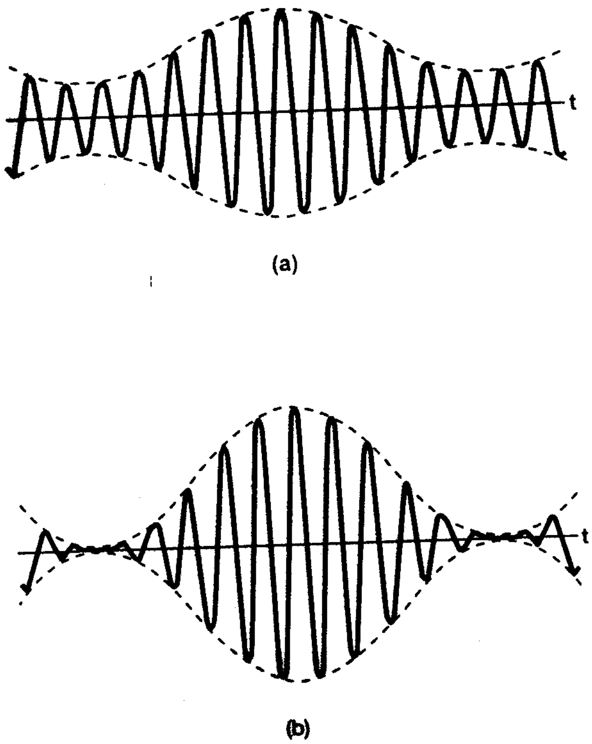
→ For example, in AM radio $\frac{\omega_M}{2\pi} \approx 20$ kHz, while $\frac{\omega_c}{2\pi} \approx 0.5$ MHz to 2 MHz.

- The first condition can always be satisfied by adding a constant A to $x(t)$. If A is picked large enough, then $x(t)+A \geq 0$.
    - Let K = maximum amplitude of $x(t)$, so that $|x(t)| \leq K$.
    - For $x(t) + A$ to be positive, we need $A > K$.

- The ratio $m = \frac{K}{A}$ is called the "modulation index."

- If m is expressed in percent, it is called the "percent modulation."

- Examples of modulated waveforms with $m = 0.5$ (50% modulation) and $m = 1.0$ (100% modulation):

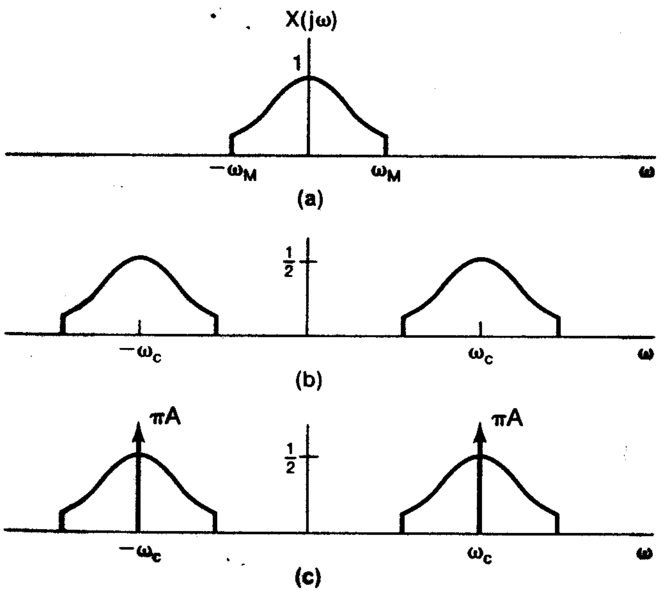


$y(t) = (x(t) + A)\cos\omega_c t$

Figure 8.13 Output of the amplitude modulation system of Figure 8.12: (a) modulation index $m = 0.5$; (b) modulation index $m = 1.0$.

- when synchronous demodulation is used, the transmitted signal is $y(t) = x(t)\cos\omega_c t$
- when asynchronous demodulation is used, the transmitted signal is $y(t) = [x(t) + A]\cos\omega_c t$
- The Fourier transform of $y(t)$ is not the same in these two cases.

- Spectrum $Y(\omega)$ of transmitted signal for the case of synchronous and asynchronous demodulation:



$Y(\omega)$ for synchronous demodulation

$Y(\omega)$ for asynchronous demodulation

- The dirac deltas in the asynchronous demodulation case come from the term $A\cos\omega_c t$ in $y(t) = [x(t)+A]\cos\omega_c t$.

→ Usually, you want to make A small to minimize the power required to transmit the signal.

→ This is equivalent to wanting a large value of the modulation index $m = \frac{K}{A}$.

- But, the smaller the modulation index, the better the simple envelope detector is able to track the envelope and approximate $x(t)$ at the receiver.

- Thus, there is always a tradeoff between transmission efficiency and quality of the demodulated signal when asynchronous demodulation is used.

Frequency Division Multiplexing (FDM)

- In many communications engineering applications, more than one information signal is transmitted through the channel simultaneously.

EX: Cable television : the transmitted and received signals going through the channel (coaxial cable) contain many channels.

EX: Broadcast TV and radio: Each station transmits one modulated signal into the channel (public airwaves). The received signal is the sum of these.

EX: internet or token ring data network:

many different data signals are being transmitted simultaneously through the network.

EX: T1 telephone trunk:

- A T1 trunk can hold about 256 ~~64~~ individual telephone calls simultaneously.

- One way of achieving the simultaneous transmission of several information signals through the channel is to divide up the channel bandwidth into frequency bands.

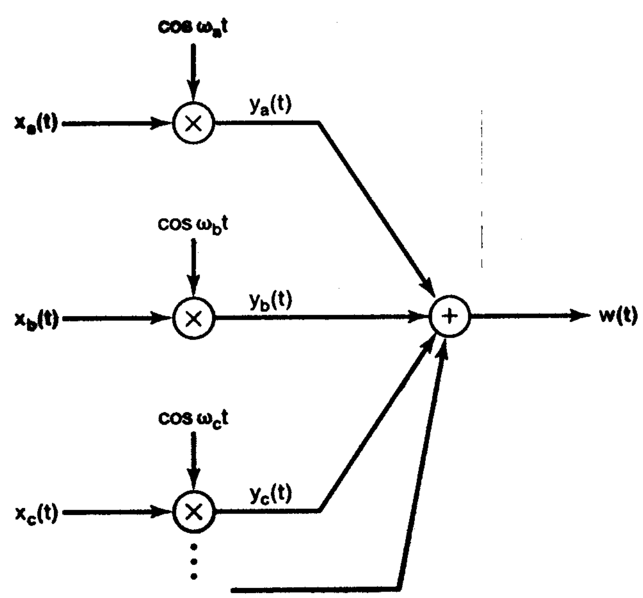
→ Each information signal gets one of the frequency bands.

