



Analogue Communications EE 325

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Course Information

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- Textbook: “Modern Digital and Analog Communication Systems”, B. Lathi & Z Ding, 4th edition, 2007, McGraw-Hill
- Grading: Midterms 60%, Final 40%

Introduction to Communication Systems

- What is a communication system?
- Any means for transmission of information.
- Examples: Telephone, Telegraph, Mobile phone, TV, Radio, Internet, hard disk in a PC, Radar, Satellite, microwave link,...

Elements of a Communication System

- Communication involves the transfer of information from a source to a recipient via a channel or medium.
- Basic block diagram of a communication system:



Brief Description

بعض

- Source: emits analog or digital data.
- Transmitter: transducer, amplifier, modulator, oscillator, power amp., antenna
- Channel: e.g. cable, optical fiber, waveguide, radio link (free space)
- Receiver: antenna, amplifier, demodulator, oscillator, power amplifier, transducer
- Recipient: e.g. person, speaker, computer

Transmitter

محول ربطه

- It may include transducer, amplifier, modulator, oscillator, power amplifier and antenna.
- It modifies the message or the baseband signal for efficient transmission by a process called modulation.
- Other functions: filtering, amplification, radiation

Modulation

Modulation: the process by which the base band signal is used to modify some parameter of a high frequency carrier.

Types of modulation

- **Continuous wave (CW) modulation.**
 - RF sinusoidal carrier wave(30K-300GHz).
- **Pulse modulation.**
 - RF pulse carrier wave.

Why modulation?

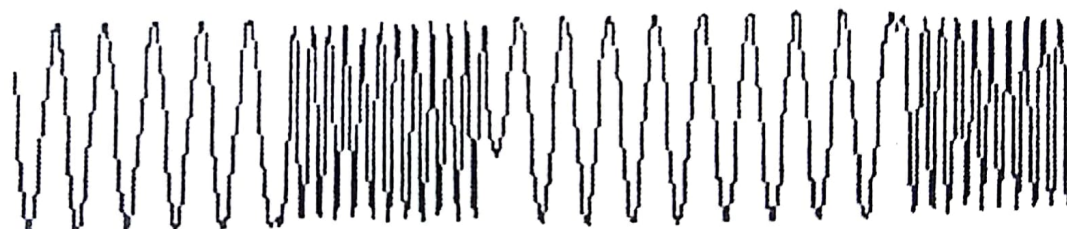
- For ease of radiation.
- Modulation for multiplexing.
- For exchange of SNR with BW.
- To over come equipment limitation.
- To match channel characteristics.

Example of analog modulation

Amplitude



Frequency



or

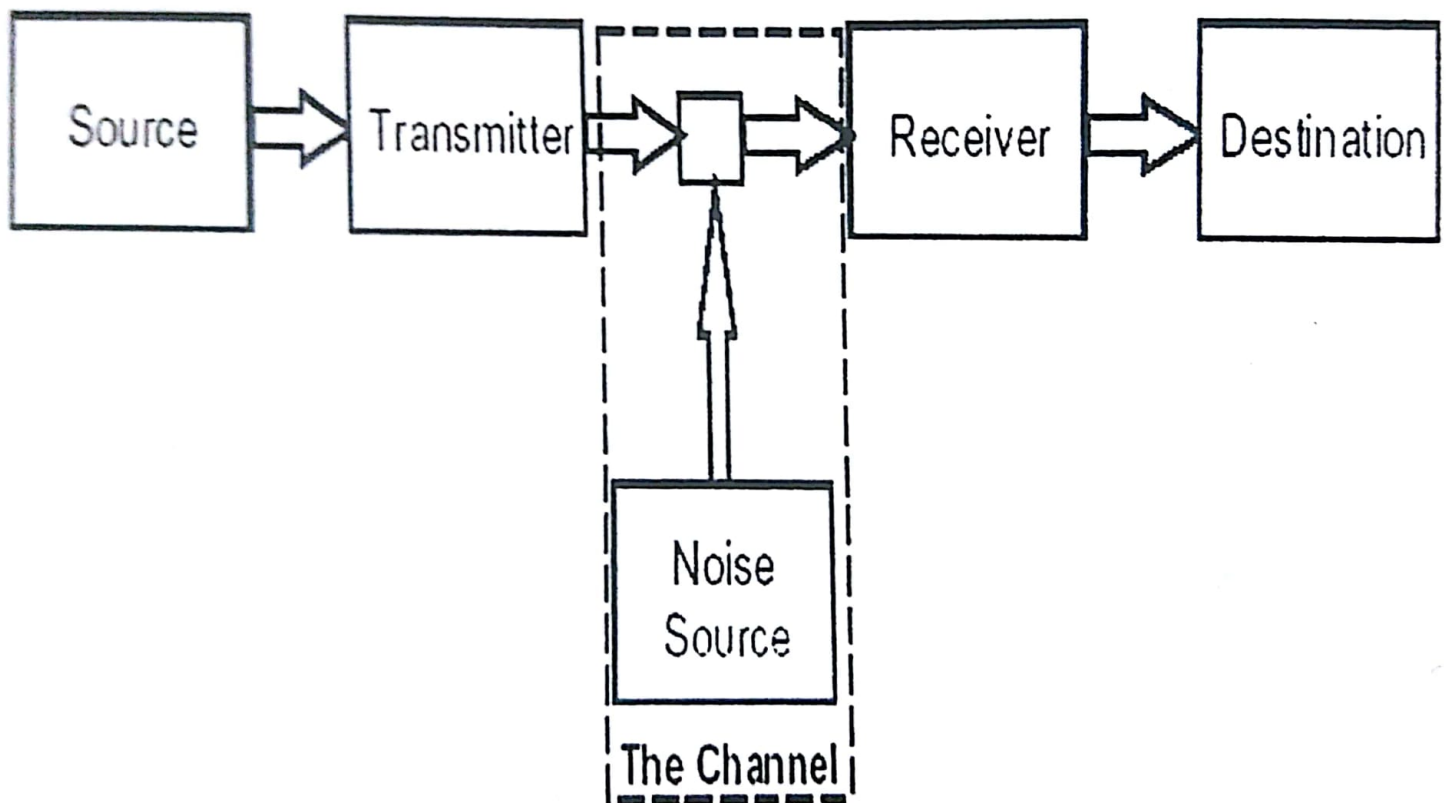
Phase



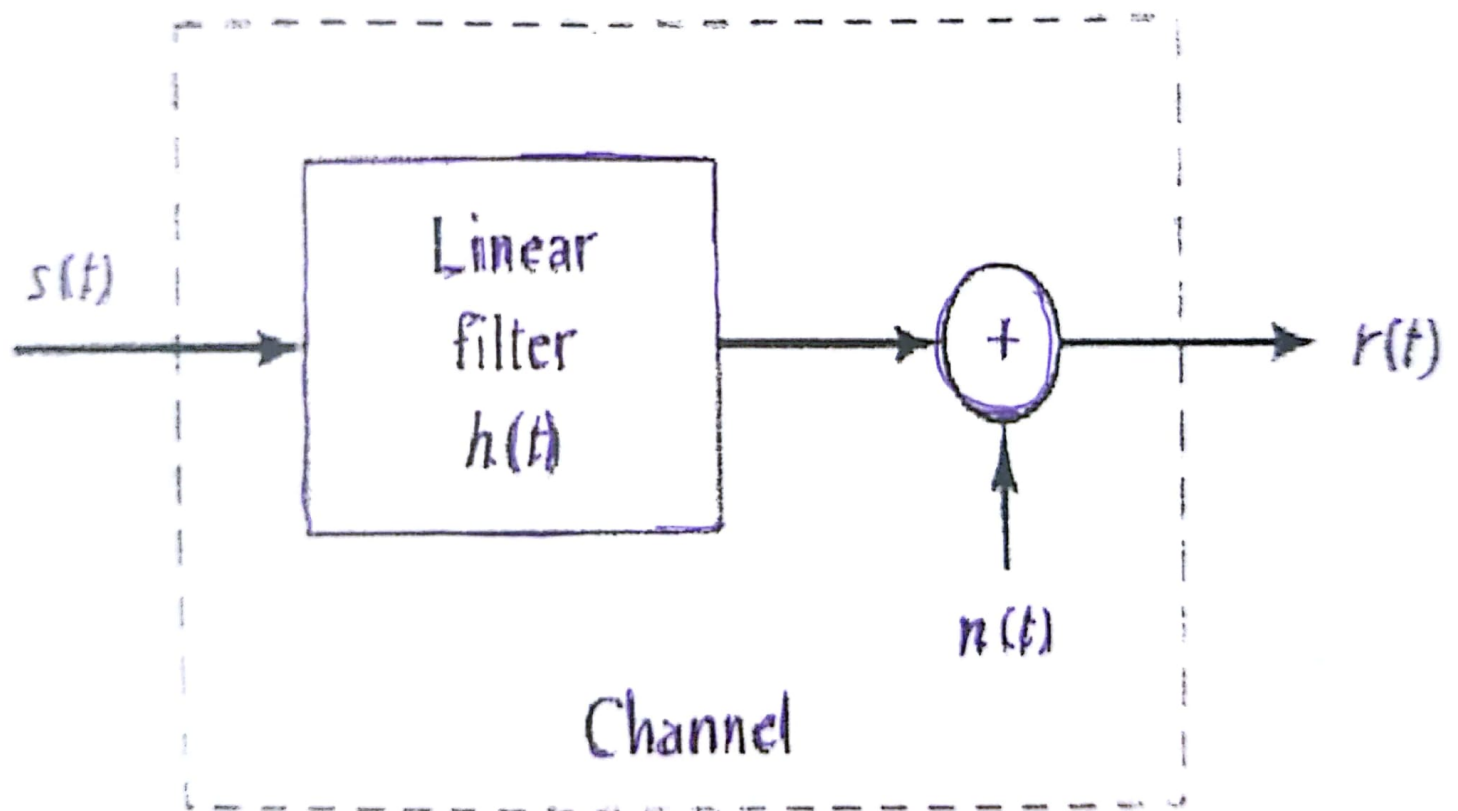
Channel

- It is the physical medium between the transmitter and the receiver. It can be guided, as optical fiber cables, waveguide, or unguided as radio link, water, free space. معطوية، مشرقة
- Whatever the medium, the signal is corrupted in a random manner by noise and interference — (thermal noise, lightning discharge, automobile ignition noise, interference from other users ...)
- Both additive and nonadditive signal distortions are usually characterized as random phenomena and described in statistical terms.

Elements of Communication System



Mathematical Model of Channel



I/O of a comm. channel

$$\begin{aligned} r(t) &= s(t) * h(t) + n(t) \\ &= \int_{-\infty}^{+\infty} h(\tau) s(t - \tau) d\tau + n(t) \end{aligned}$$

Channel Bandwidth

- The bandwidth of a channel is the range of frequencies that it can transmit with reasonable fidelity.
- For example, the bandwidth of
 - twisted pair: several hundred kHz
 - coax cable: several hundred MHz
 - wave guide: few GHz
 - optic fiber: very wide

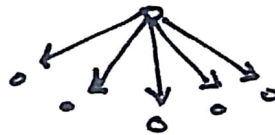
Receiver

- Its main function is to recover the message from the received signal.
- It includes antenna, amplifier, demodulator, oscillator, power amplifier, transducer
- Demodulation: inverse of the modulation
- Operates in the presence of noise & interference. Hence, some distortions are unavoidable.
- Some other functions: filtering, suppression of noise & interference

رسول الله ﷺ

* Types of Communication Systems

- guided & unguided (wireless).
- Digital & analog,
- Point-to-point & broadcasting,



Types of comm. systems

- Analog comm. system
 - Transport analog information using analog modulation techniques (AM, FM, PM).
- Digital comm. system.
 - Transport digital information using digital modulation techniques (ASK, FSK, PSK).
- Hybrid comm. system.
 - Transport digitized analog information using one of the following digital techniques:
 1. Analog pulse modulation schemes (PAM, PDM, PPM).
 2. digital modulation schemes (ASK, FSK, PSK).
 3. Pulse code modulation schemes (PCM, DPCM, δ).

Types of Transmission

- Base-band transmission:

- 1 – Short distance.

- 2 – No modulation is needed. → μ

- Band-pass transmission:

- ① – long distance.

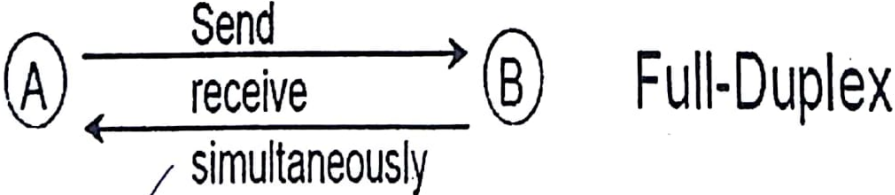
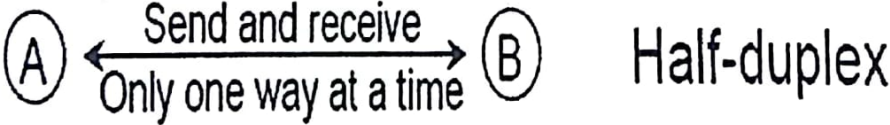
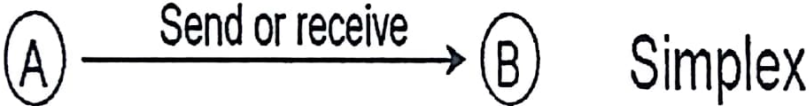
- ② – Modulation is needed. → μ

- ③ – Analog or digital.

Transmission Terminology

- Simplex transmission
 - One direction
 - e.g. Radio and television broadcast.
- Half duplex transmission
 - Either direction, but only one way at a time
 - e.g. police radio(walki-talki)
- Full duplex transmission
 - Both directions at the same time
 - e.g. telephone,

Simplex vs. Duplex



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Analog Transmission

- Analog signal transmitted without regard to their content (May be analog or digital data)
- Attenuated over distance
- Use amplifiers to boost signal
- Also amplifies noise, thus received signal will be distorted.
- If digital data is encoded then amplifiers will increase BER (bit error rate).

↓
تکثیر شد و نویز را هم تقویت کرد.
سیگنال را تقویت کرد.

Digital Transmission

- Concerned with content of the signal.
- Integrity endangered by noise, attenuation etc.
- Repeaters used to achieve greater distance.
- Repeater receives signal
 - Extracts bit pattern
 - Retransmits new signal
 - Attenuation is overcome
 - Noise is not amplified

comp

Transmission Impairments

- Signal received may differ from signal transmitted
- Analog - degradation of signal quality
- Digital - bit errors
- Caused by
 - Attenuation and attenuation distortion
 - Delay distortion
 - Noise

Attenuation

- Signal strength falls off with distance
- Depends on medium
 - guided: attenuation is logarithmic.
 - unguided: attenuation depends on atmospheric structure.
- Received signal strength:
 - must be enough to be detected
 - must be sufficiently higher than noise to be received without error

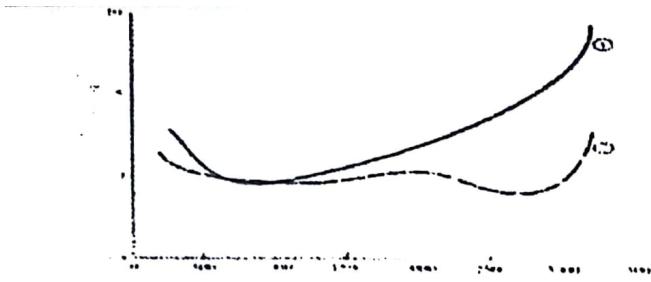
Atten. Cont.

- Attenuation is an increasing function of frequency.
- attenuation distortion affects analog signals much more than digital signals.
- Fading channel.
- Equalizers: reduce attenuation distortion.

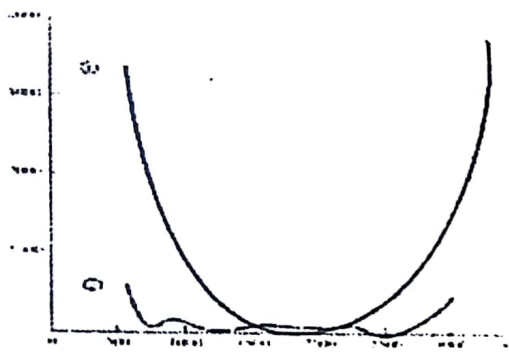
Delay Distortion

in lines
finite length

- Only in guided media
- Caused by: Propagation velocity varies with frequency.
 - different frequency components arrive at the receiver at different times causing phase shifts.
- for digital data delay distortion introduces inter-symbol interference (ISI).
- Equalizers : reduce delay distortion.



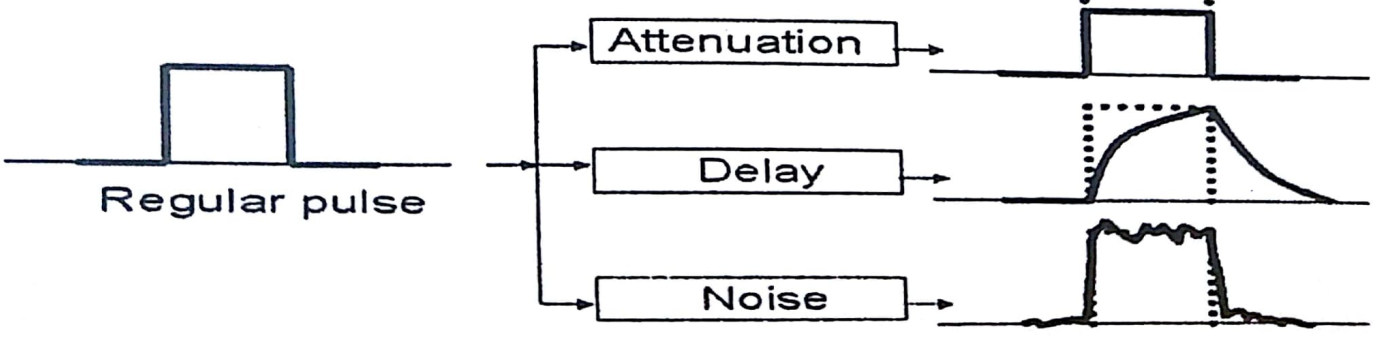
Attenuation



Delay

Attenuation and delay as a function of frequency

Signal level! Power, noise



✓ Noise

- Additional signals inserted between transmitter and receiver
- Thermal
 - Due to thermal agitation of electrons
 - Uniformly distributed
 - White noise
- Intermodulation
 - produce signals at frequency that is the sum and difference of original frequencies sharing a medium.
 - Caused by nonlinearity in Tx, Rx, or channel because of signal strength.

➤ Noise cont.

- Crosstalk
 - A signal from one line is picked up by another
- Impulse
 - Irregular pulses or spikes
 - e.g. External electromagnetic interference
 - Short duration
 - High amplitude
 - Severe effect on digital signal of high data rate.

Radio Communication Channels

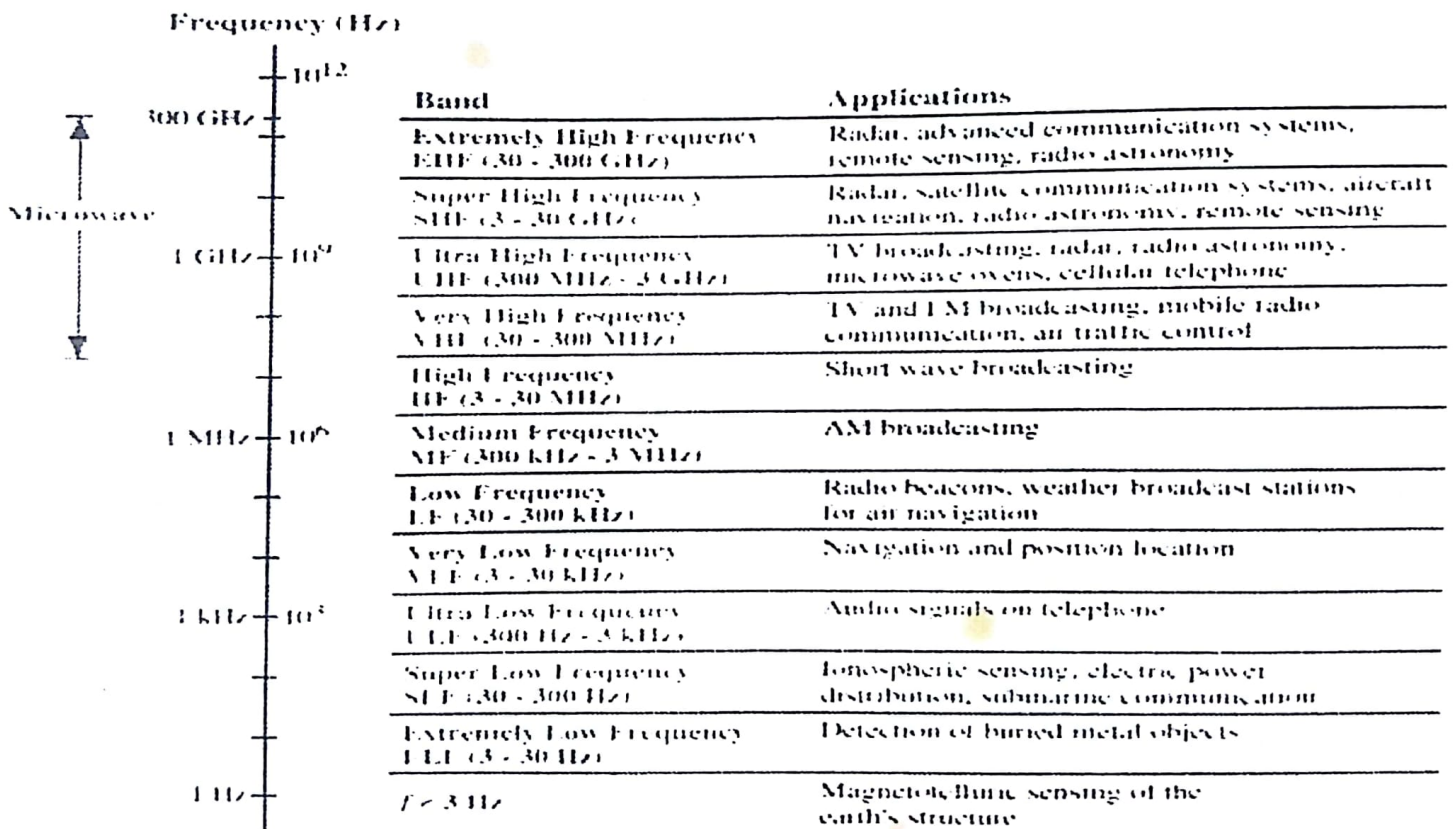
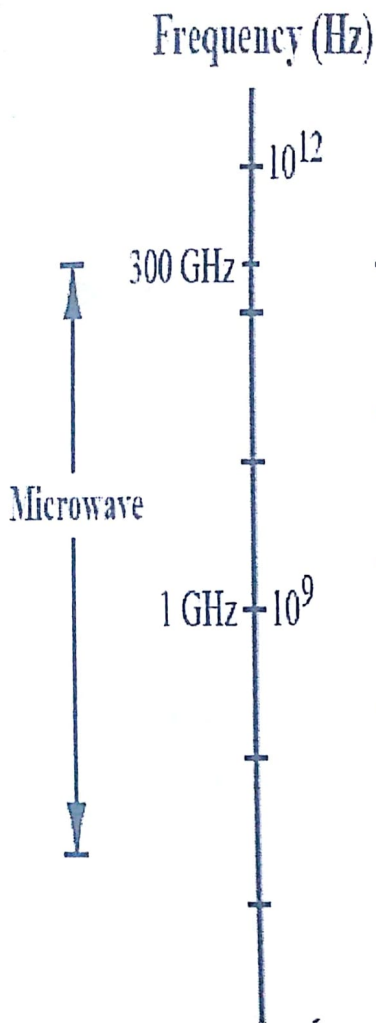


Figure 1-16

1 MHz	10^6	Medium Frequency MF (300 kHz - 3 MHz)	AM broadcasting
		Low Frequency LF (30 - 300 kHz)	Radio beacons, weather broadcast stations for air navigation
		Very Low Frequency VLF (3 - 30 kHz)	Navigation and position location
1 kHz	10^3	Ultra Low Frequency ULF (300 Hz - 3 kHz)	Audio signals on telephone
		Super Low Frequency SLF (30 - 300 Hz)	Ionospheric sensing, electric power distribution, submarine communication
		Extremely Low Frequency ELF (3 - 30 Hz)	Detection of buried metal objects
1 Hz		$f < 3$ Hz	Magnetotelluric sensing of the earth's structure



Band	Applications
Extremely High Frequency EHF (30 - 300 GHz)	Radar, advanced communication systems, remote sensing, radio astronomy
Super High Frequency SHF (3 - 30 GHz)	Radar, satellite communication systems, aircraft navigation, radio astronomy, remote sensing
Ultra High Frequency UHF (300 MHz - 3 GHz)	TV broadcasting, radar, radio astronomy, microwave ovens, cellular telephone
Very High Frequency VHF (30 - 300 MHz)	TV and FM broadcasting, mobile radio communication, air traffic control
High Frequency HF (3 - 30 MHz)	Short wave broadcasting



EE 325: Chapter 2

Introduction to Signals and systems

M. A. Smadi

Outlines

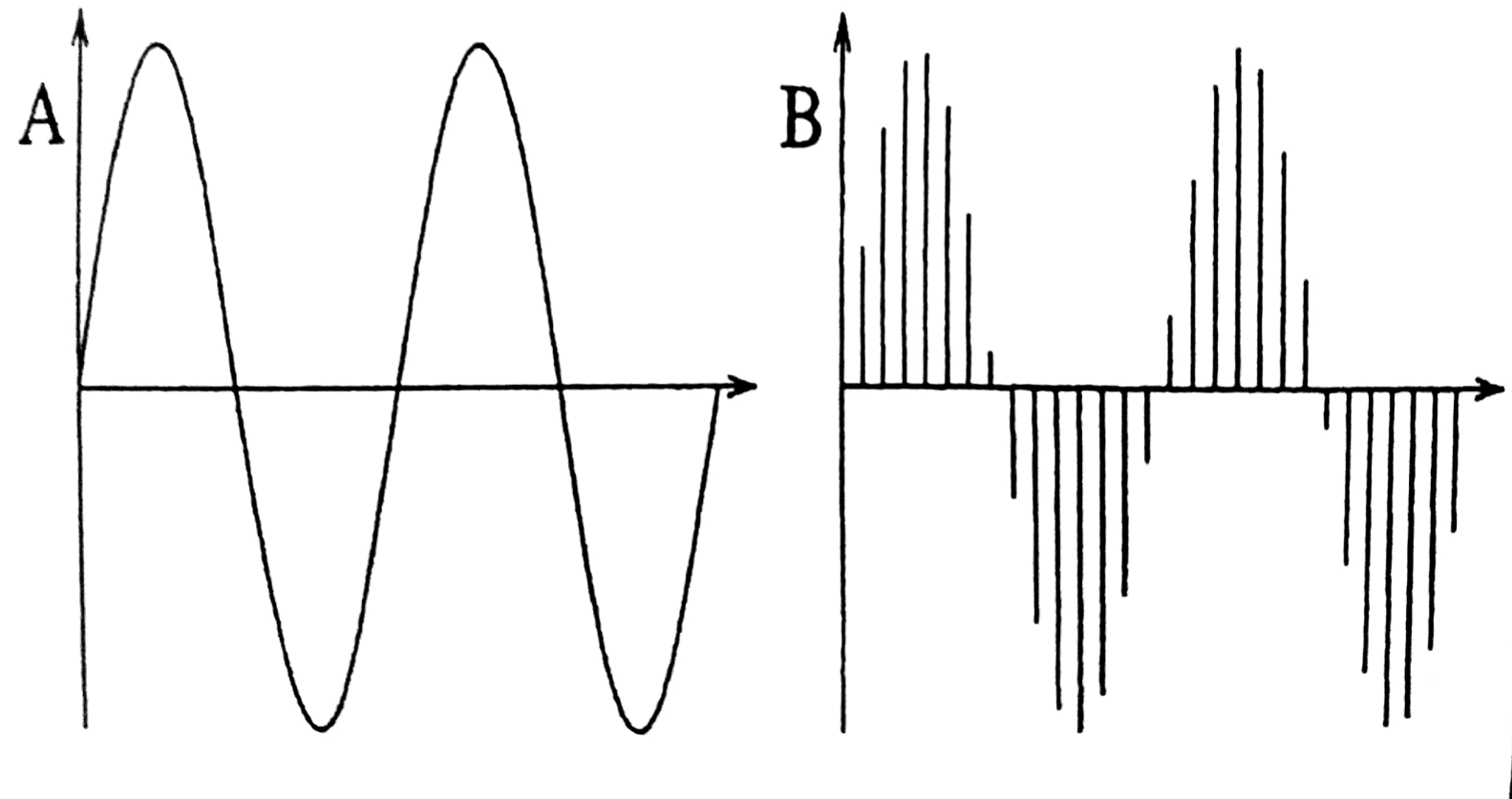
- **Classification of signals and systems**
- **Some useful signal operations**
- **Some useful signals.**
- **Frequency domain representation for periodic signals**
- **Fourier Series Coefficients**
- **Power content of a periodic signal and Parseval' s theorem for the Fourier series**

Classification of Signals

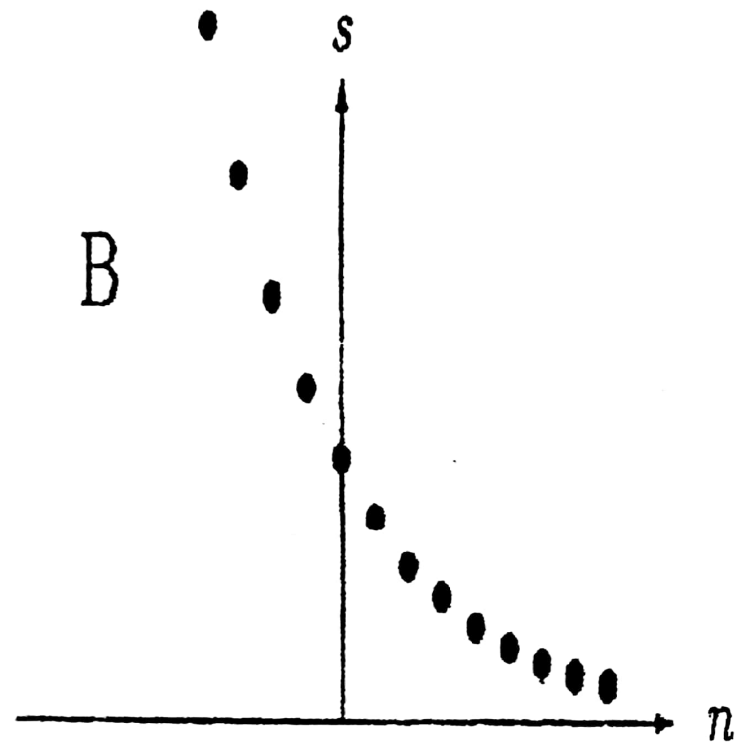
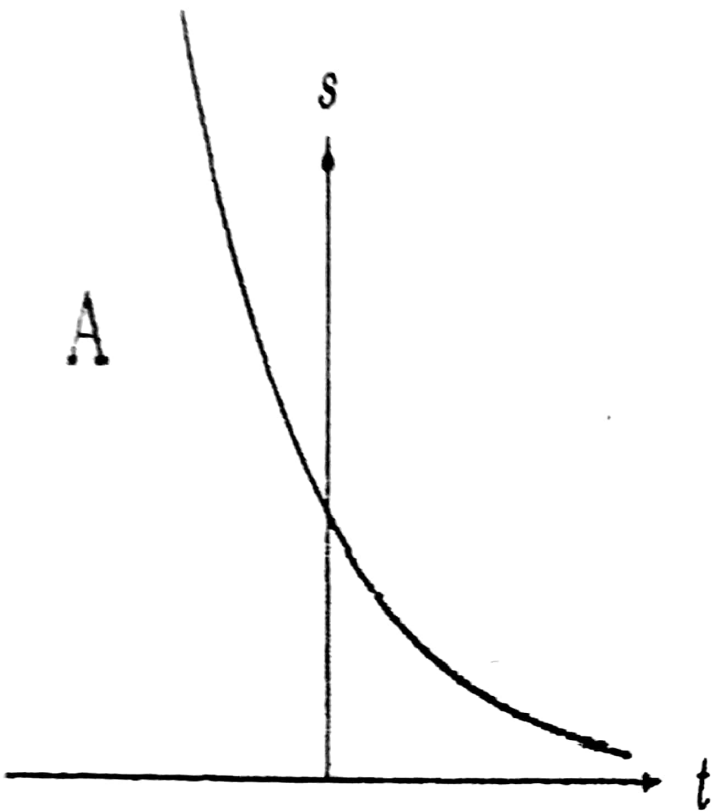
- Continuous-time and discrete-time signals
- Analog and digital signals
- Deterministic and random signals
- Periodic and aperiodic signals
- Power and energy signals
- Causal and non-causal.
- Time-limited and band-limited.
- Base-band and band-pass.
- Wide-band and narrow-band. (BW)

con spectrum

Continuous-time and discrete-time periodic signals



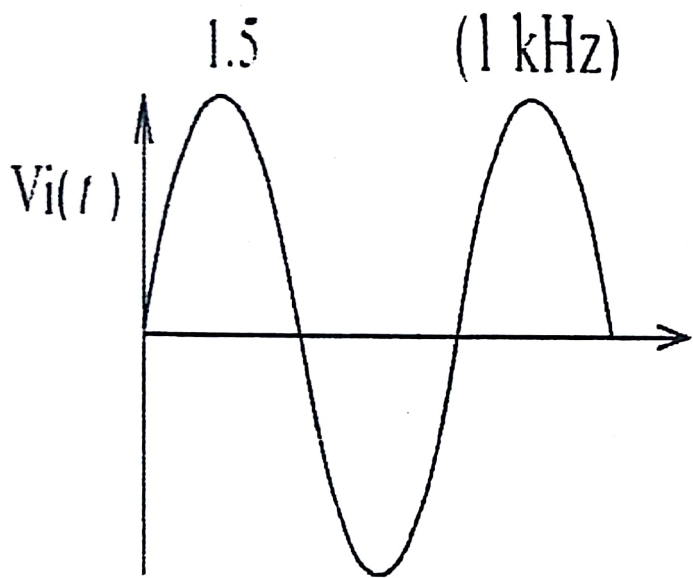
Continuous-time and discrete-time aperiodic signals



Analog & digital signals

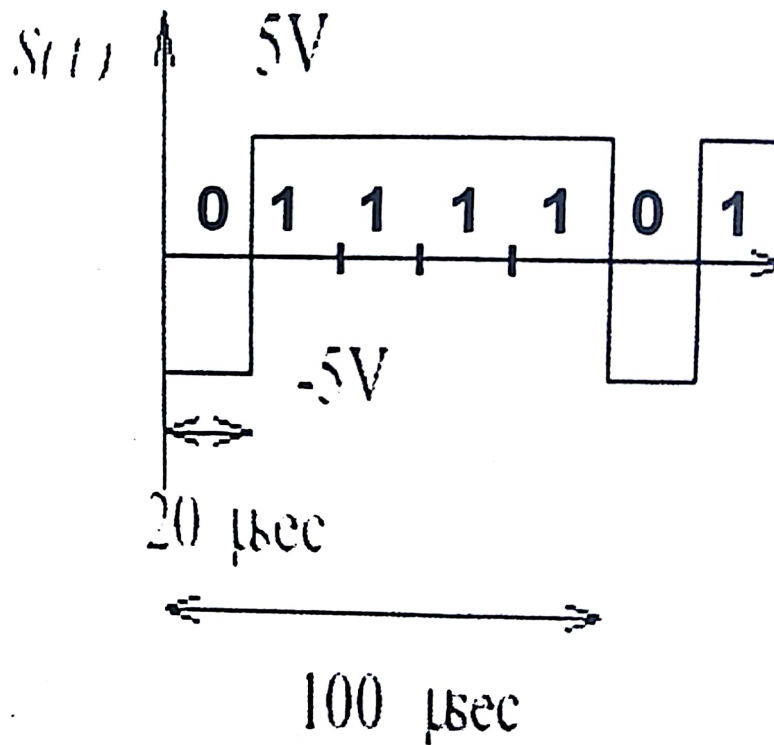
- If a continuous-time signal $g(t)$ can take on any values in a continuous time interval, then $g(t)$ is called an *analog* signal.
- If a discrete-time signal can take on only a finite number of distinct values, $g[n]$ then the signal is called a *digital* signal.

Analog and Digital Signals



peak
= 3 volt

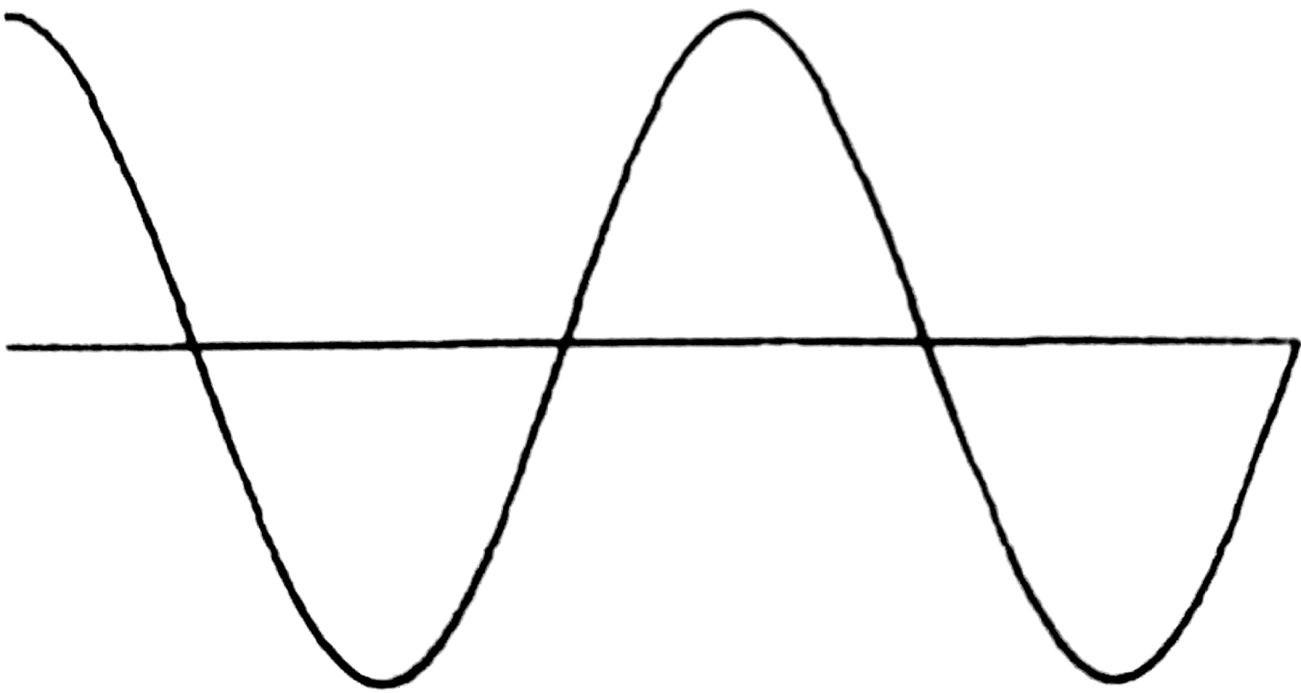
-1.5
 $T = 1 \text{ m second}$



Deterministic signal

- A *Deterministic signal* is uniquely described by a mathematical expression.
- They are reproducible, predictable and well-behaved mathematically.
- Thus, everything is known about the signal for all time.