→ Each information signal uses a carrier signal $c(t) = \cos\omega_0 t$ or $c(t) = e^{j\omega_c t}$ that frequency shifts the information signal $x(t)$ into its assigned frequency band.

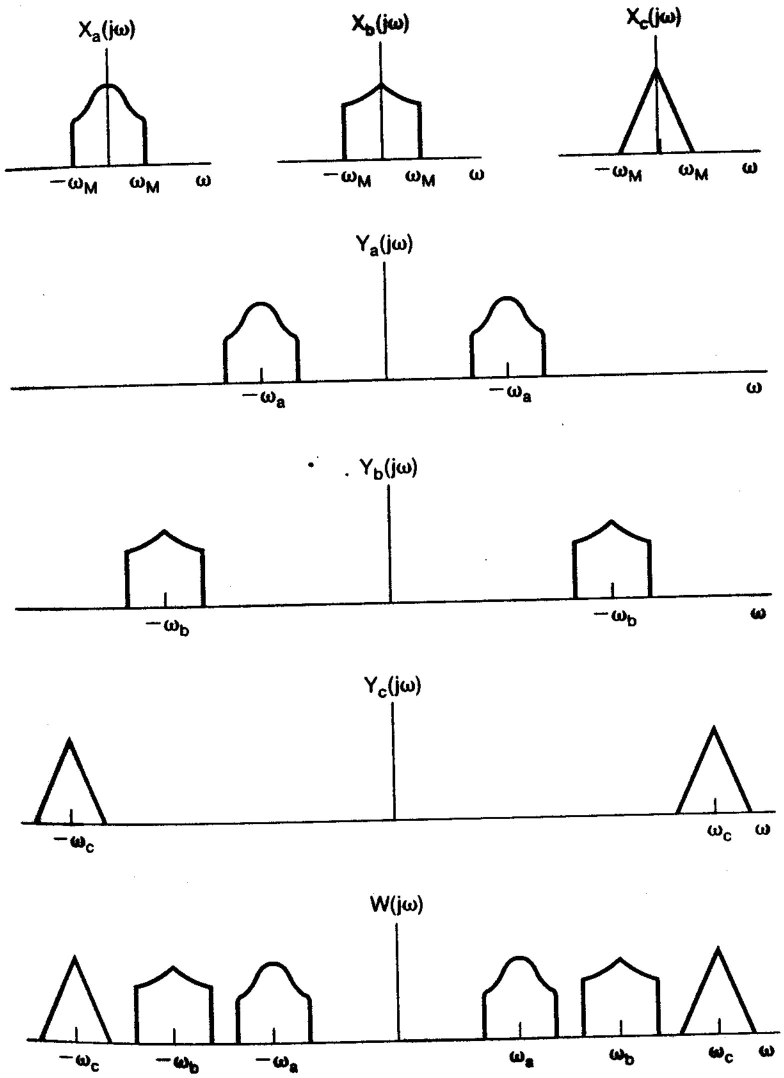
- This is called "frequency division multiplexing", or FDM.

- The carrier frequencies have to be spaced far enough apart so that the modulated signals $y(t)$ have Fourier spectra $Y(\omega)$ that do <u>not</u> overlap one another.

<u>EX</u>:
- The information signals are $x_a(t)$, $x_b(t)$, $x_c(t)$, ...
- The carrier signals are $\cos\omega_a t$, $\cos\omega_b t$, $\cos\omega_c t$, ..., with carrier frequencies ω_a, ω_b, ω_c, ...
- The modulated signals are $y_a(t)$, $y_b(t)$, $y_c(t)$, ...
- The signal that is transmitted through the channel is $w(t) = y_a(t) + y_b(t) + y_c(t) + \dots$



– In the frequency domain:



– To recover any one given information signal, say $x_a(t)$, from $w(t)$ requires two steps...

⟶

- Step 1: Demultiplex $w(t)$ to recover $y_a(t)$.
- Step 2: Demodulate $y_a(t)$ to recover $x_a(t)$.

- To Demultiplex, you bandpass filter $w(t)$ to get $y_a(t)$.

- To demodulate, you demodulate the recovered $y_a(t)$ as before.

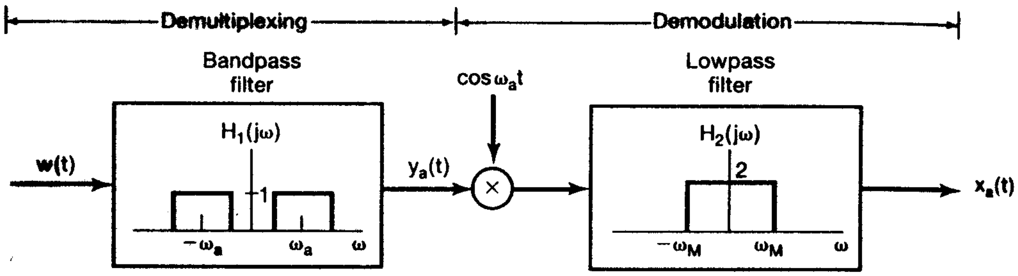


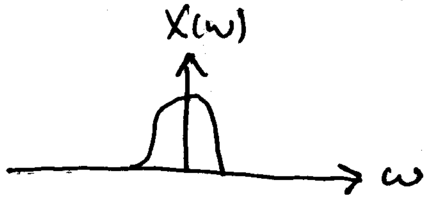
Figure 8.17 Demultiplexing and demodulation for a frequency-division multiplexed signal.

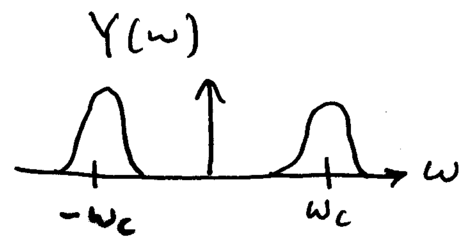
NOTE: in this figure, synchronous demodulation is used.

- For AM radio, the tuning dial on your radio controls the center frequency of the bandpass filter. It also controls the carrier frequency used for demodulation.

NOTE: For broadcast radio, asynchronous demodulation is used to keep the receivers from becoming too expensive.

NOTE: For the AM we have discussed so far, the modulated signal spectrum $Y(\omega)$ contains <u>two</u> copies of the information signal spectrum $X(\omega)$:



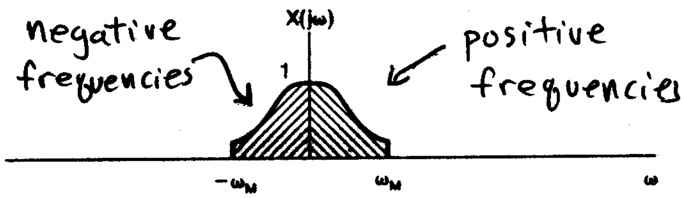


- This scheme is called "double sideband AM".
- Since both copies must be transmitted, the modulated signal $y(t)$ needs twice as much bandwidth in the channel as the original information signal $x(t)$.
- Thus, double sideband AM is inherently bandwidth inneficient.