A deterministic signal



Random signal

- *Random signals* are unpredictable.
- They are generated by systems that contain randomness.
- At any particular time, the signal is a random variable, which may have well defined average and variance, but is not completely defined in value.

A random signal



Periodic and aperiodic Signals

- A signal $x(t)$ is a *periodic* signal if

$$x(t) = x(t + nT_0), \forall t, n \text{ is integer.}$$

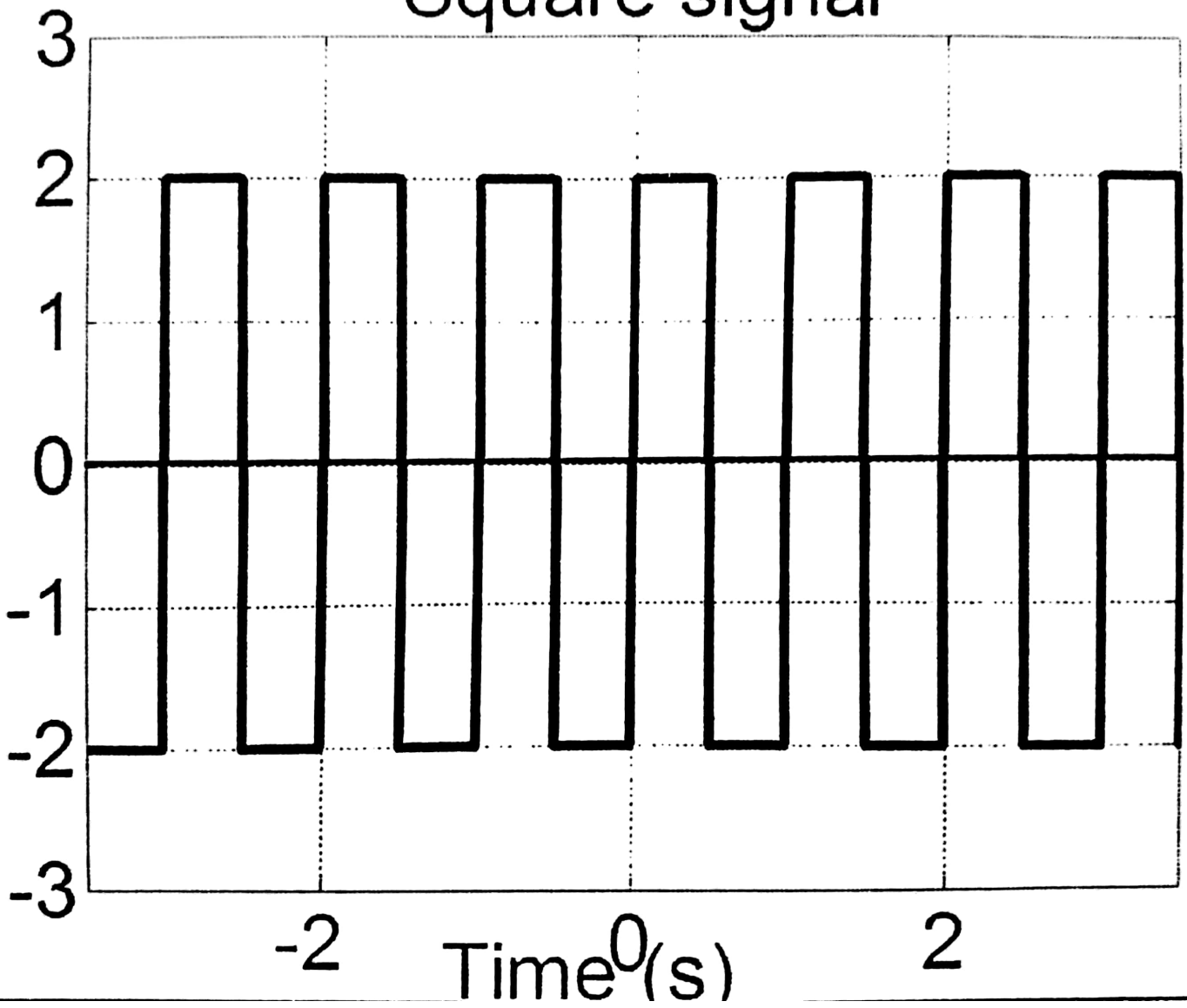
T_0 : period(second)

$f_0 = \frac{1}{T_0}$ (Hz), fundamental frequency

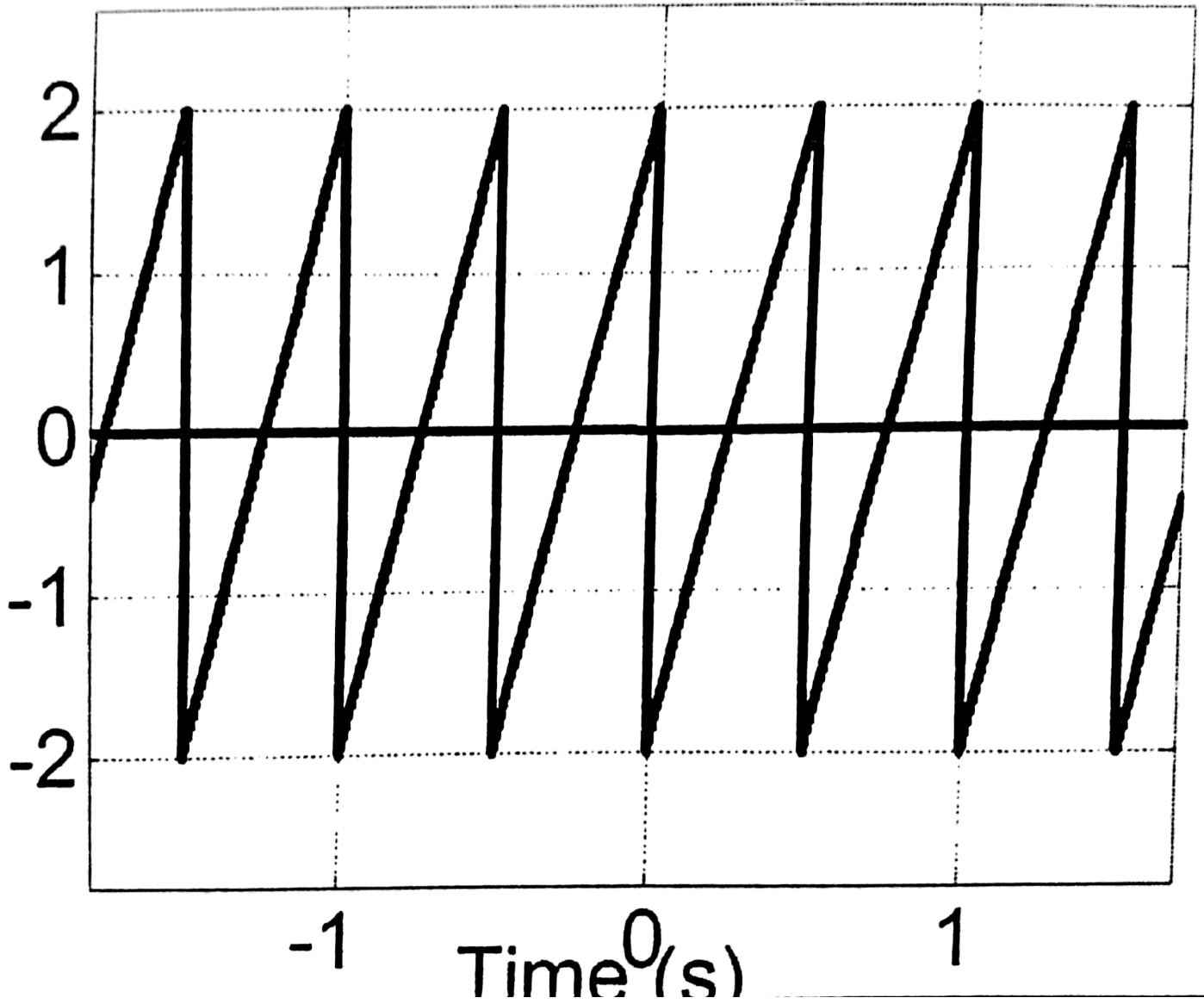
$\omega = 2\pi f$ (rad/sec), angular (radian) frequency

- Otherwise, it is *aperiodic* signal.

Square signal



Sawtooth signal



- A simple harmonic oscillation is mathematically described by

$$x(t) = A \cos(\omega t + \theta), \quad \text{for } -\infty < t < \infty$$

- This signal is completely characterized by three parameters:

A: is the amplitude (peak value) of $x(t)$.

ω : is the radial frequency in (rad/s),

θ : is the phase in radians (rad)

Example:

Determine whether the following signals are periodic. In case a signal is periodic, specify its fundamental period. T_0

a) $x_1(t) = 3 \cos(3\pi t + \pi/6)$, periodic

b) $x_2(t) = 2 \sin(100\pi t)$, "

c) $x_3(t) = x_1(t) + x_2(t)$, "

d) $x_4(t) = 3 \cos(3\pi t + \pi/6) + 2 \sin(10 t)$, non periodic

e) $x_5(t) = 2 \exp(-j 20 \pi t)$

$T_0 = 2\pi$

$e^{j\theta} = \cos \theta + j \sin \theta$
 $e^{j\omega t} = \cos \omega t + j \sin \omega t$

Power and Energy signals

- A signal with finite energy is an **energy signal**

$$E_g = \int_{-\infty}^{+\infty} |g(t)|^2 dt < \infty$$

- A signal with finite power is a **power signal**

$$P_g = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{+T/2} |g(t)|^2 dt < \infty$$

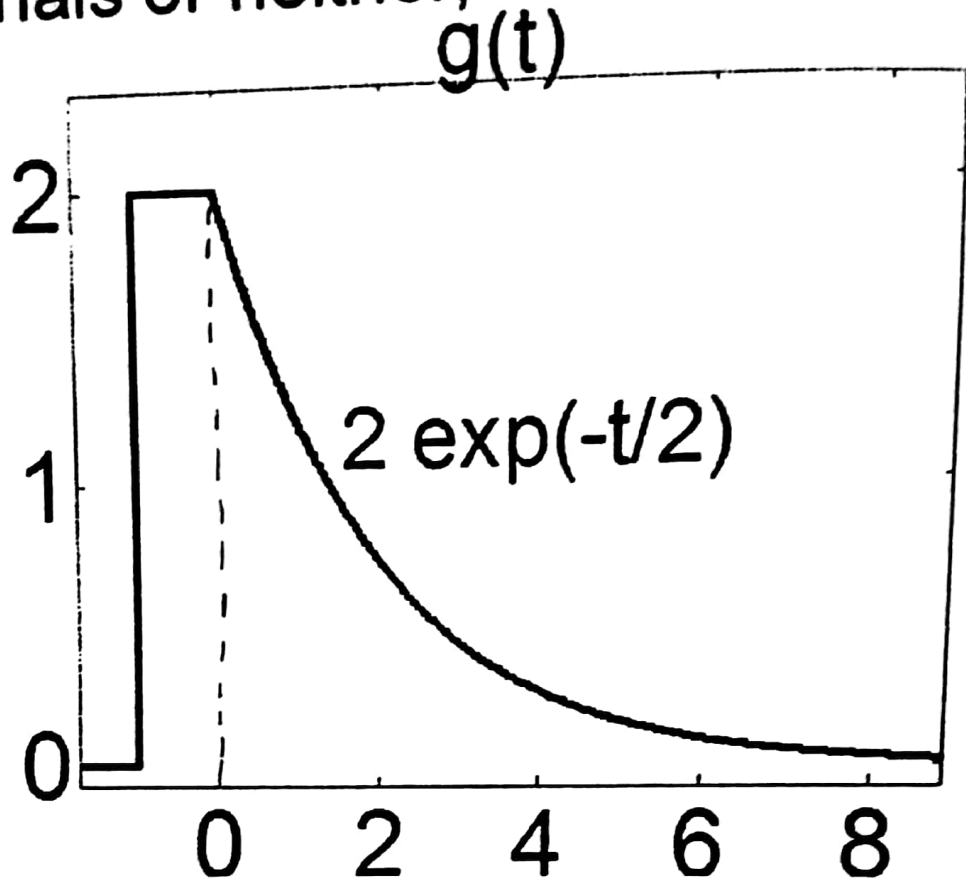
Power of a Periodic Signal

- The power of a periodic signal $x(t)$ with period T_0 is defined as the mean-square value over a period

$$P_x = \frac{1}{T_0} \int_{-T_0/2}^{+T_0/2} |x(t)|^2 dt$$

Example

- Determine whether the signal $g(t)$ is power or energy signals or neither;



$$E_g = 8$$

Exercise

- Determine whether the signals are power or energy signals or neither:

1) $x(t) = u(t)$ *Power*

2) $y(t) = A \sin t$

3) $s(t) = t u(t)$

4) $z(t) = \delta(t)$ *energy*

5) $v(t) = \cos(10\pi t)u(t)$

6) $w(t) = \sin 2\pi t [u(t) - u(t - 2\pi)]$

Exercise

- Determine whether the signals are power or energy signals or neither

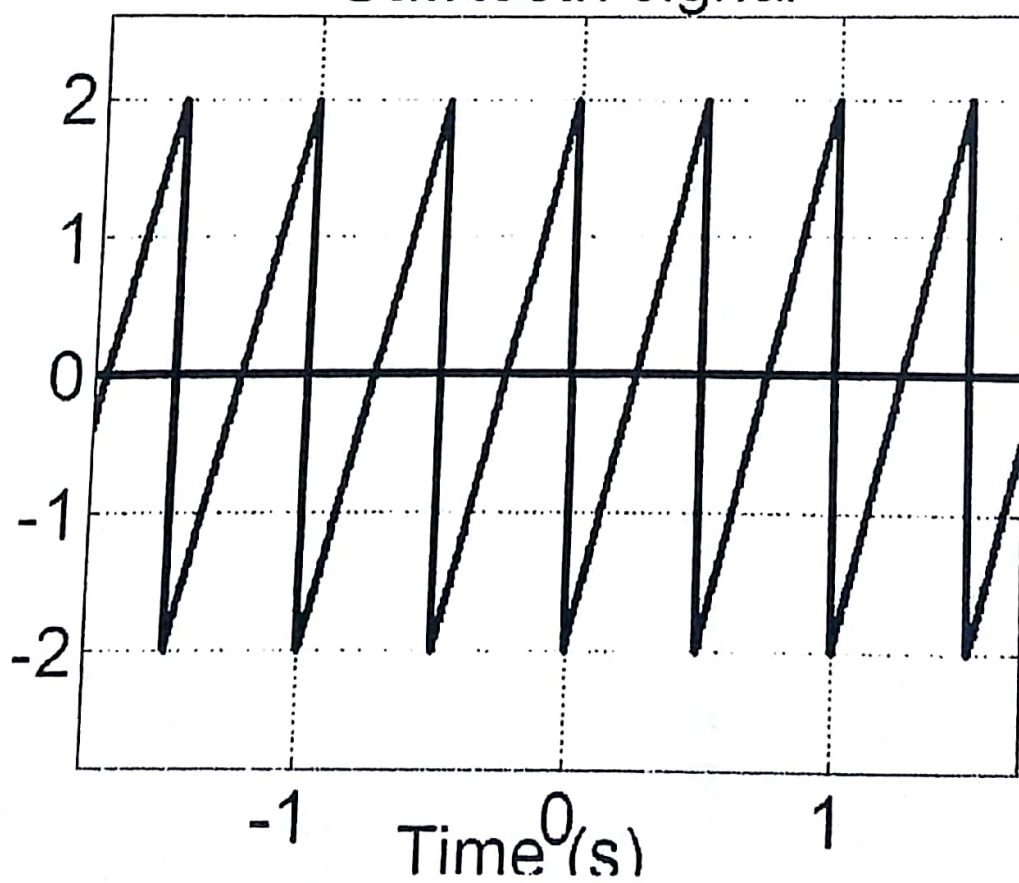
$$1) \quad x(t) = a \cos(\omega_1 t + \theta_1) + b \cos(\omega_2 t + \theta_2)$$

$$2) \quad x(t) = a \cos(\omega_1 t + \theta_1) + b \cos(\omega_1 t + \theta_2)$$

$$3) \quad y(t) = \sum_{n=1}^{\infty} c_n \cos(\omega_n t + \theta_n)$$

Exercise: Determine the suitable measures for the signal $x(t)$

Sawtooth signal



Some Useful Functions

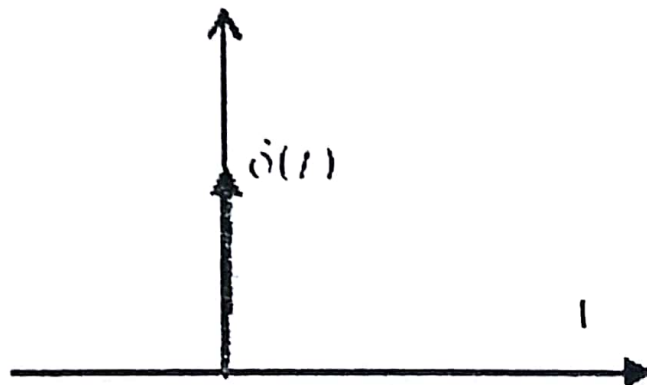
- Unit impulse function
- Unit step function
- Rectangular function
- Triangular function
- Sampling function
- Sinc function
- Sinusoidal, exponential and logarithmic functions

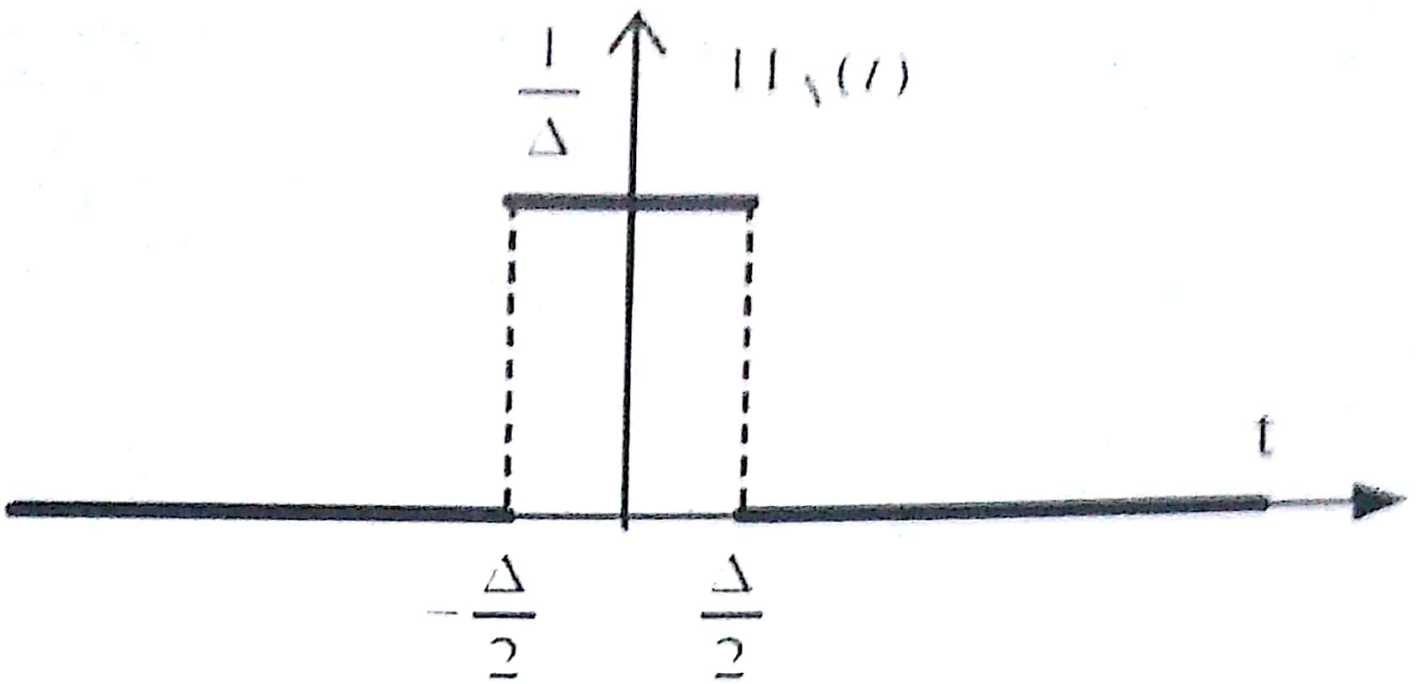
Unit impulse function

- The *unit impulse function*, also known as the *dirac delta function*, $\delta(t)$, is defined by

$$\delta(t) = \begin{cases} \infty, & t = 0 \\ 0, & t \neq 0 \end{cases}$$

$$\text{and } \int_{-\infty}^{+\infty} \delta(t) dt = 1$$





$$\Delta \rightarrow 0 \longrightarrow$$

area
 constant
 impulse

- Multiplication of a function by $\delta(t)$

$$g(t) \delta(t - \tau) = g(\tau) \delta(t - \tau)$$

$$g(t) \delta(t) = g(0) \delta(t)$$

- We can also prove that

$$\int_{-\infty}^{+\infty} s(t) \delta(t - \tau) dt = s(\tau)$$

$$\int_{-\infty}^{+\infty} s(t) \delta(t) dt = s(0)$$

Unit step function

- The *unit step function* $u(t)$ is

$$u(t) = \begin{cases} 1, & t \geq 0 \\ 0, & t < 0 \end{cases}$$

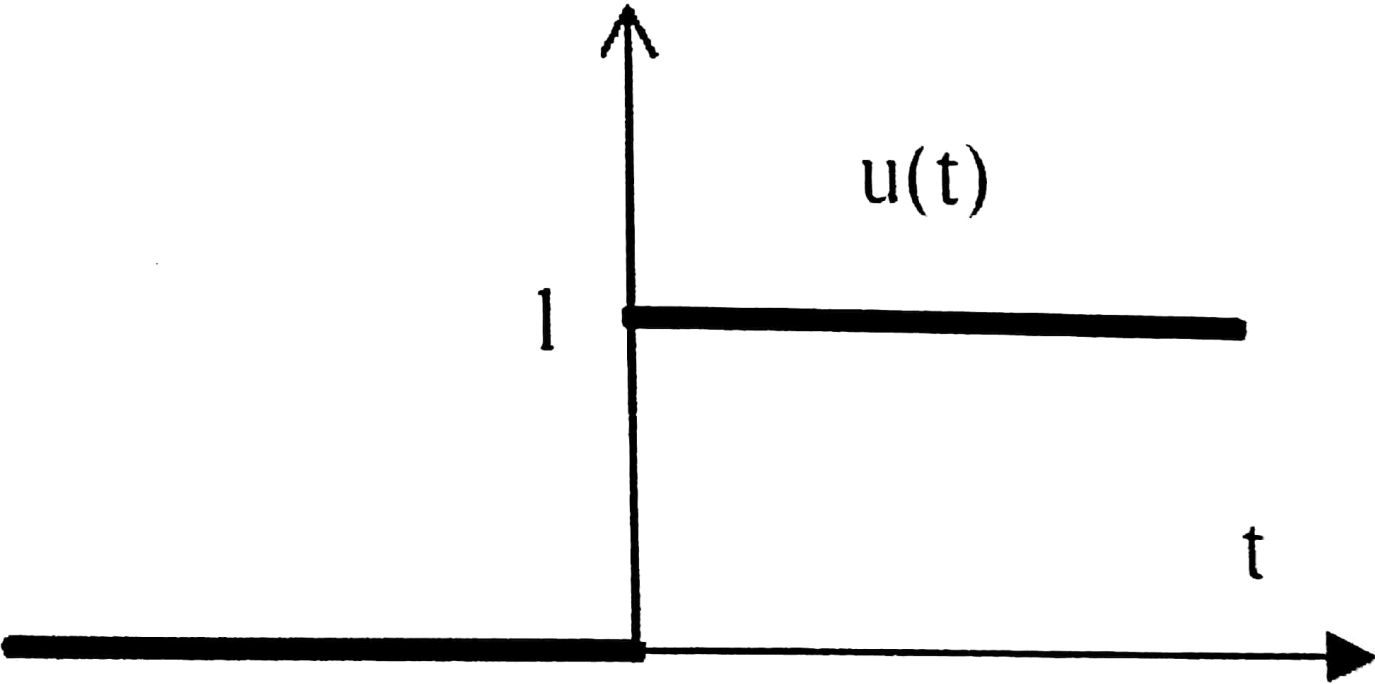
- $u(t)$ is related to $\delta(t)$ by

$$u(t) = \int_{-\infty}^t \delta(\tau) d\tau$$

$$\frac{du}{dt} = \delta(t)$$

27

Unit step

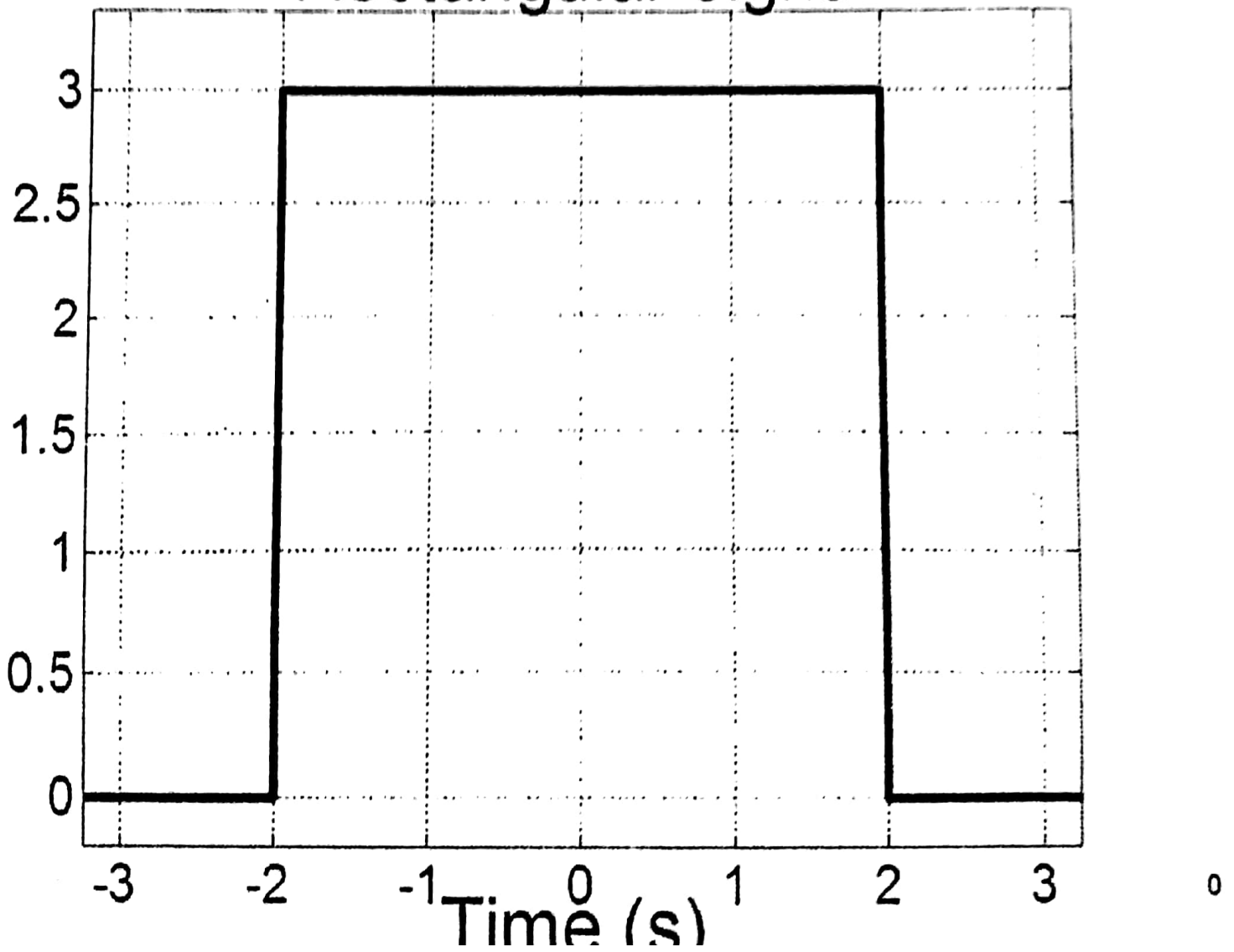


Rectangular function

- A single *rectangular pulse* is denoted by

$$\text{rect}\left(\frac{t}{\tau}\right) = \begin{cases} 1, & |t| < \tau/2 \\ 0.5, & |t| = \tau/2 \\ 0, & |t| > \tau/2 \end{cases}$$

Rectangular signal



تکاملی Triangular function

- A *triangular function* is denoted by

$$\Delta\left(\frac{t}{\tau}\right) = \begin{cases} 1 - 2\left|\frac{t}{\tau}\right|, & \left|\frac{t}{\tau}\right| < \frac{1}{2} \\ 0, & \left|\frac{t}{\tau}\right| > \frac{1}{2} \end{cases}$$

- Sinc function

$$\text{Sinc}(0) = 1$$

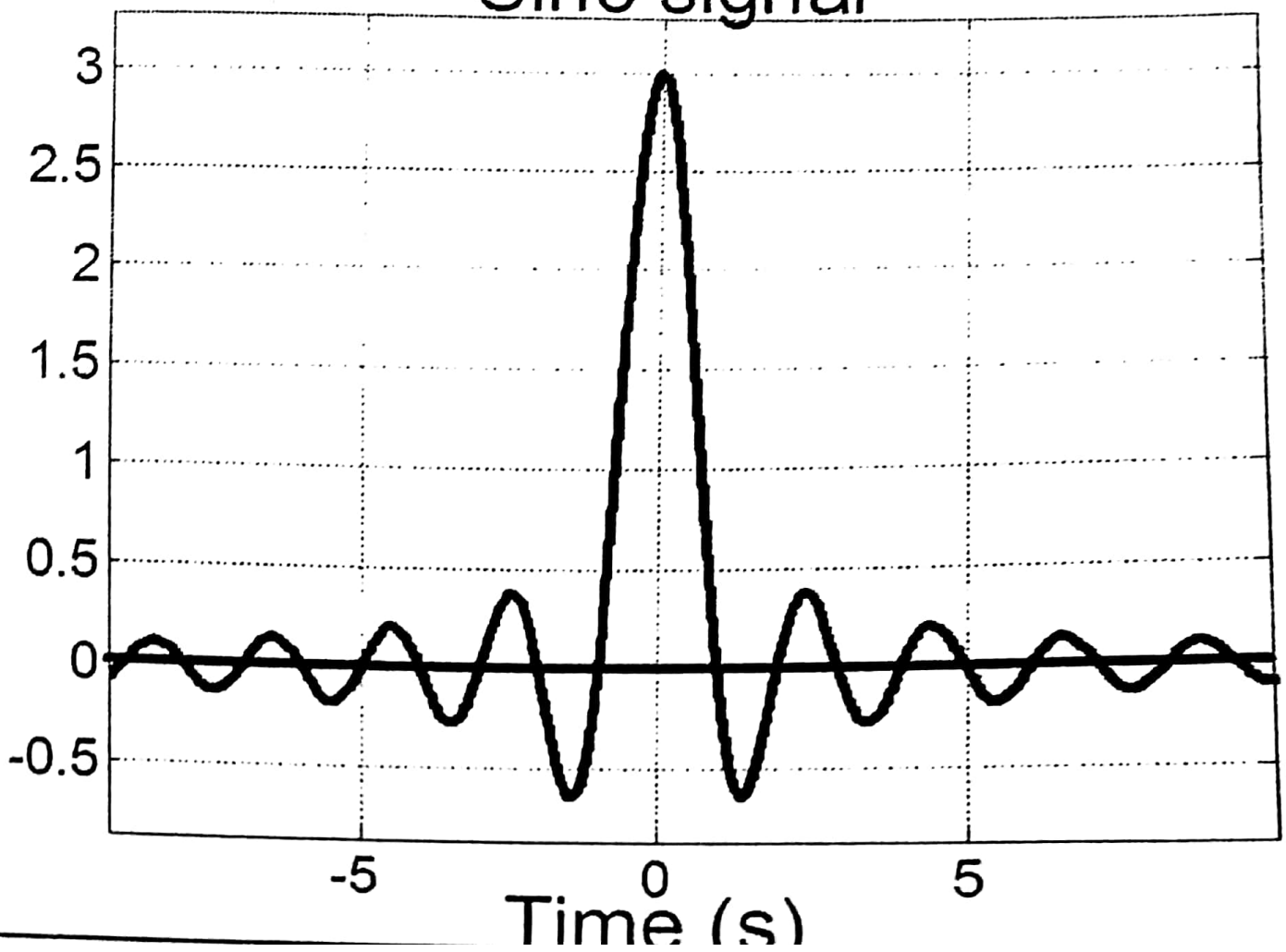
$$\text{sinc}(x) = \frac{\sin(\pi x)}{\pi x}$$

→ zero crossing
integer
1, 2, 3 — 7 etc

- Sampling function

$$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s), \quad T_s : \text{samplig interval}$$

Sinc signal



Some Useful Signal Operations

- Time shifting

$$g(t - \tau) \quad \text{(shift right or delay)}$$

$$g(t + \tau) \quad \text{(shift left or advance)}$$

- Time scaling

$$g(at), |a| > 1 \text{ is compression}$$

$$g(at), |a| < 1 \text{ is expansion}$$

$$g\left(\frac{t}{a}\right), |a| > 1 \text{ is expansion}$$

$$g\left(\frac{t}{a}\right), |a| < 1 \text{ is compression}$$

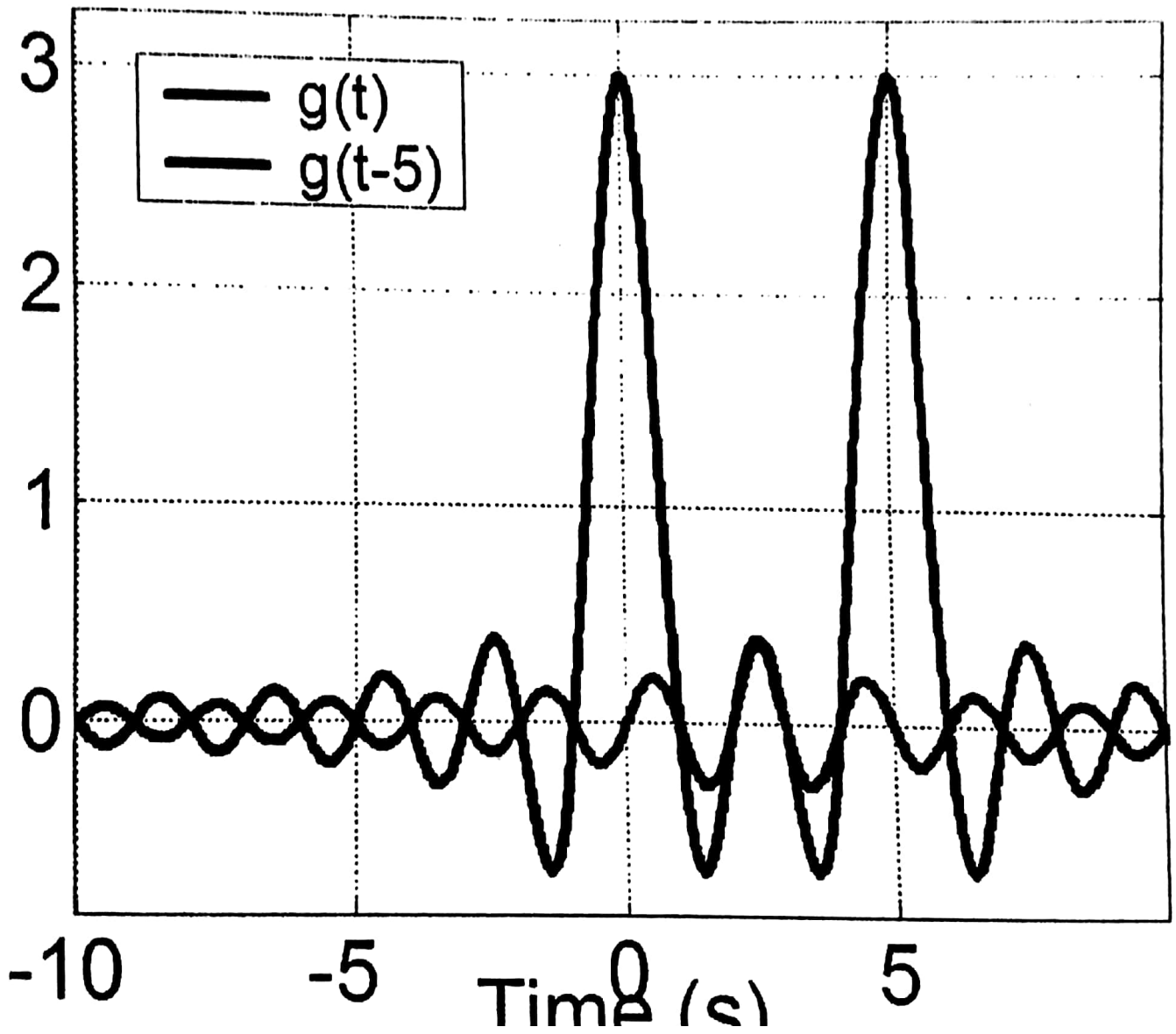
Signal operations cont.