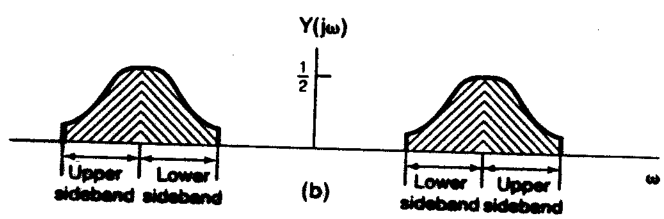
Single-Sideband AM

- With a real carrier $\cos \omega_c t$, we have seen that the modulated signal $y(t)$ occupies twice as much bandwidth as the information signal $x(t)$.
- With a complex carrier $e^{j\omega_c t}$, the modulated signal $y(t)$ occupied the same bandwidth as $x(t)$.
  - → But this requires a complex modulated signal and a complex carrier signal, which is undesirable in many applications.
- Single-Sideband AM is a way to get the best of both worlds:
  - → The carrier signal and modulated signal are real.
  - → The modulated signal occupies the same bandwidth as the information signal $x(t)$.

- The information signal spectrum $X(\omega)$ contains positive frequencies and negative frequencies.



- When we modulate with a real carrier $c(t) = \cos\omega_c t$, $Y(\omega)$ contains two copies of the positive frequencies and two copies of the negative frequencies.
- These copies are called upper and lower <u>sidebands</u>, as shown below:



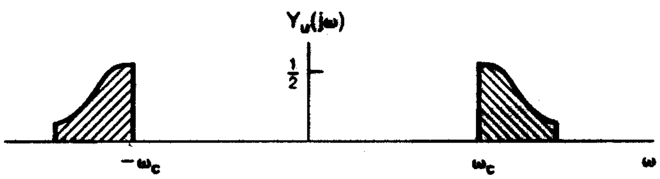
- There are two copies of $X(\omega)$, one centered at ω_c and one centered at $-\omega_c$.
- Each copy of $X(\omega)$ contains a copy of the positive frequencies and a copy of the negative frequencies.

→

- The <u>upper sideband</u> contains the negative frequencies from the copy of $X(\omega)$ centered at $-\omega_c$ and the positive frequencies from the copy of $X(\omega)$ centered at ω_c.

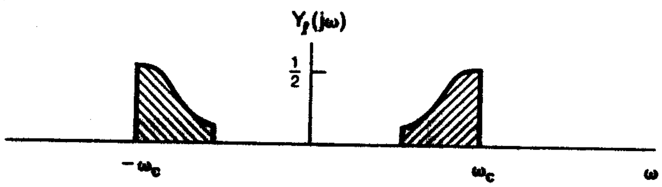
- The <u>lower sideband</u> consists of the negative frequencies from the copy of $X(\omega)$ centered at ω_c and the positive frequencies from the copy of $X(\omega)$ centered at $-\omega_c$.

<u>NOTE</u>: $X(\omega)$ can be recovered from the upper sideband alone:



← The upper sideband

- Similarly, $X(\omega)$ can be recovered from the lower sideband alone:



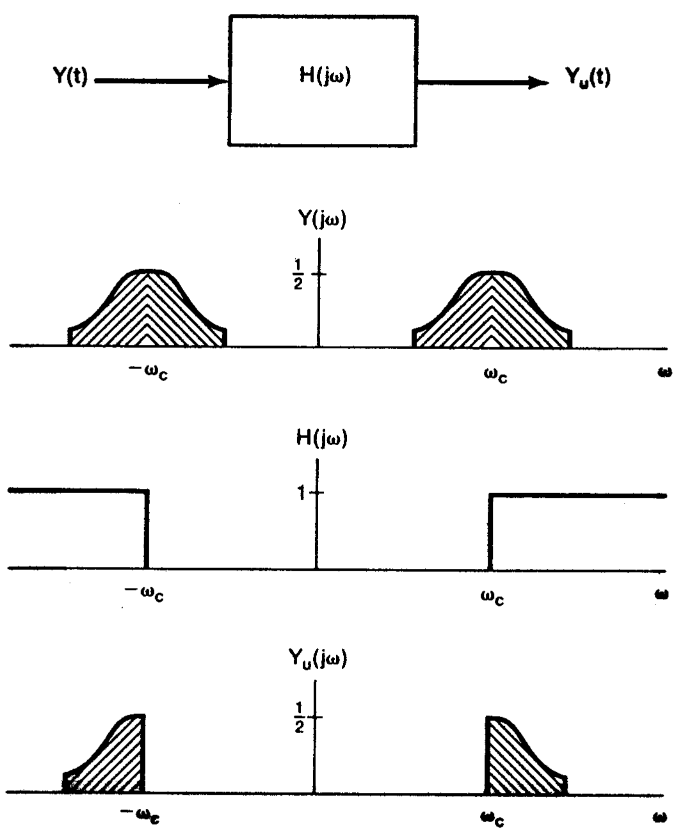
← The lower sideband

- So, by transmitting only <u>one</u> sideband, we can still recover $X(\omega)$,

  → but the required transmission bandwidth is cut in half as compared to double sideband AM.

- One way to get just the upper sideband is to apply a high-pass filter to $y(t)$.

  → The filtered signal is called $y_u(t)$, for "upper sideband."



<u>NOTE</u>: the filter must have a <u>very</u> sharp cutoff frequency.

- Another way to keep just one sideband is to use phase shifting to cancel the unwanted sideband:

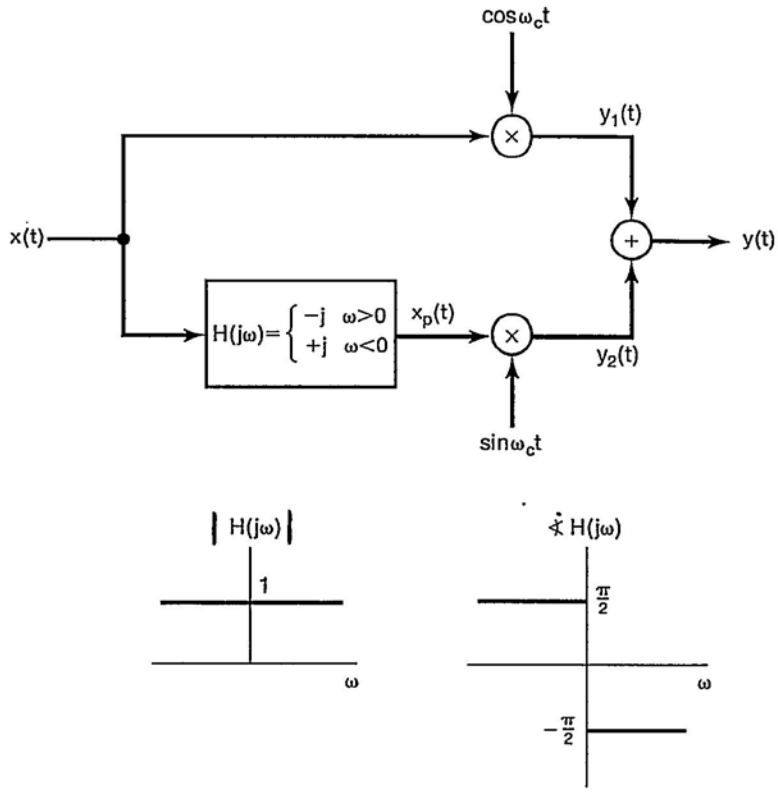


Figure 8.21 System for single-sideband amplitude modulation, using a 90° phase-shift network, in which only the lower sidebands are retained.

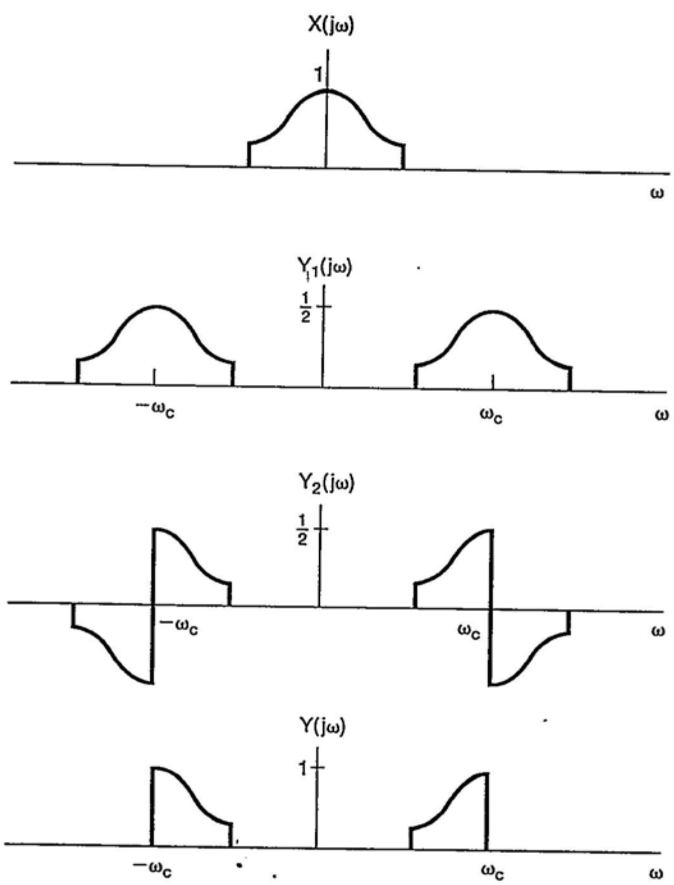
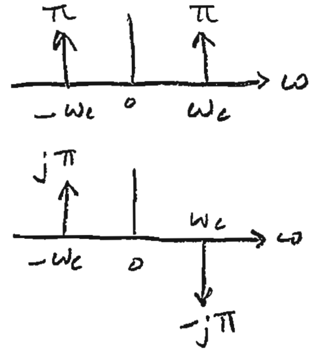


Figure 8.22 Spectra associated with the single-sideband system of Figure 8.21.

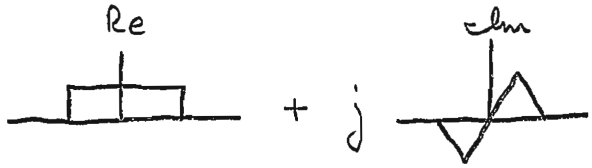
MORE DETAILS ABOUT HOW THE SINGLE-SIDEBAND AM TRANSMITTER ON PAGE 8.27 WORKS:

$$\cos \omega_c t \overset{\mathcal{F}}{\longleftrightarrow} \pi[\delta(\omega-\omega_c)+\delta(\omega+\omega_c)]$$

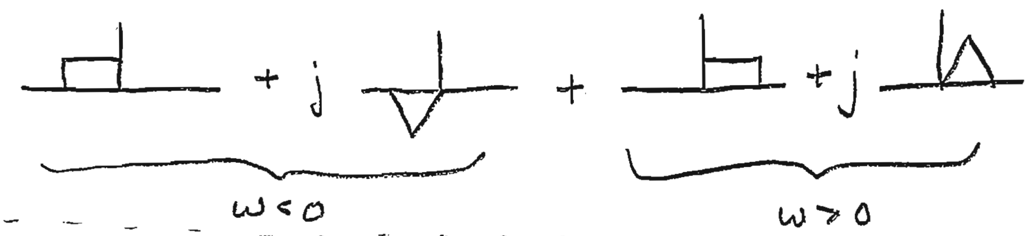
$$\sin \omega_c t \overset{\mathcal{F}}{\longleftrightarrow} -j\pi[\delta(\omega-\omega_c)-\delta(\omega+\omega_c)]$$