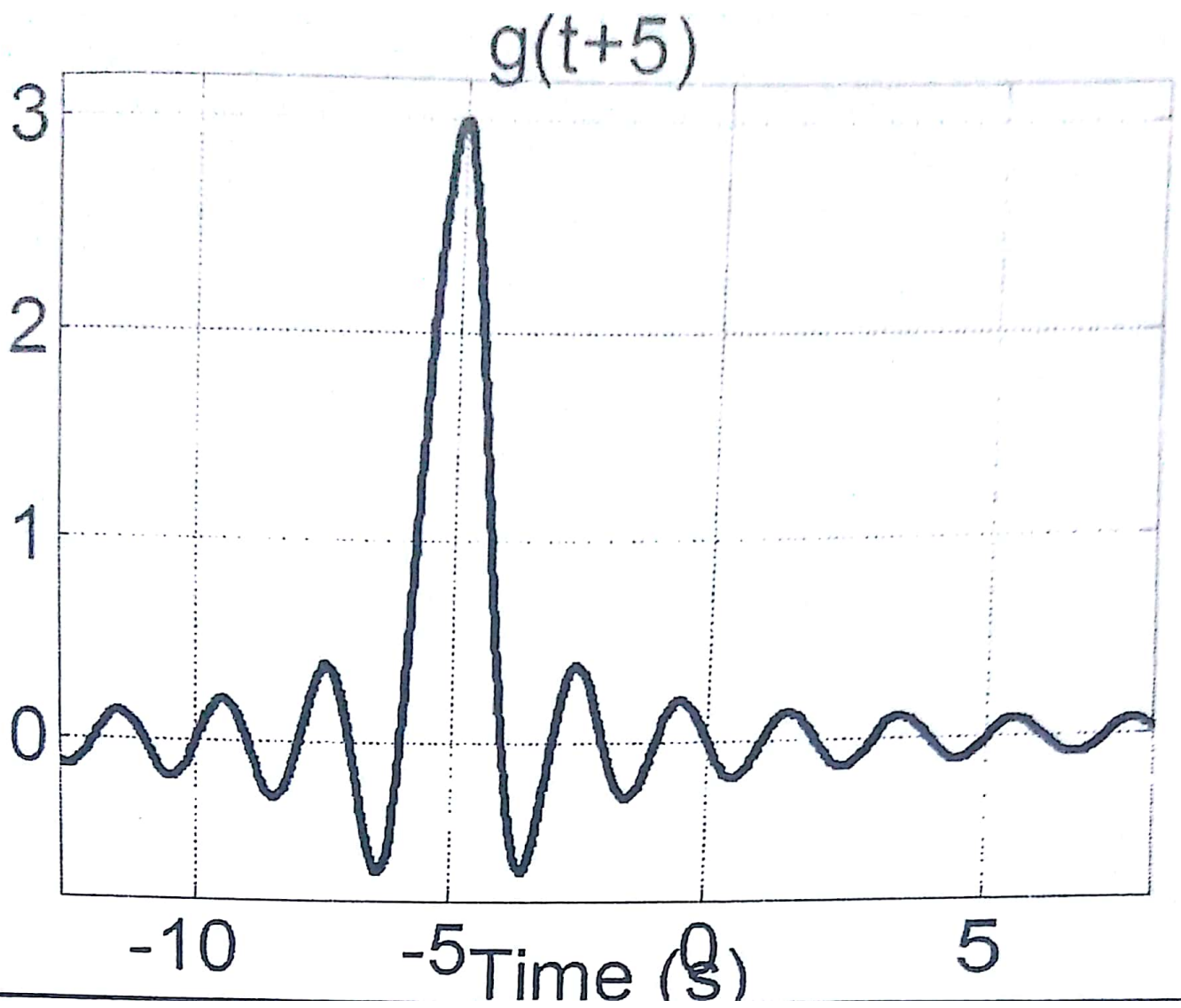
- Time inversion

$g(-t)$: mirror image of $g(t)$ about Y-axis

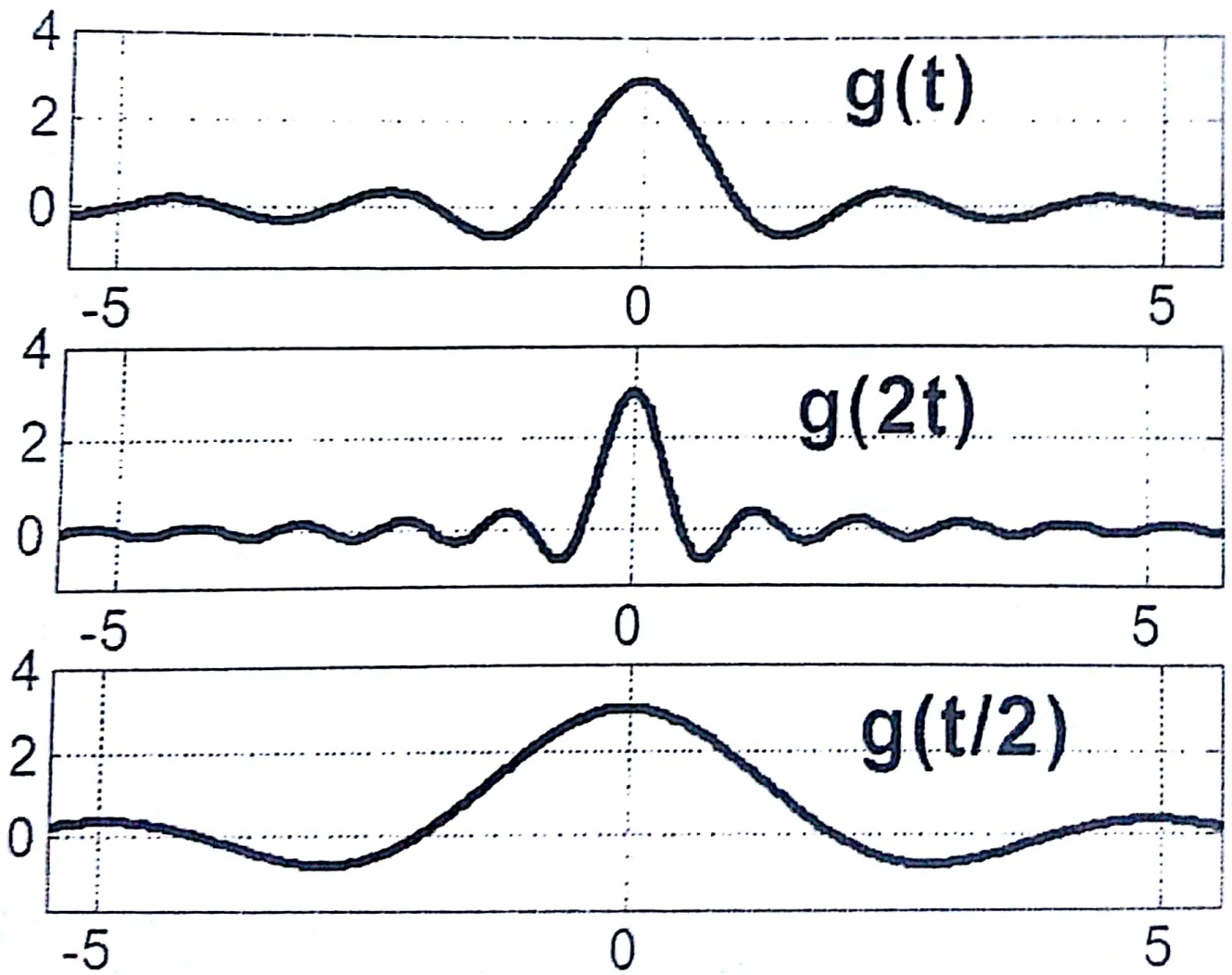
$g(-t + \tau)$: shift right of $g(-t)$

$g(-t - \tau)$: shift left of $g(-t)$

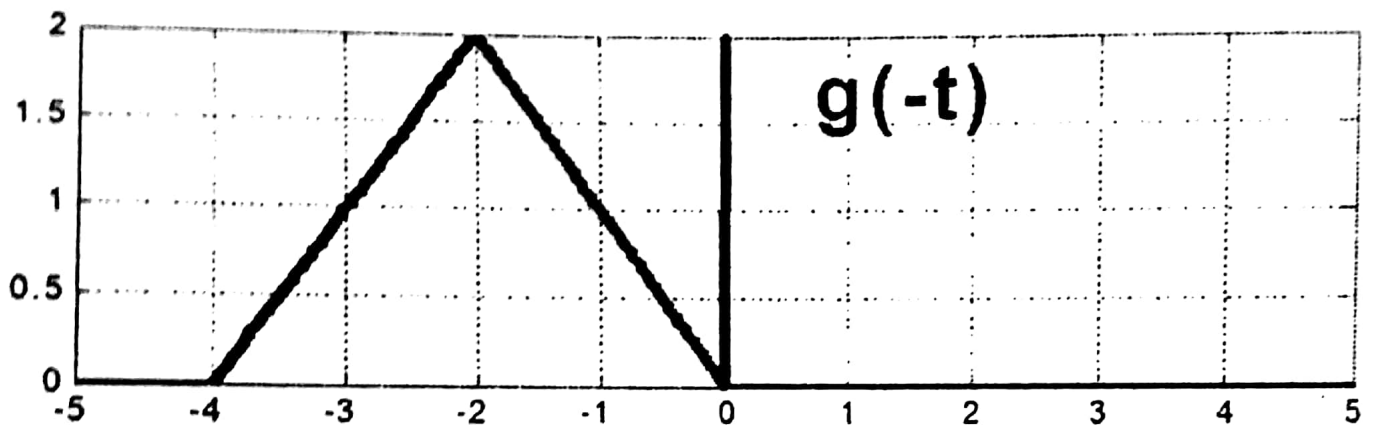
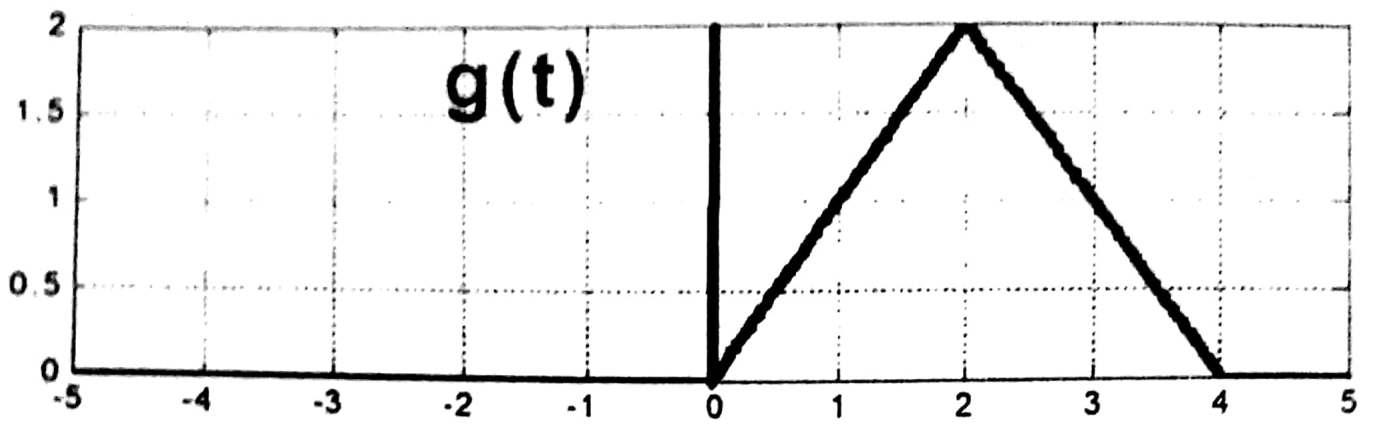




Scaling



Time Inversion



Inner product of signals

- Inner product of two complex signals $x(t)$, $y(t)$ over the interval $[t_1, t_2]$ is

$$(x(t), y(t)) = \int_{t_1}^{t_2} x(t) y^*(t) dt$$

If inner product=0, $x(t)$, $y(t)$ are orthogonal.

سینالوں کے درمیان
عاماً
عاماً

Inner product cont.

- The approximation of $x(t)$ by $y(t)$ over the interval $[t_1, t_2]$ is given by

$$x(t) = C y(t)$$

- The optimum value of the constant C that minimize the energy of the error signal

$$e(t) = x(t) - C y(t)$$

is given by

$$C = \frac{1}{E_y} \int_{t_1}^{t_2} x(t) y(t) dt$$

Power and energy of orthogonal signals

- The power/energy of the sum of mutually orthogonal signals is sum of their individual powers/energies ,i.e., if

$$x(t) = \sum_{i=1}^n g_i(t)$$

Such that $g_i(t), i = 1, \dots, n$ are mutually orthogonal, then

$$P_x = \sum_{i=1}^n P_{g_i}$$

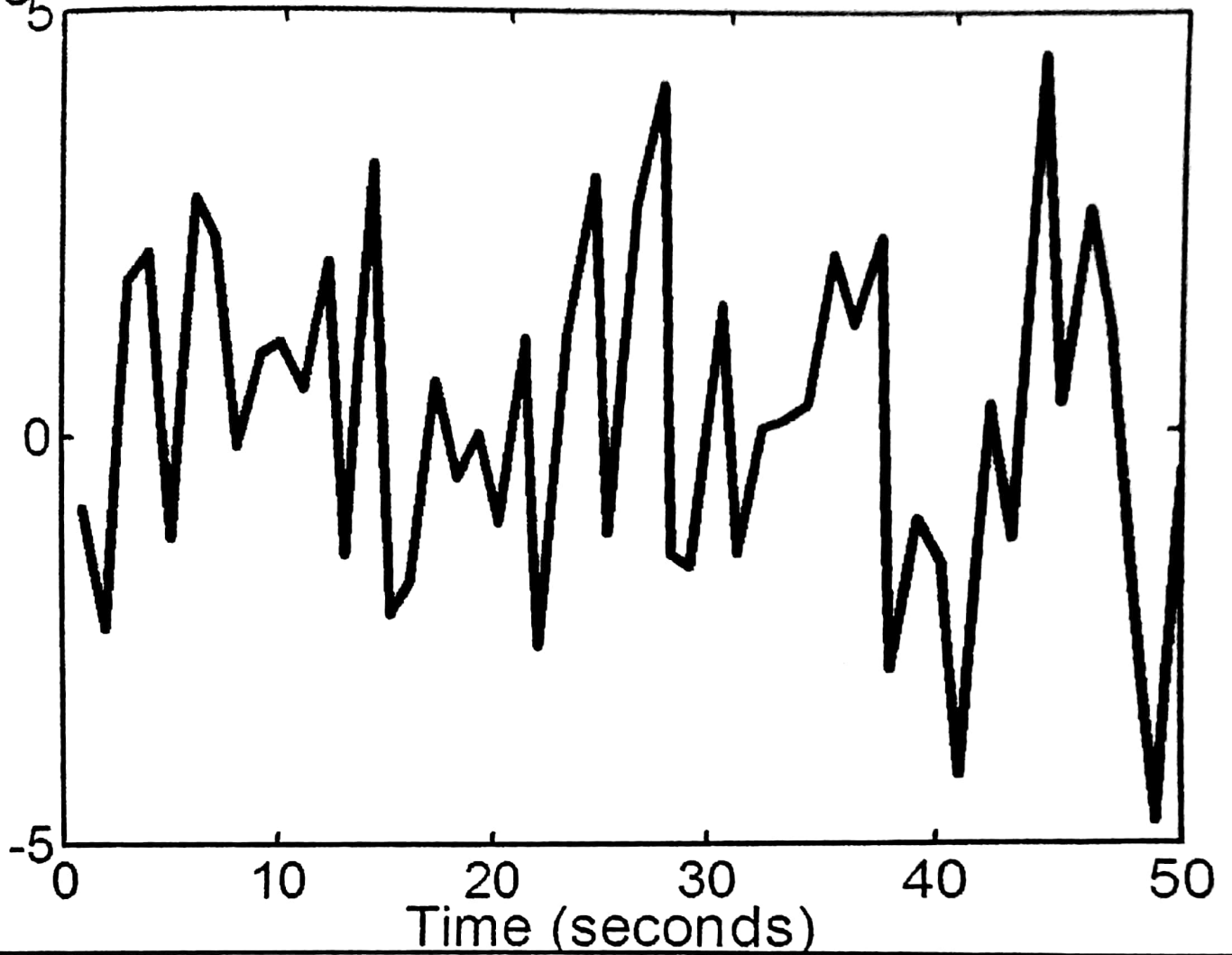
Time and Frequency Domains representations of signals

- Time domain: an oscilloscope displays the amplitude versus time
- Frequency domain: a spectrum analyzer displays the amplitude or power versus frequency
- Frequency-domain display provides information on bandwidth and harmonic components of a signal

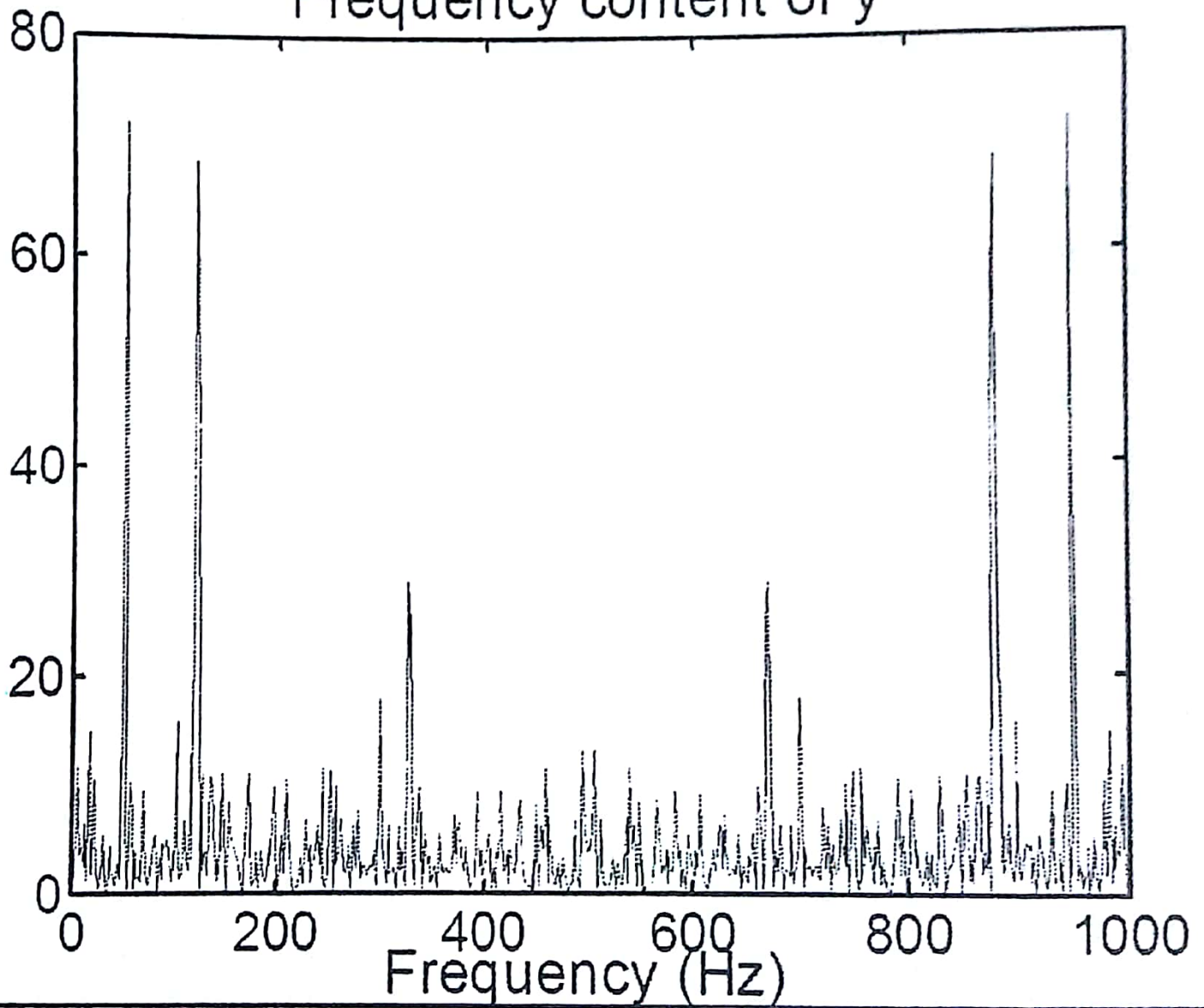
Benefit of Frequency Domain Representation

- Distinguishing a signal from noise
 $x(t) = \sin(2\pi 50t) + \sin(2\pi 120t);$
 $y(t) = x(t) + \text{noise};$
- Selecting frequency bands in Telecommunication system

Signal Corrupted with Zero-Mean Random Noise



Frequency content of y



Fourier Series Coefficients

- The frequency domain representation of a ***periodic signal*** is obtained from the **Fourier series expansion**.
- The frequency domain representation of a ***non-periodic signal*** is obtained from the **Fourier transform**.

-
- The ***Fourier series*** is an effective technique for describing periodic functions. It provides a method for expressing a periodic function as a linear combination of sinusoidal functions.
 - Trigonometric Fourier Series
 - Compact trigonometric Fourier Series
 - Complex Fourier Series

Trigonometric Fourier Series

$$x(t) = a_0 + \sum_{n=1}^{\infty} (a_n \cos 2\pi n f_0 t + b_n \sin 2\pi n f_0 t)$$

$$a_n = \frac{2}{T_0} \int_{T_0} x(t) \cos(2\pi n f_0 t) dt$$

$$b_n = \frac{2}{T_0} \int_{T_0} x(t) \sin(2\pi n f_0 t) dt$$

Trigonometric Fourier Series cont.

$$a_0 = \frac{1}{T_0} \int_{T_0} x(t) dt$$

Compact trigonometric Fourier series

$$x(t) = c_0 + \sum_{k=1}^{\infty} c_n \cos(2\pi n f_0 t - \theta_n)$$

$$c_n = \sqrt{a_n^2 + b_n^2}, \quad c_0 = a_0$$

$$\theta_n = \tan^{-1} \left(\frac{b_n}{a_n} \right)$$

Complex Fourier Series

- If $x(t)$ is a periodic signal with a fundamental period $T_0 = 1/f_0$

$$x(t) = \sum_{n=-\infty}^{\infty} D_n e^{j2\pi n f_0 t}$$

- D_n are called the *Fourier coefficients*

$$D_n = \frac{1}{T_0} \int_{T_0} x(t) e^{-j2\pi n f_0 t} dt$$

Complex Fourier Series cont.

$$D_n = \frac{1}{2} c_n e^{j\theta_n}$$

$$D_{-n} = \frac{1}{2} c_n e^{-j\theta_n}$$

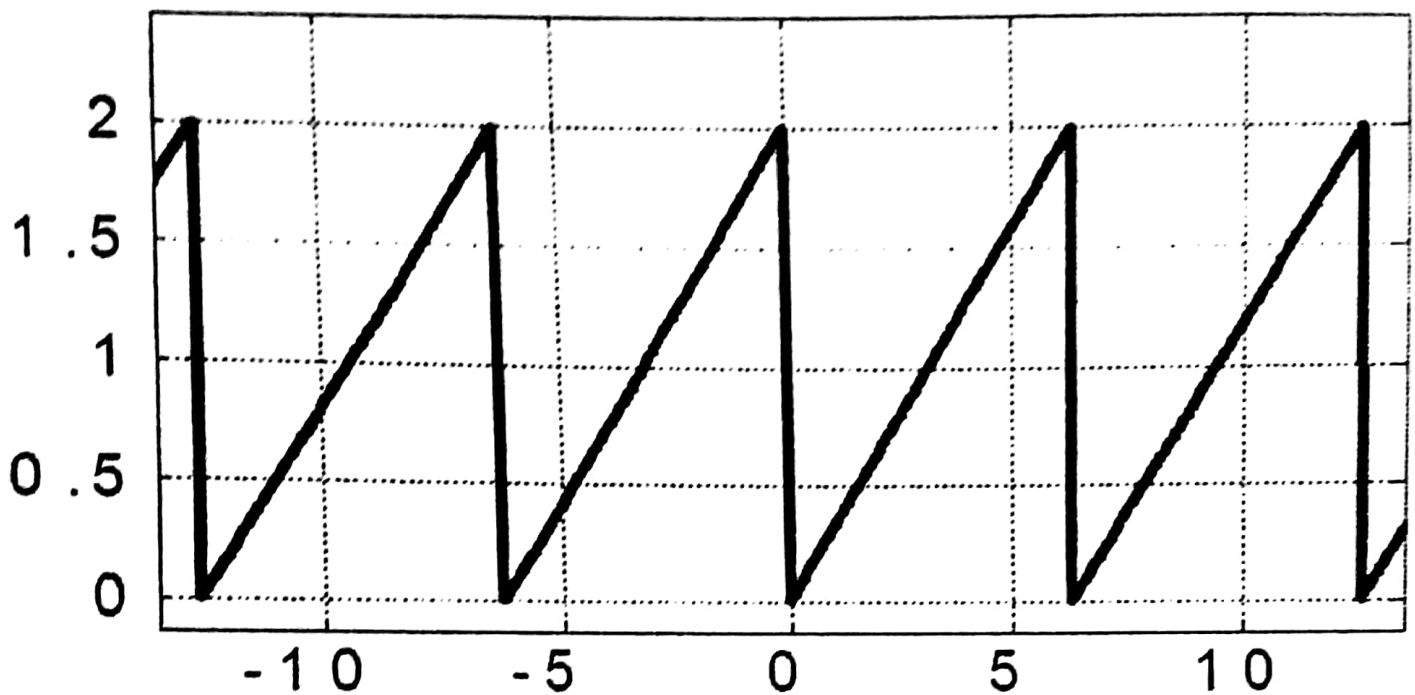
$$D_n = |D_n| e^{j\theta_n} \quad \text{and} \quad D_{-n} = |D_n| e^{-j\theta_n}$$

Frequency Spectra

- A plot of $|D_n|$ versus the frequency is called the **amplitude spectrum** of $x(t)$.
- A plot of the phase θ_n versus the frequency is called the **phase spectrum** of $x(t)$.
- **The frequency spectra** of $x(t)$ refers to the amplitude spectrum and phase spectrum.

Example

- Find the exponential Fourier series and sketch the corresponding spectra for the sawtooth signal with period 2π

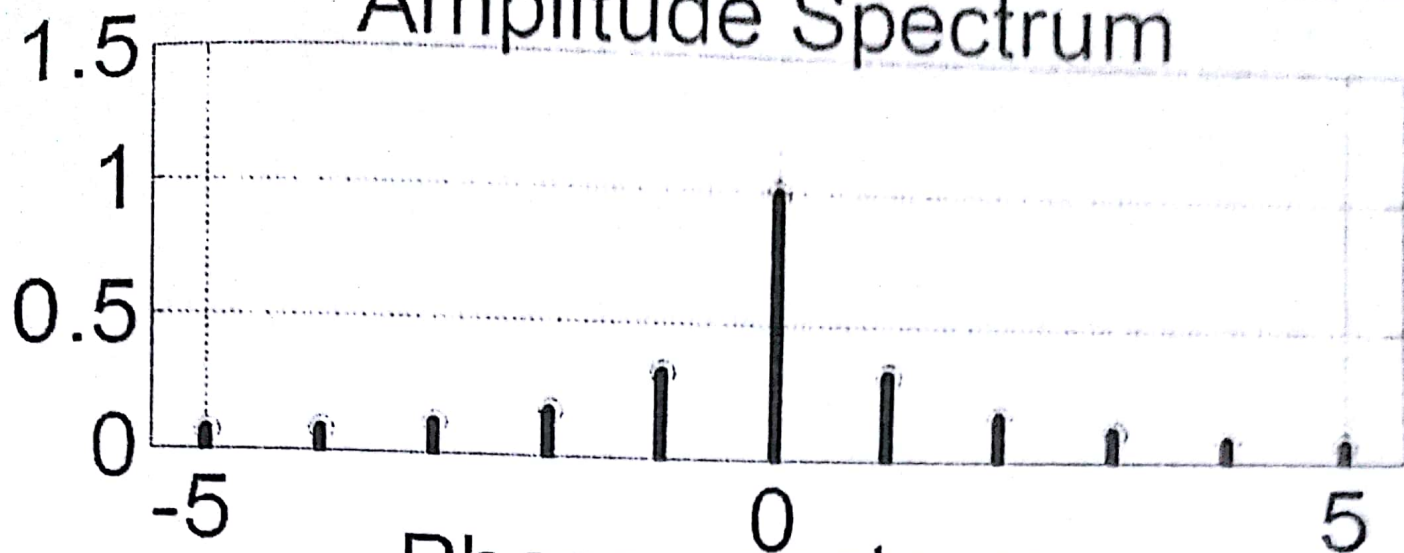


$$D_n = \frac{1}{T_0} \int_{T_0} x(t) e^{-j2\pi n f_0 t} dt$$

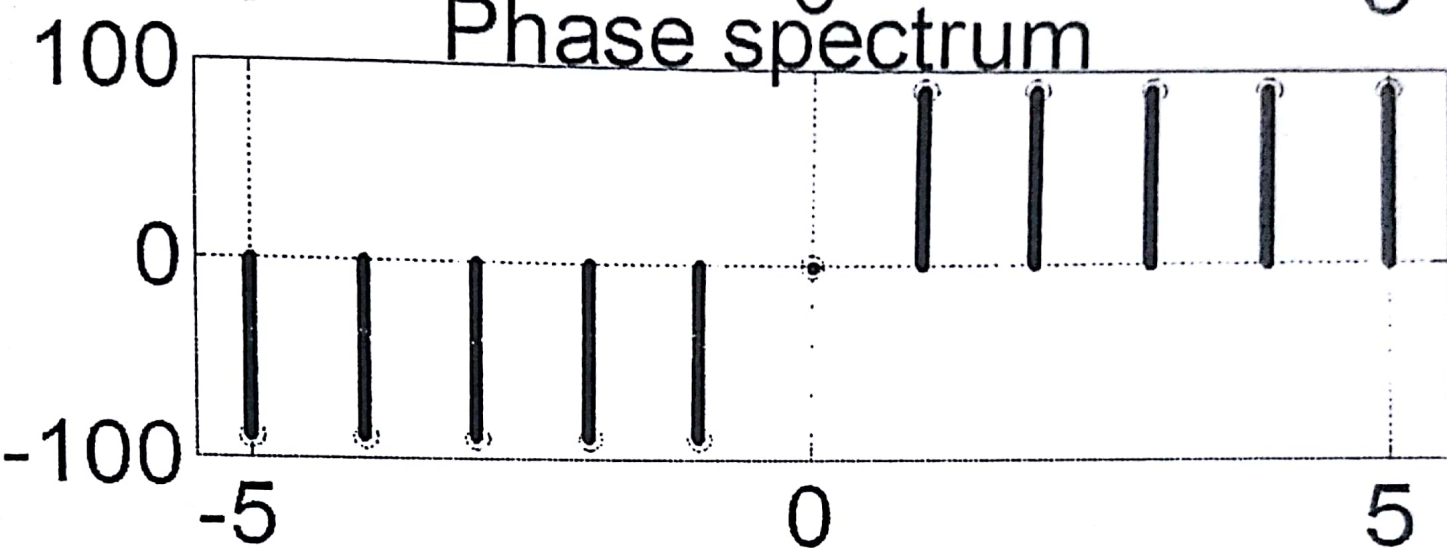
$$\int t e^{at} dt = \frac{e^{at}}{a^2} (at - 1)$$

- $D_n = j/(\pi n)$; for $n \neq 0$
- $D_0 = 1$;

Amplitude Spectrum



Phase spectrum



Power Content of a Periodic Signal

- The power content of a periodic signal $x(t)$ with period T_0 is defined as the mean-square value over a period

$$P = \frac{1}{T_0} \int_{-T_0/2}^{+T_0/2} |x(t)|^2 dt$$

Parseval's Power Theorem

- Parseval's power theorem series states that if $x(t)$ is a periodic signal with period T_0 , then

$$\frac{1}{T_0} \int_{-T_0/2}^{+T_0/2} |x(t)|^2 dt = \begin{cases} \sum_{n=-\infty}^{\infty} |D_n|^2 \\ c_0^2 + \sum_{n=1}^{\infty} \frac{c_n^2}{2} \\ a_0^2 + \sum_{n=1}^{\infty} \frac{a_n^2}{2} + \sum_{n=1}^{\infty} \frac{b_n^2}{2} \end{cases}$$

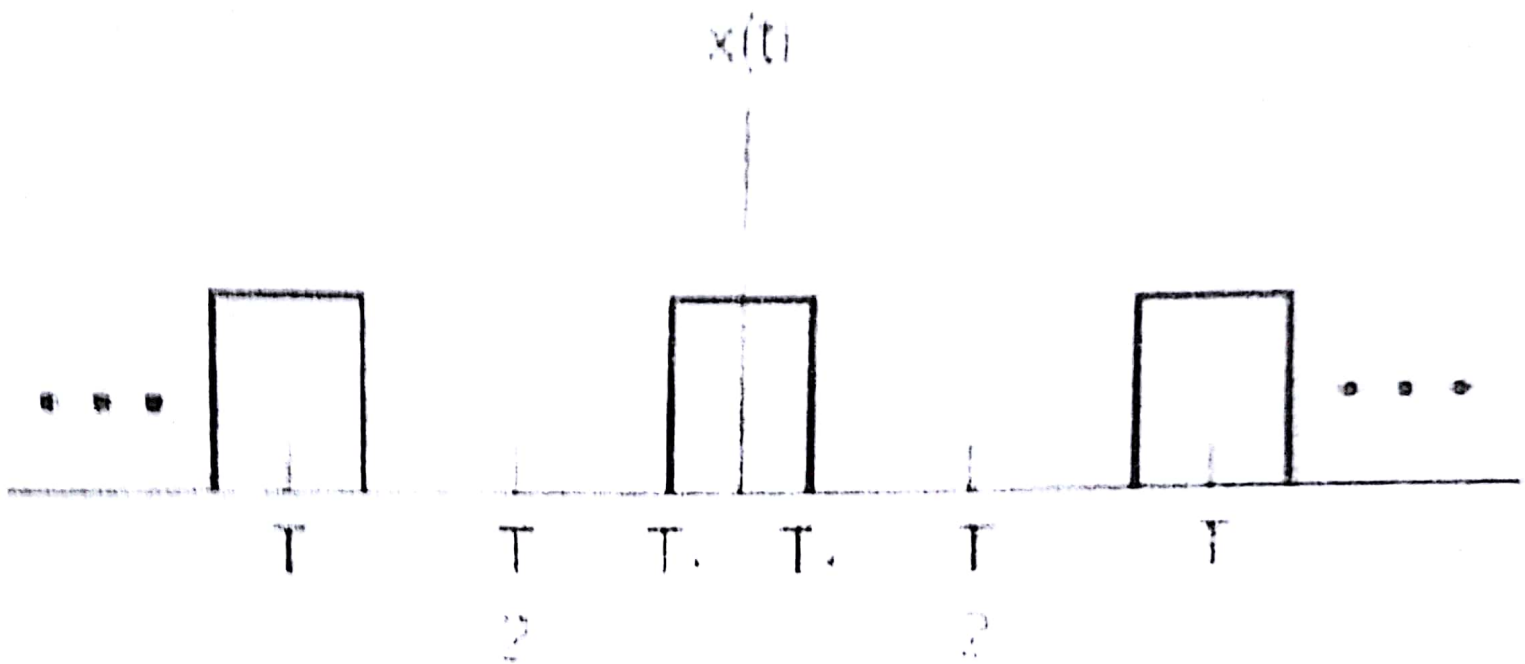
Example 1

- Compute the complex Fourier series coefficients for the first ten positive harmonic frequencies of the periodic signal $f(t)$ which has a period of 2π and defined as:

$$f(t) = 5e^{-t}, 0 \leq t \leq 2\pi$$

Example 2

- Plot the spectra of $x(t)$ if $T_1 = T/4$



Example 3

- Plot the spectra of $x(t)$.

$$x(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_0)$$

Classification of systems

- Linear and non-linear:

-linear :if system i/o satisfies the superposition principle. i.e.

$$F[ax_1(t) + bx_2(t)] = ay_1(t) + by_2(t)$$

where $y_1(t) = F[x_1(t)]$

and $y_2(t) = F[x_2(t)]$

Classification of sys. Cont.

- Time-shift invariant and time varying
-invariant: delay i/p by t_0 the o/p delayed by same amount. i.e

$$\text{if } y(t) = F[x(t)]$$

$$\text{then } y(t - t_0) = F[x_2(t - t_0)]$$

Classification of sys. Cont.

- Causal and non-causal system
- causal: if the o/p at $t=t_0$ only depends on the present and previous values of the i/p. i.e

$$y(t_0) = F[x(t), t \leq t_0]$$

LTI system is causal if its impulse response is causal.
i.e.

$$h(t) = 0, \forall t < 0$$

Suggested problems

- 2.1.1,2.1.2,2.1.4,2.1.8
- 2.3.1,2.3.3,2.3.4
- 2.4.2,2.4.3
- 2.7.1, 2.7.4, 2.7.5
- 2.8.1