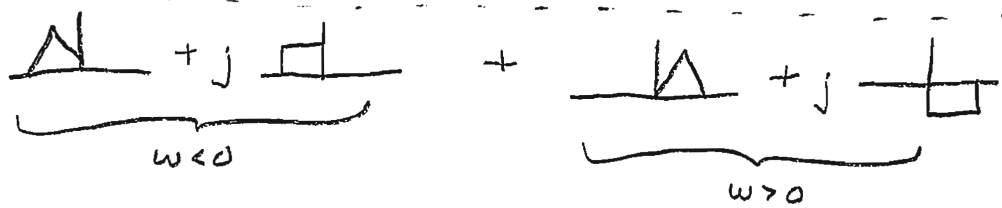
Suppose $X(\omega) =$



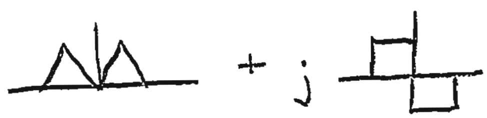
$=$



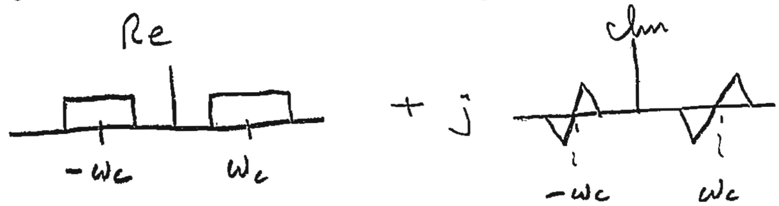
Then $X_p(\omega) =$



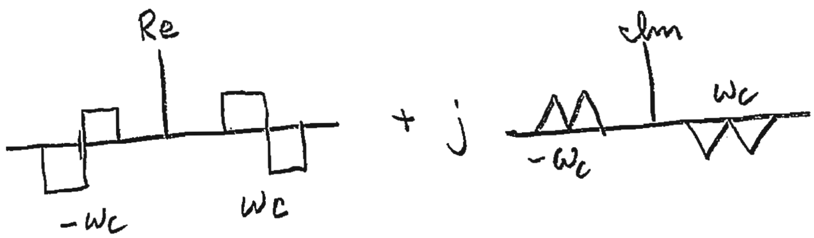
$=$



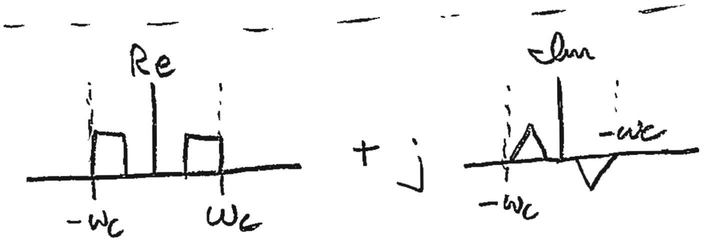
So $Y_1(\omega) =$



and $Y_2(\omega) =$



$Y(\omega) = Y_1(\omega) + Y_2(\omega) =$



$\longrightarrow$ Only the lower sideband is retained.

- To retain the lower sideband, the frequency response of the phase shifting filter is

$$H(\omega) = \begin{cases} -j , & \omega > 0 \\ j , & \omega < 0 \end{cases}$$

- To retain the upper sideband, the filter frequency response is

$$H(\omega) = \begin{cases} j , & \omega > 0 \\ -j , & \omega < 0 \end{cases}$$

- Some acronyms:

AM-DSB/WC : double-sideband AM with carrier for asynchronous demodulation.

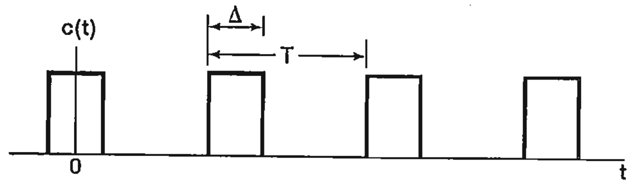
AM-DSB/SC : double sideband AM without carrier (supressed carrier) for synchronous demodulation.

AM-SSB/WC : single-sideband AM with carrier for asynchronous demodulation.

AM-SSB/SC : single-sideband AM with supressed carrier for synchronous demodulation.

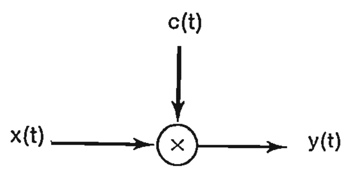
AM with a Pulse Train Carrier

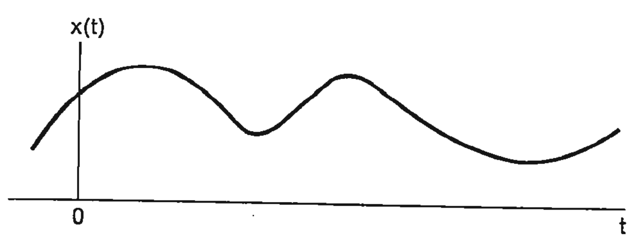
- In this case, the carrier signal $c(t)$ is not sinusoidal, but is instead a sequence of equally spaced rectangular pulses of equal height and equal width:

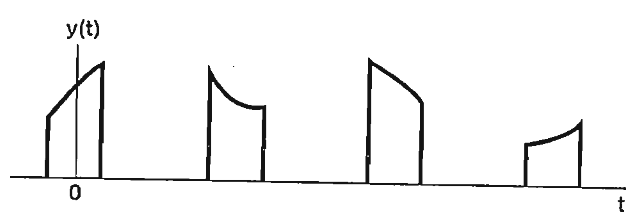


- Pulse height $= 1$
- Pulse width $= \Delta$
- Pulse spacing $= T$

→ The information signal is $x(t)$.
→ The transmitted signal is $y(t) = x(t)c(t)$







- Fourier transform time multiplication property:

$$Y(\omega) = \frac{1}{2\pi} X(\omega) * C(\omega).$$

- Let's find $C(\omega)$:

  - $c(t)$ is periodic with period T.
  - write $c(t)$ in a Fourier series (like we did in chap. 7):

$$c(t) = \sum_{k=-\infty}^{\infty} a_k e^{jk\omega_c t} \qquad \omega_c = \frac{2\pi}{T}$$

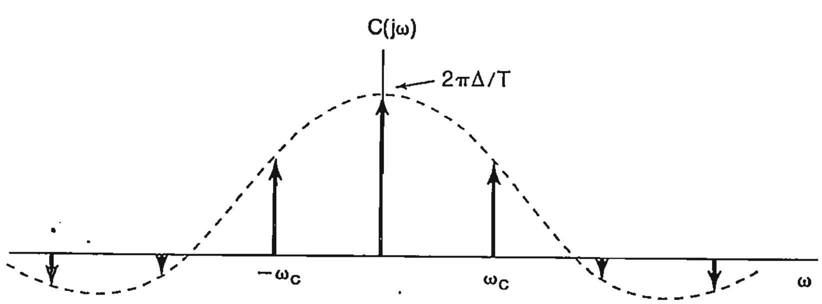
$$a_k = \frac{1}{T}\int_{-T/2}^{T/2} c(t) e^{-jk\omega_c t} dt$$

$$= \frac{1}{T}\int_{-\Delta/2}^{\Delta/2} e^{-jk\omega_c t} dt = \begin{cases} \Delta/T, & k=0 \\ \dfrac{\sin(k\omega_c \Delta/2)}{\pi k}, & k \neq 0 \end{cases}$$