EE325: Chapter 3

Analysis and Transmission of Signals

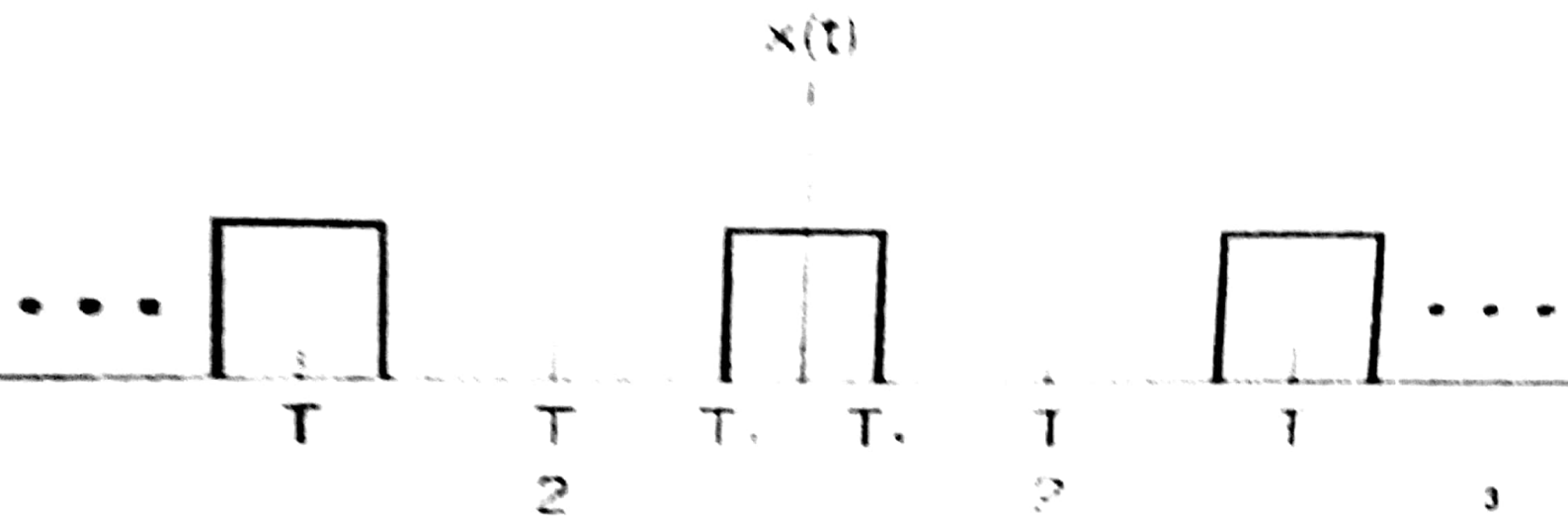
M. A. Smadi

Outline

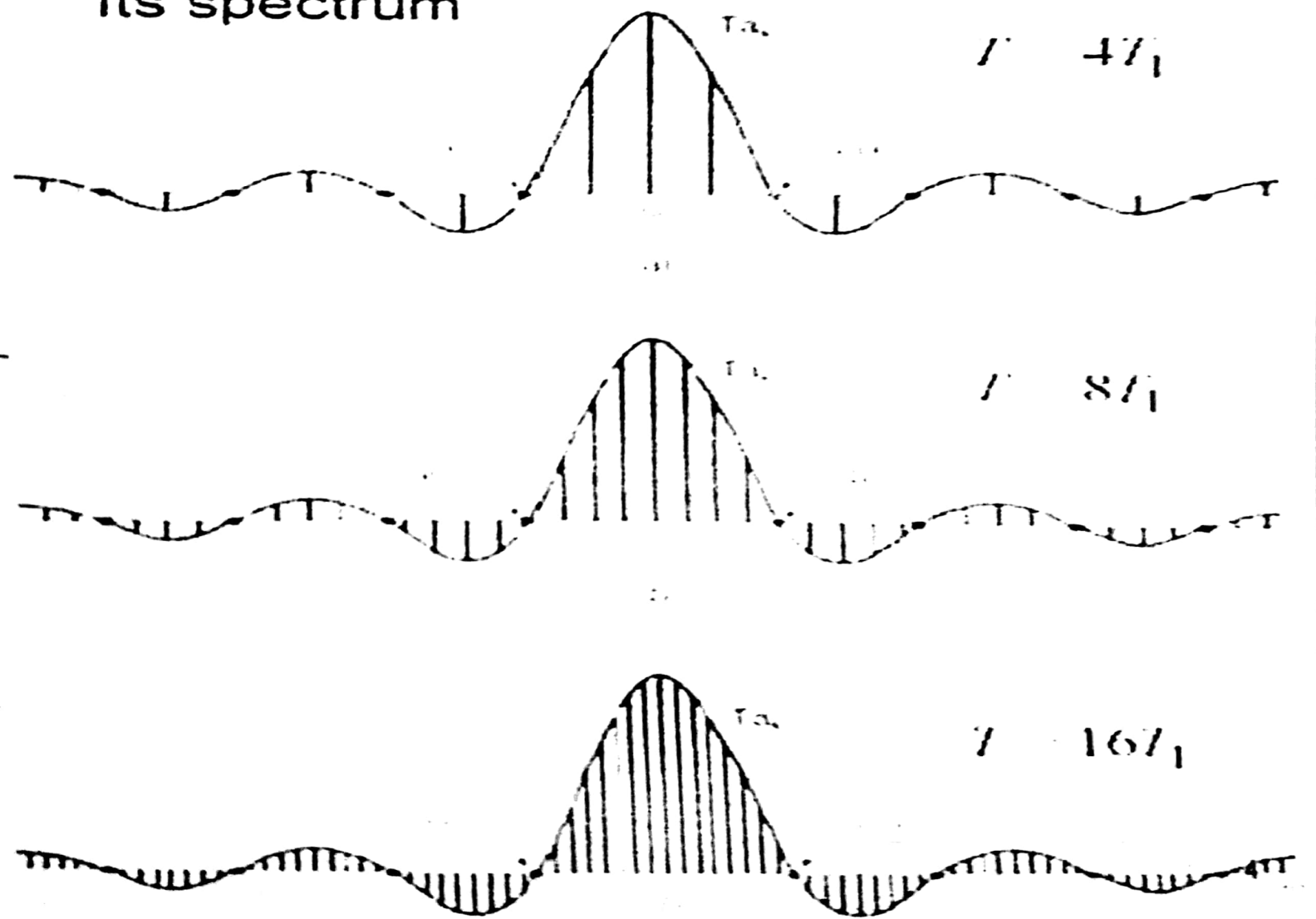
- Introduction
- Fourier transform and its inverse
- Fourier transform of some useful functions
- Properties of Fourier transform
- Transmission through LTI system
- Correlation functions and spectral densities.

Introduction

- Fourier series works for periodic signals only. What's about aperiodic signals? This is very large & important class of signals



Its spectrum



Introduction (cont.)

- Aperiodic signal can be considered as periodic for $T \rightarrow \infty$
- Fourier series changes to Fourier transform, complex exponents are infinitesimally close in frequency
- Discrete spectrum becomes a continuous one, also known as spectral density

Fourier Transform and Its Inverse

- **Fourier transform: if $g(t)$ is aperiodic signal then**

$$G(f) = \int_{-\infty}^{\infty} g(t) e^{-j2\pi f t} dt$$

$$g(t) = \int_{-\infty}^{\infty} G(f) e^{j2\pi f t} df$$

$$g(t) \Leftrightarrow G(f)$$

FT cont.

$$G(f) = |G(f)| e^{j\theta(f)}$$

- For real $g(t)$,

$$G(-f) = G^*(f) = |G(f)| e^{-j\theta(f)}$$

$|G(f)|$: ***amplitude spectrum***

$\theta(f)$: ***phase spectrum***

Example

- Find the FT and plot amplitude and phase spectra of:

1) $g(t) = e^{-at}u(t)$

2) $x(t) = e^{at}u(-t)$

Fourier Transform of Some Useful Functions

1) $\delta(t) \Leftrightarrow 1$

2) $1 \Leftrightarrow \delta(f)$
 $e^{j2\pi f_c t} \Leftrightarrow \delta(f - f_c)$

3) $e^{-j2\pi f_c t} \Leftrightarrow \delta(f + f_c)$

4) $\text{rect}(\frac{t}{T}) \Leftrightarrow T \text{sinc}(fT)$

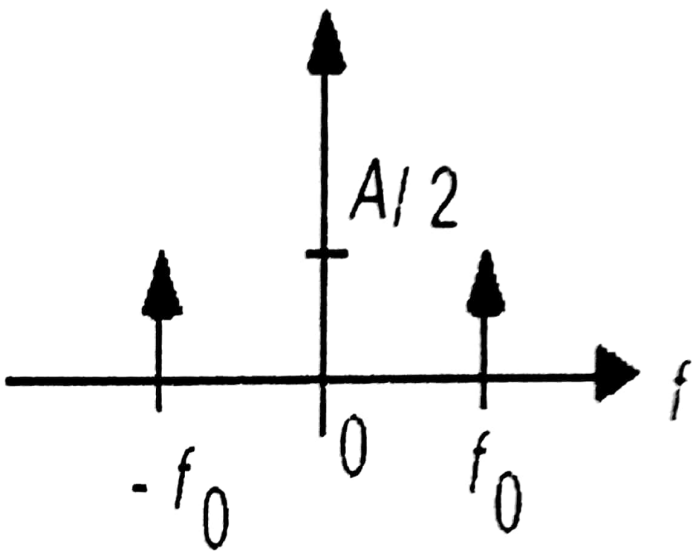
5) $\Delta(\frac{t}{T}) \Leftrightarrow T \text{sinc}^2(fT)$

6) $\cos(2\pi f_c t) \Leftrightarrow \frac{1}{2}[\delta(f - f_c) + \delta(f + f_c)]$
 $\sin(2\pi f_c t) \Leftrightarrow \frac{1}{2j}[\delta(f - f_c) - \delta(f + f_c)]$

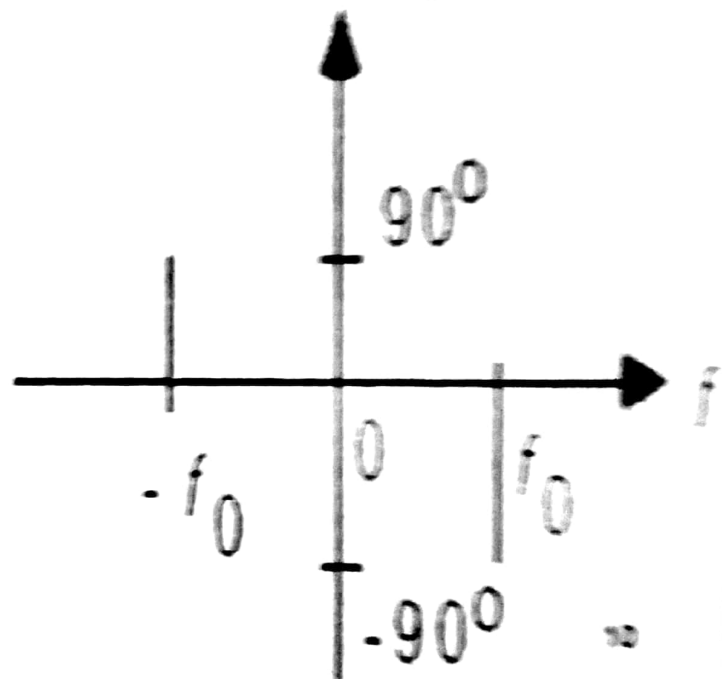
Spectra of

$$v(t) = A \sin 2\pi f_0 t$$

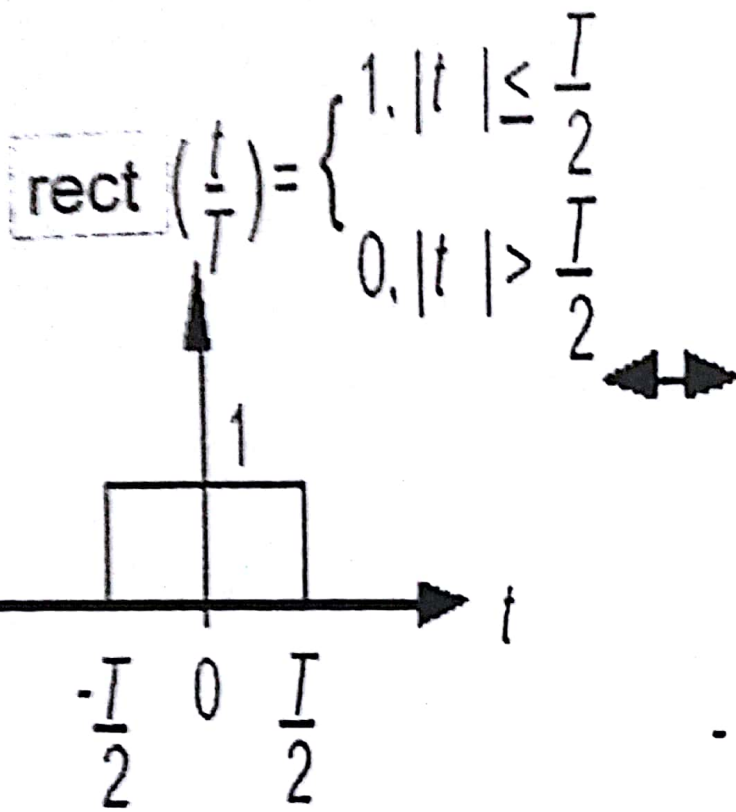
$|V(f)|$



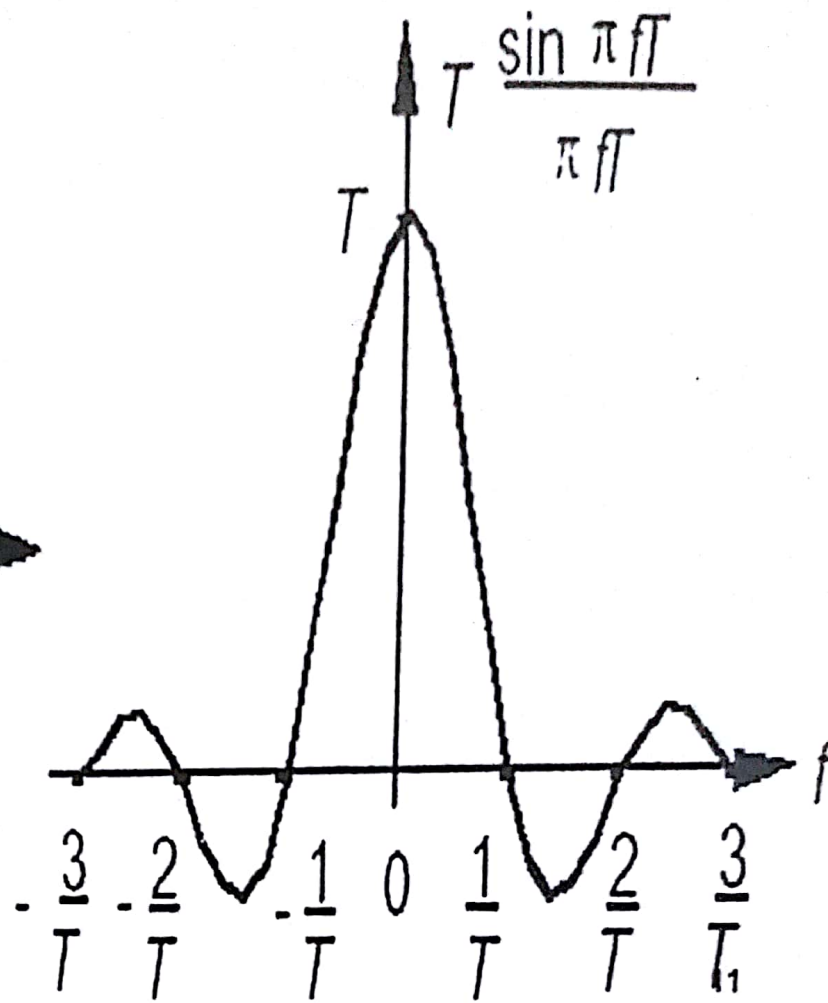
$\theta(f)$

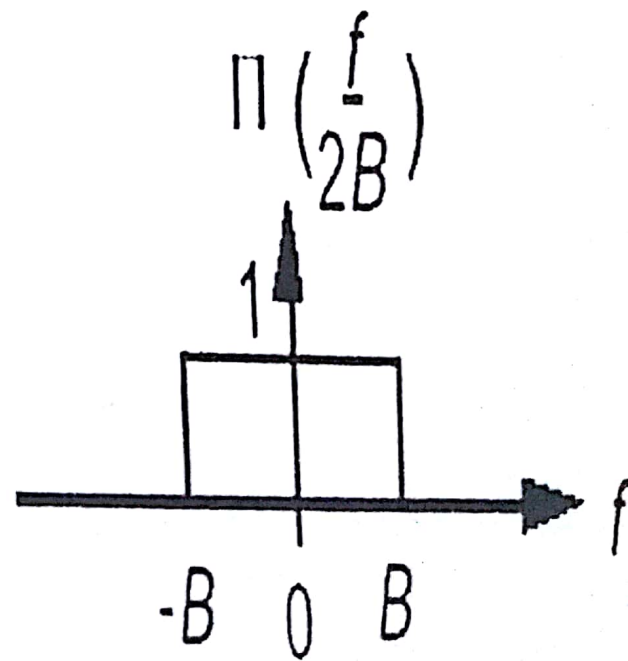
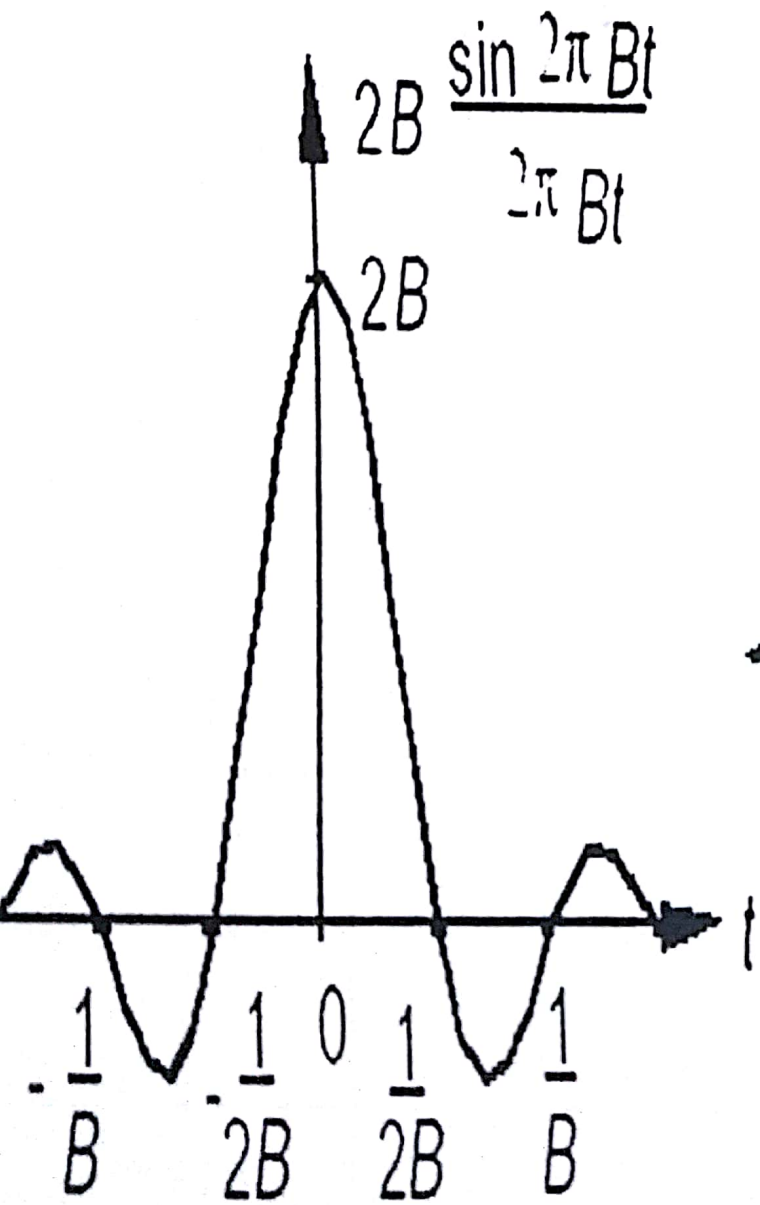


Time domain

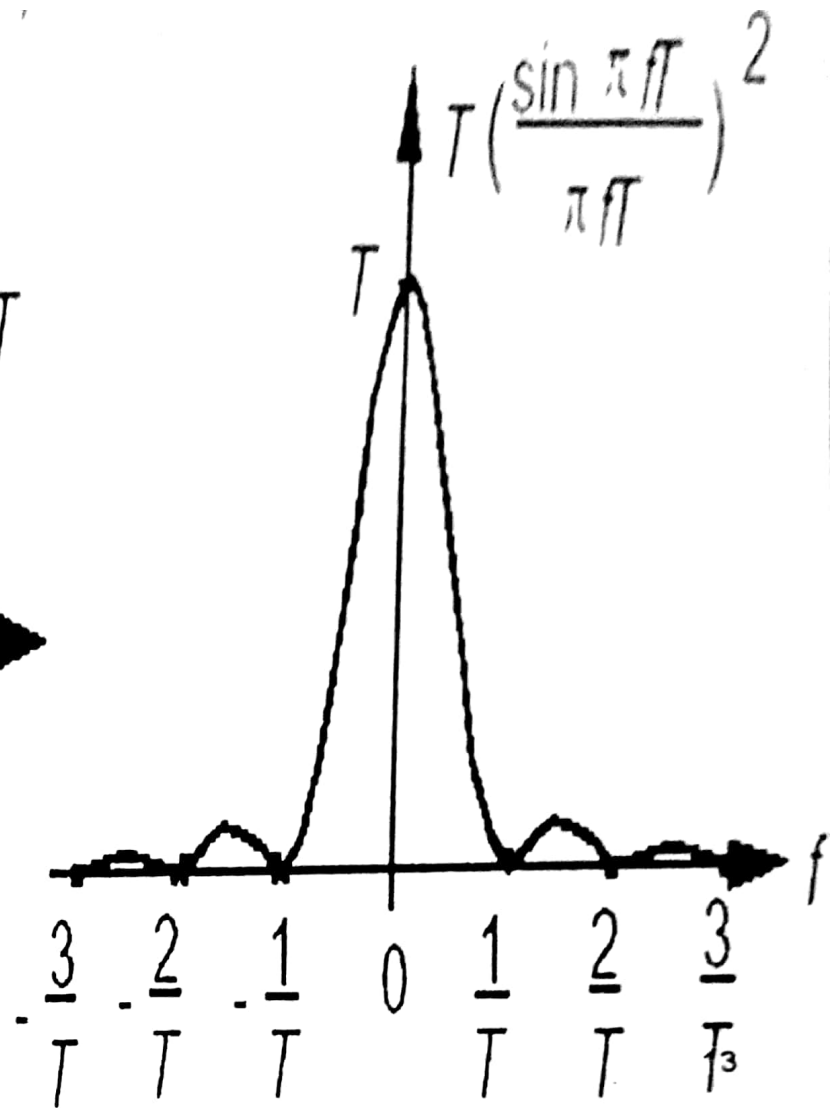
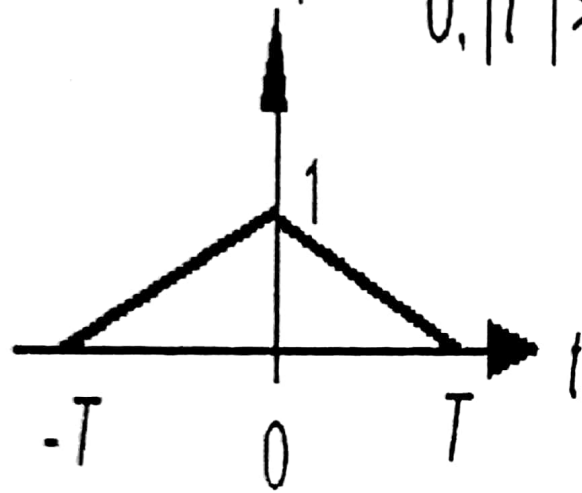