So

$$C(\omega) = \mathcal{F}\{c(t)\} = \mathcal{F}\left\{\sum_{k=-\infty}^{\infty} a_k e^{jk\omega_c t}\right\}$$

$$= 2\pi \sum_{k=-\infty}^{\infty} a_k \delta(\omega - k\omega_c)$$

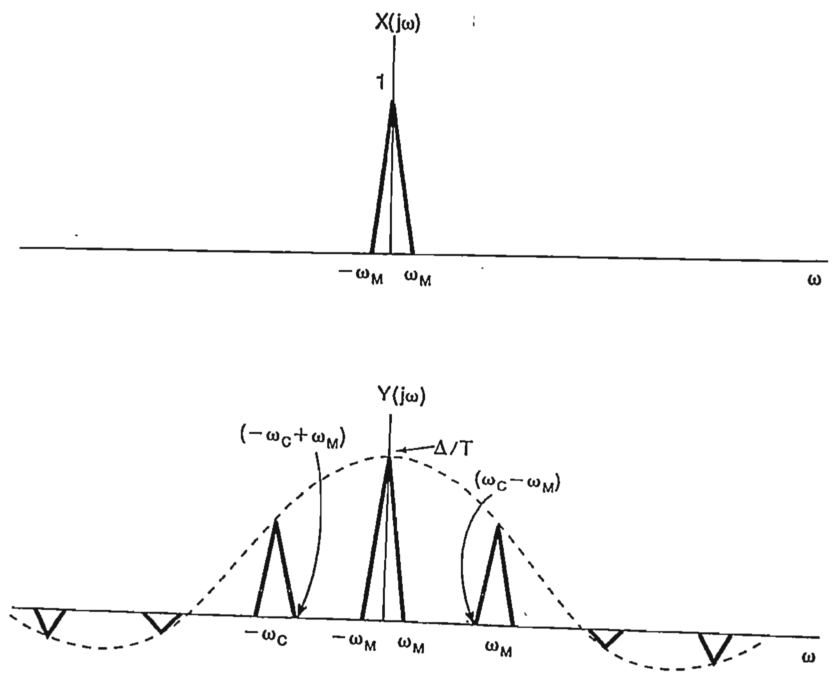


- So the transmitted signal $y(t) = x(t)c(t)$ has a Fourier transform

$$Y(\omega) = \frac{1}{2\pi} X(\omega) * C(\omega)$$

$$= \sum_{k=-\infty}^{\infty} a_k X(\omega - k\omega_c)$$

→ $Y(\omega)$ contains equally spaced copies of $X(\omega)$ that are weighted by the Fourier series coefficients a_k:



NOTE: we are <u>assuming</u> that $x(t)$ is bandlimited to some frequency ω_M.

- As in our discussion of sampling, there will be no aliasing provided that $\omega_c > 2\omega_M$, so that the copies of $X(\omega)$ do not overlap.

  → As long as this is the case, $x(t)$ can be recovered from the modulated signal $y(t)$ by applying a low-pass filter.

  → If there IS aliasing, then $x(t)$ cannot be recovered from $y(t)$.

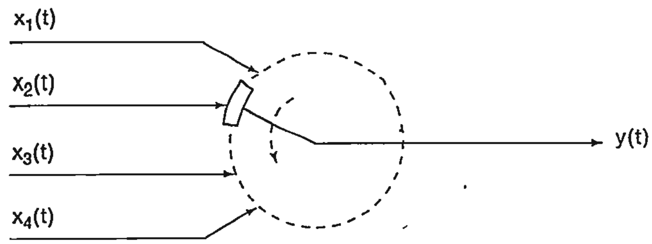
- We have already seen how frequency division multiplexing (FDM) can be used to simultaneously transmit several signals over a single channel.

- With pulse-train AM, we can also accomplish this using Time Division Multiplexing (TDM).

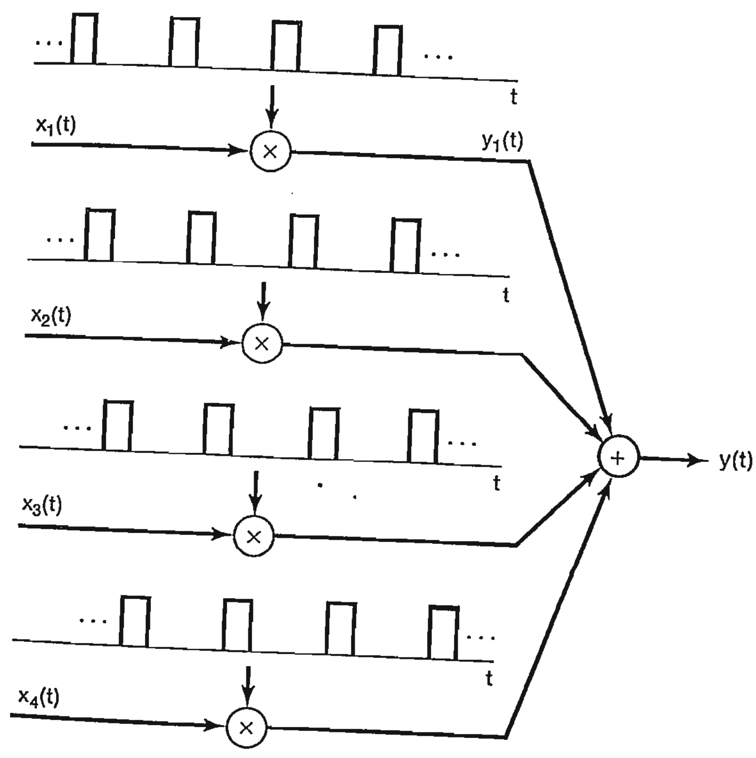
  → With FDM, the channel is divided into frequency "slices", or slots. Each signal gets one slice.

  → With TDM, the channel is divided into time slots. The signals share the time slots in round-robin order.

- TDM can be accomplished with a switch:



- Each information signal $x_i(t)$ is multiplied by a pulse-train carrier
- The pulses for the different carriers do not overlap in time:



Pulse Amplitude Modulation (PAM)

- In pulse-train AM, as in sampling, it is the pulse frequency $\omega_c = \frac{2\pi}{T}$, not the pulse width Δ, that determines if we can recover $x(t)$ from $y(t)$.

- So, as far as demodulation is concerned, using "time slices" of $x(t)$ as we did in pulse-train AM is really no better than just using samples $x(nT)$ of $x(t)$.

- If the carrier pulse frequency $\omega_c = \frac{2\pi}{T}$ is fast enough to recover $x(t)$, we might as well just sample $x(t)$ and set the pulse heights in $y(t)$ equal to the samples $x(nT)$.

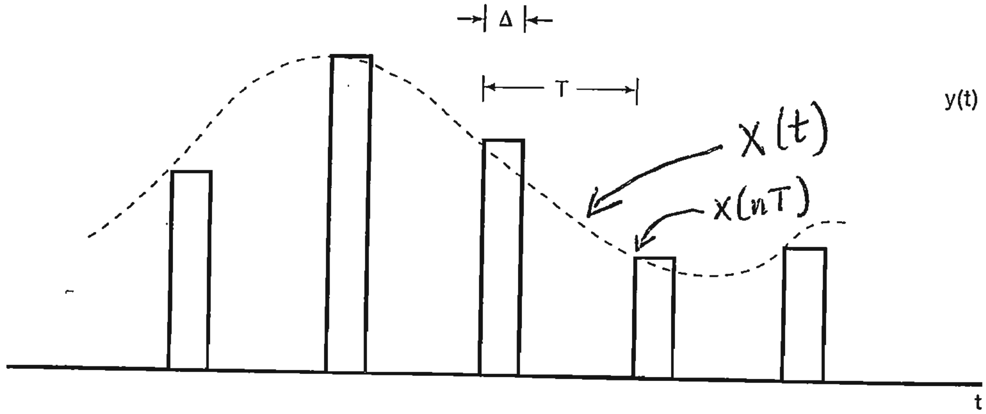
$\Rightarrow$ This is called <u>Pulse Amplitude Modulation</u>, or "PAM".

$\rightarrow$ It is a way to modulate a pulse-train carrier $c(t)$ with a sampled, or "digital" signal $x[n] = x(nT)$:

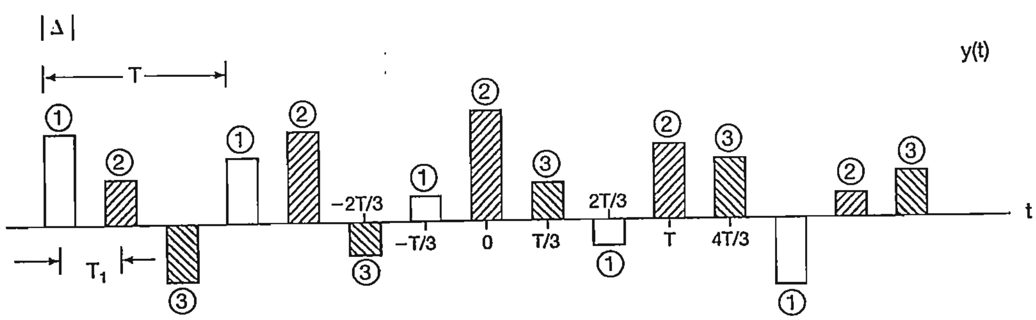
$$y(t) = x[nT]\,c(t).$$

- If the carrier frequency $\omega_c = \frac{2\pi}{T}$ exceeds the Nyquist rate, then we can recover $x(t)$ from $y(t)$.

- PAM for a single information signal $x(t)$:

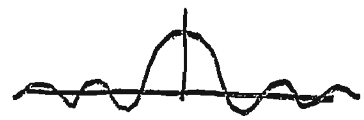


- As with pulse-train AM, we can use time division multiplexing to transmit several PAM signals simultaneously through a single channel:

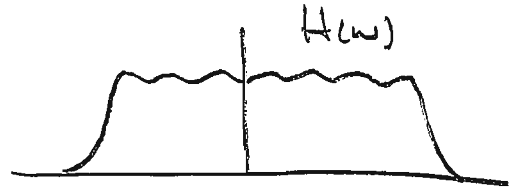


- For this PAM TDM system to work, we need for the received signal to look like the transmitted signal $y(t)$.
- In particular, we need the pulses in the received signal to not overlap, so that we can "undo" the TDM and recover the individual samples of each information signal.
- But is this practical?

- Each pulse in the transmitted signal $y(t)$ is a boxcar.
  - → each one has a Fourier transform that is a sync.
  - → each sync has infinite bandwidth:



- The Fourier transform of $y(t)$ is the sum of these syncs.
  - ⟶ So the transmitted signal $y(t)$ has <u>infinite</u> bandwidth.
- But any real channel has <u>finite</u> bandwidth.
  - So the transmitted signal will get distorted in the channel,
- Suppose that the frequency response of the channel is $H(\omega)$:

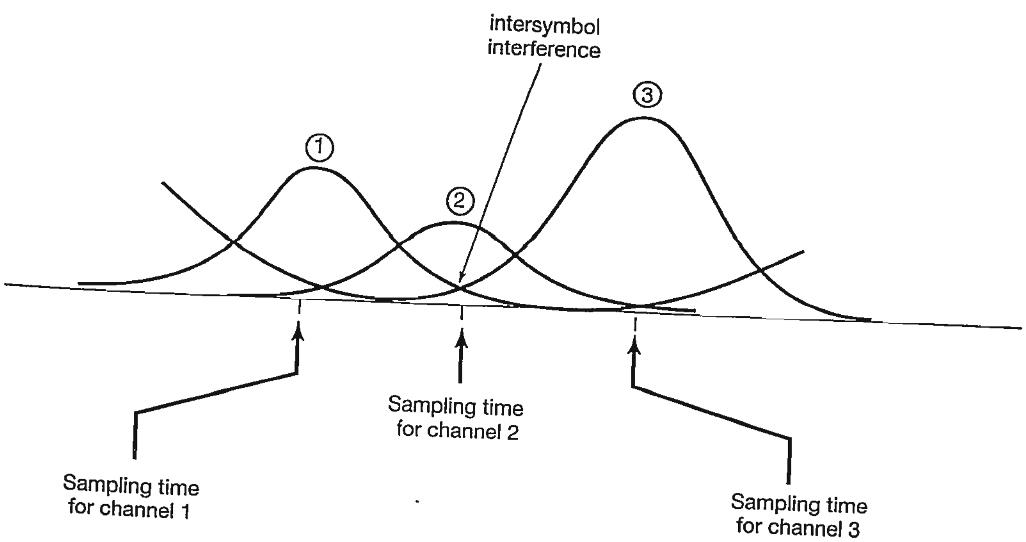


  - For frequencies where $H(\omega) \neq 0$, we can correct the distortion by applying the inverse filter $H^{-1}(\omega) = \frac{1}{H(\omega)}$.
  - This is called "channel equalization."
  - But channel equalization cannot help for the frequencies where $H(\omega) = 0$.
  - So there will always be low-pass distortion when $y(t)$ is transmitted through the channel.
  - If the received signal is $r(t)$, then $R(\omega) = Y(\omega)\,H(\omega)$ and $r(t) \neq y(t)$, even if channel equalization is used.

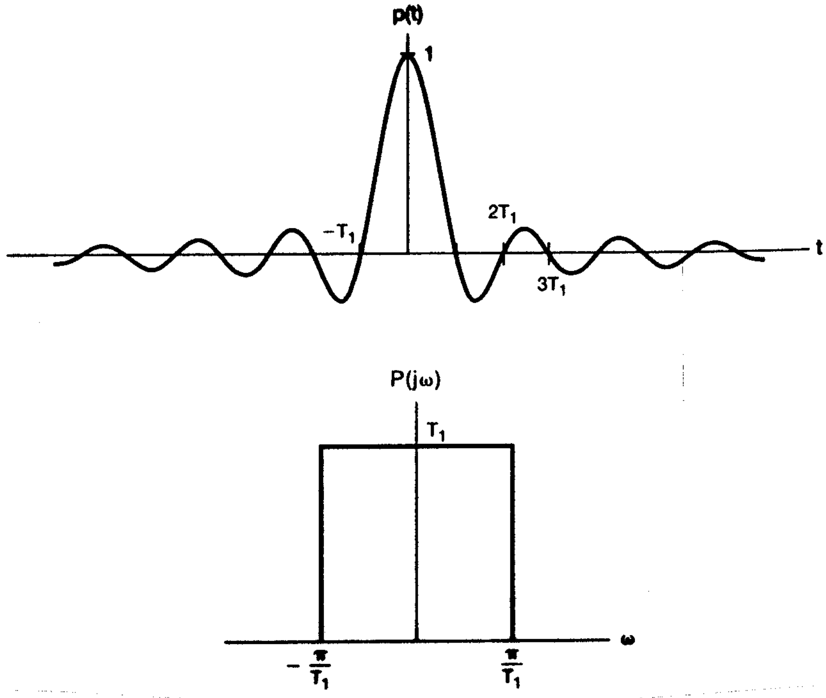
- In practice, the best we can do is to make the channel equalizer the <u>pseudo-inverse</u> of $H(\omega)$:

$$H^{-1}_{pseudo}(\omega) = \begin{cases} \frac{1}{H(\omega)} &, \ H(\omega) \neq 0 \\ 0 &, \ H(\omega) = 0 \end{cases}$$

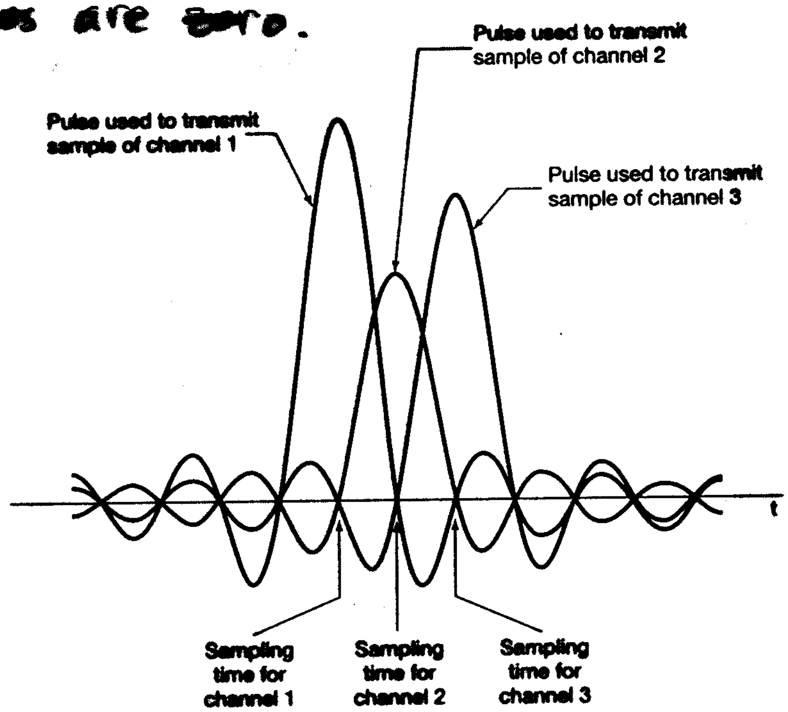
- But some frequencies will always be lost in the received signal $r(t)$.
- This generally has two effects:
  - The sharp corners of the pulses will get rounded off
  - The pulses will get spread out or "smeared" in time.
- When we attempt to demultiplex the received PAM TDM signal, adjacent pulses will generally interfere with each other... degrading the values of the demultiplexed information signal samples.
- This is called <u>intersymbol</u> <u>interference</u>.



- One way to deal with this is to build the carrier waves out of bandlimited pulses:

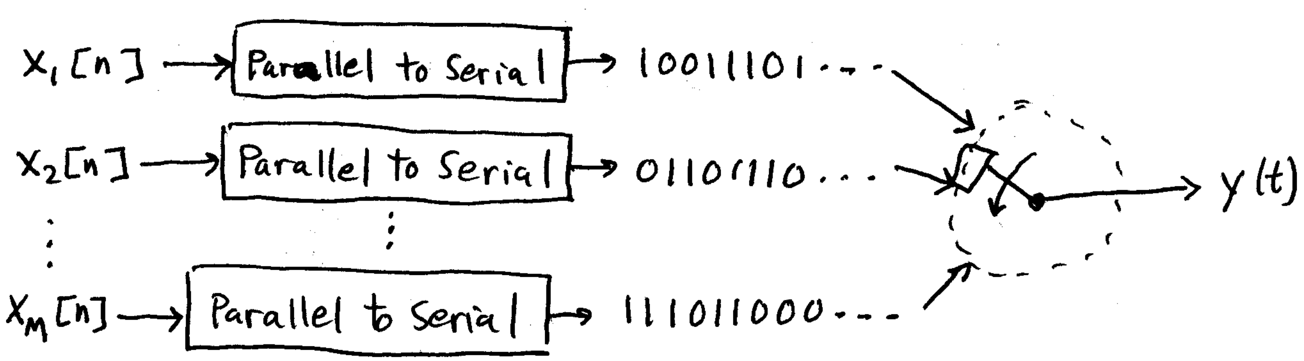


- Then, the pulses will not be degraded by the channel.
- However, we must design the carrier signal c(t) carefully, so that, at the time instant **we sample one pulse**, all the other pul**ses are zero**.



Digital Pulse Code Modulation (PCM)

- In many modern communications systems, the information signals are digital.
  - → We begin with an analog signal $x(t)$.
  - → We sample to get a discrete-time signal $x[n] = x(nT)$.
  - → We quantize to get a digital signal $x[n]$ that only takes integer values in a finite range.
- For example, with 8-bit quantization, each $x[n]$ is an integer in the range $0 \leq x[n] \leq 255$.
- 16 bits is often used for digital speech and telephony.
- If the samples $x[n]$ are converted from parallel to serial, we get a PAM TDM system where all the pulse heights are zero or one:



- This is called <u>Pulse Code Modulation</u>, or "PCM".
- With digital PCM, extra bits can be added to implement parity codes or cyclic redundancy checks (CRCs) for error correction.