Frequency domain





$$\Lambda\left(\frac{t}{T}\right) = \begin{cases} 1 - \frac{|t|}{T}, & |t| \leq T \\ 0, & |t| > T \end{cases}$$



Properties of Fourier Transform

- Linearity:

$$a_1 x_1(t) + a_2 x_2(t) \leftrightarrow a_1 X_1(f) + a_2 X_2(f)$$

- Time shifting: $x(t - t_0) \leftrightarrow X(f) e^{-j 2\pi f t_0}$

- Time reversal: $x(-t) \leftrightarrow X(-f)$

- Time scaling: $x(at) \leftrightarrow \frac{1}{|a|} X\left(\frac{f}{a}\right)$

- Frequency shift (modulation):

$$x(t)e^{j2\pi f_c t} \leftrightarrow X(f - f_c)$$

$$x(t)e^{-j2\pi f_c t} \leftrightarrow X(f + f_c)$$

- Time differentiation: $\frac{d}{dt}x(t) \leftrightarrow j2\pi f X(f)$

- Time integration:

$$\int_{-\infty}^t x(\tau) d\tau \leftrightarrow \frac{1}{j2\pi f} X(f) + \frac{1}{2} X(0) \delta(f)$$

- Time Convolution:

$$x(t) * h(t) \leftrightarrow X(f) H(f)$$

$$x(t) * h(t) = \int_{-\infty}^{\infty} x(\tau) h(t - \tau) d\tau$$

- Time Multiplication (Frequency convolution)

$$x(t)y(t) \leftrightarrow X(f) * Y(f)$$

- Duality

If $x(t) \leftrightarrow X(f)$

then $X(t) \leftrightarrow x(-f)$

- Differentiation in frequency

$$(-j 2\pi t)x(t) \leftrightarrow \frac{dX(f)}{df}$$

Examples

- Use the Fourier transform properties to find the Fourier transform of the following:

1) $x(t) = e^{-|t|}$

2) $g(t) = \text{sinc}(2Bt)$

3) $y(t) = \text{rect}\left(\frac{t-T}{T}\right)$

4) $v(t) = e^{-t} \sin(2\pi f_c t) u(t)$

5) $x(t) = \text{sgn}(t)$

6) $g(t) = u(t)$

Fourier transform of periodic signal

- If $x_p(t)$ is periodic signal of period T_0 then

$$x_p(t) = \sum_{m=-\infty}^{\infty} x(t - mT_0) = \frac{1}{T_0} \sum_{m=-\infty}^{\infty} X(nf_0) e^{j2\pi nft}$$

Then the Fourier transform of $x_p(t)$ is

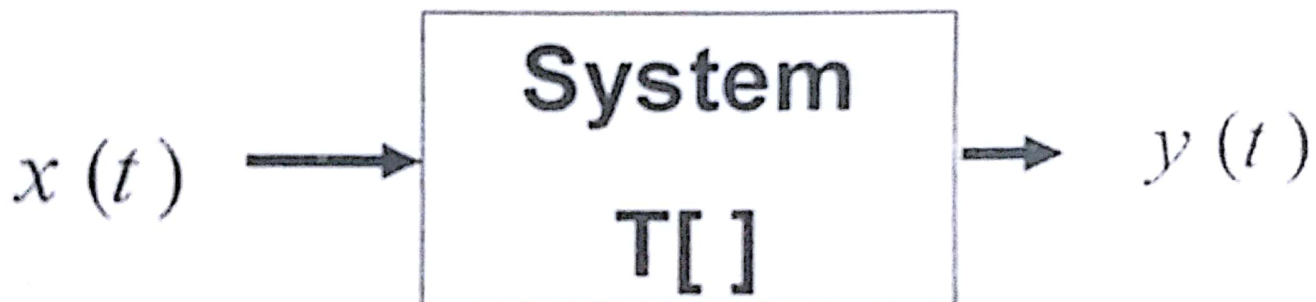
$$\sum_{m=-\infty}^{\infty} x(t - mT_0) \leftrightarrow \frac{1}{T_0} \sum_{n=-\infty}^{\infty} X(nf_0) \delta(f - nf_0)$$

Signal Transmission Through a Linear Time Invariant System

- System representation
- Impulse response and transfer function
- Distortionless transmission

System Representation

- A system is defined mathematically as a transformation or operator that maps an input $x(t)$ into an output $y(t)$.



Impulse Response of an LTI system

- The impulse response of an LTI system is defined as the response of the system when the input is $\delta(t)$. i.e

$$h(t) = y(t) \downarrow_{x(t)=\delta(t)}$$

- For any arbitrary input signal $x(t)$, the response

$$y(t) = x(t) * h(t) = \int_{-\infty}^{+\infty} x(\tau) h(t - \tau) d\tau$$

Frequency response of an LTI system

- The transfer function of an LTI system is

$$H(f) = \frac{y(t)}{x(t)} \downarrow_{x(t)=e^{j2\pi ft}}$$

$$H(f) = |H(f)| e^{j\theta(f)}$$

- The response to an input $x(t)$ is

$$Y(f) = X(f)H(f)$$

Signal Distortion during Transmission

- The transmission of an input signal $x(t)$ through a system changes it into the output signal $y(t)$.
- During transmission through the system, some frequency components may be boosted in amplitude while others may be attenuated.
- The relative phases of the various components also change due to different delays.

Distortionless Transmission

- Transmission is said to be distortionless if

$$y(t) = k x(t - t_d)$$

$$Y(f) = X(f)H(f) = kX(f)e^{-j2\pi f t_d}$$

$$\rightarrow H(f) = k e^{-j2\pi f t_d}$$

Dispersive channel

- Channel which adds distortion is dispersive channel.
 - Amplitude distortion: when $|H(f)| \neq k$, channel is a fading channel.
 - Phase distortion: when $\theta(f) \neq \alpha f$, channel is a jittering channel.

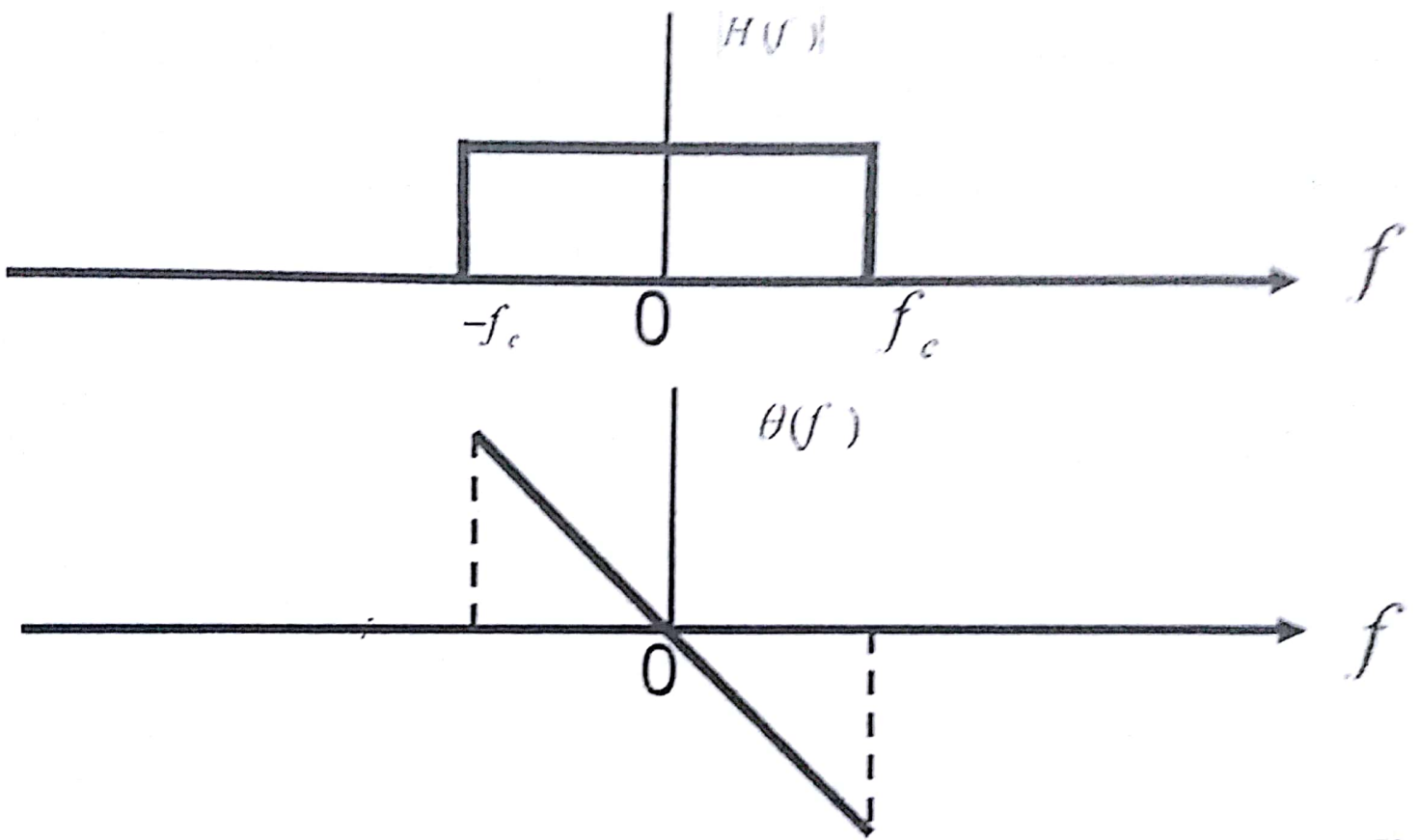
The Nature of Distortion in Audio and Video Signals

- The human ear can perceive amplitude distortion but it is relatively insensitive to phase distortion.
- The human eye is sensitive to phase distortion but is relatively insensitive to amplitude distortion.

Ideal and Practical Filters

- A filter is a system whose transfer function takes significant values only in certain frequency bands. Filter are usually classified as
 - Low-pass,
 - high-pass,
 - Band-pass, or
 - Band-stop

Ideal Low-Pass Filter



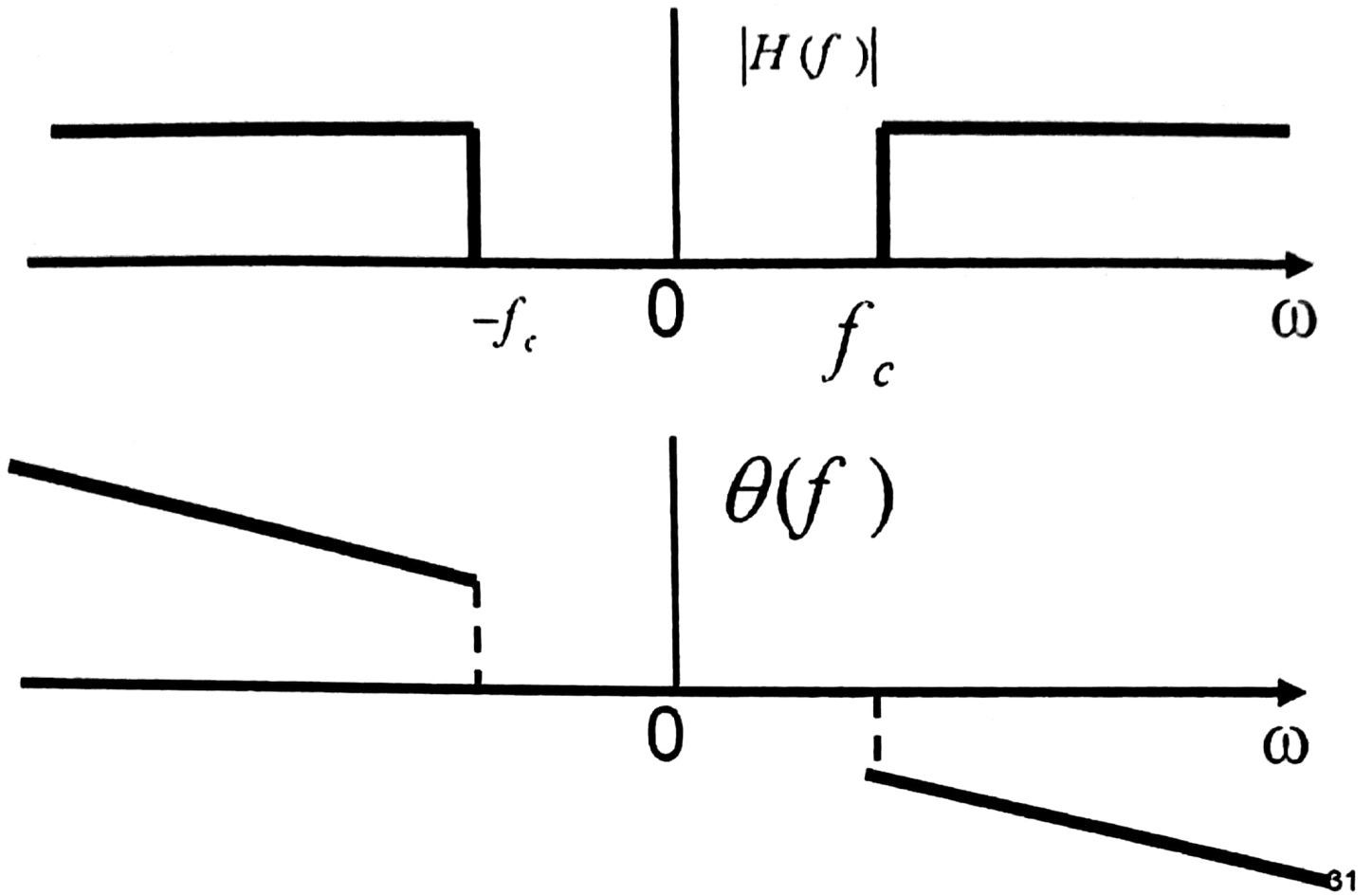
Transfer function of an ideal LPF

$$H_{LPF}(f) = \text{rect}\left(\frac{f}{2f_c}\right) e^{-j2\pi f t_d}$$

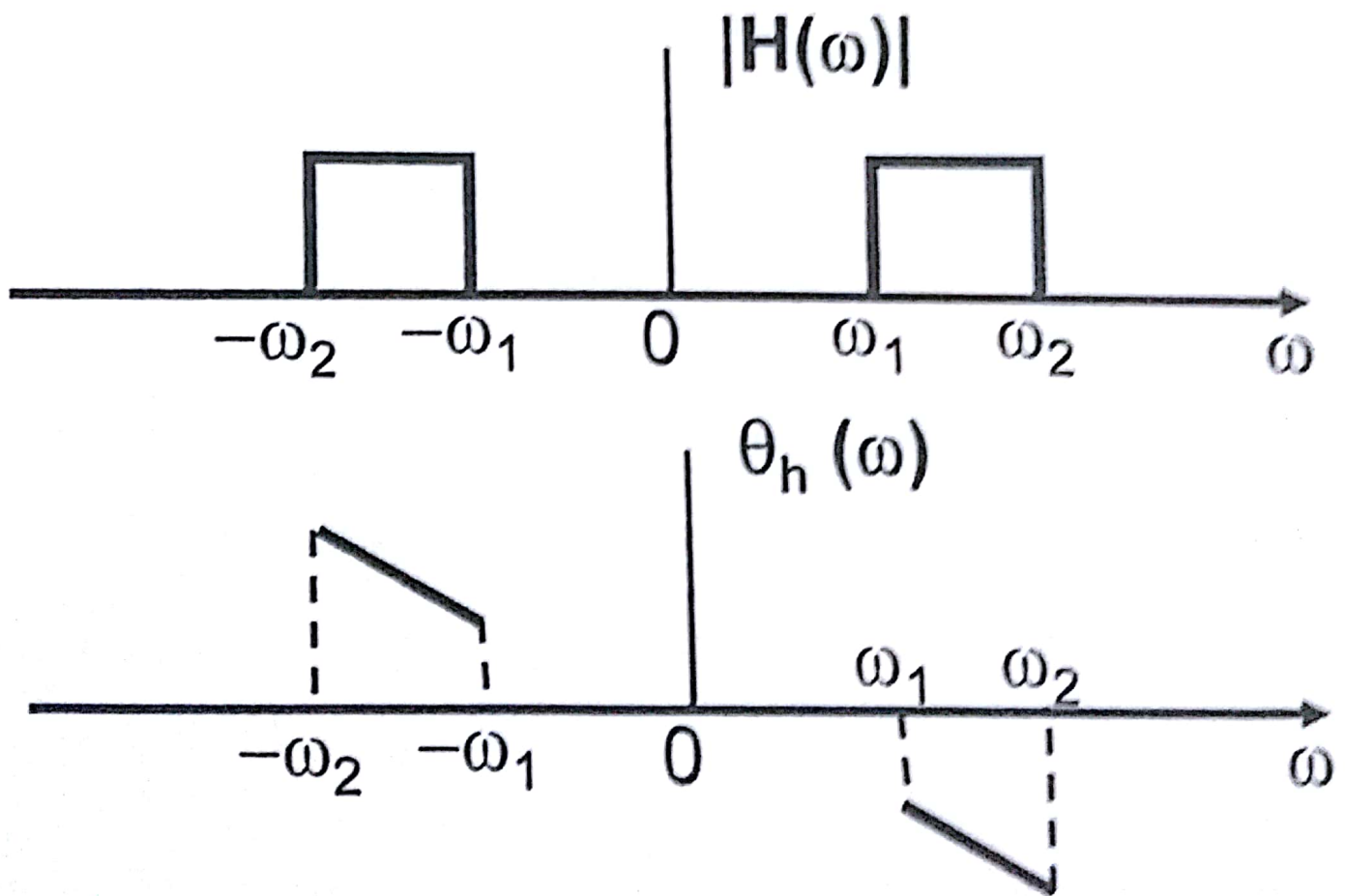
$$\rightarrow h(t) = 2f_c \text{sinc}(2f_c(t - t_d))$$

→ unrealizable

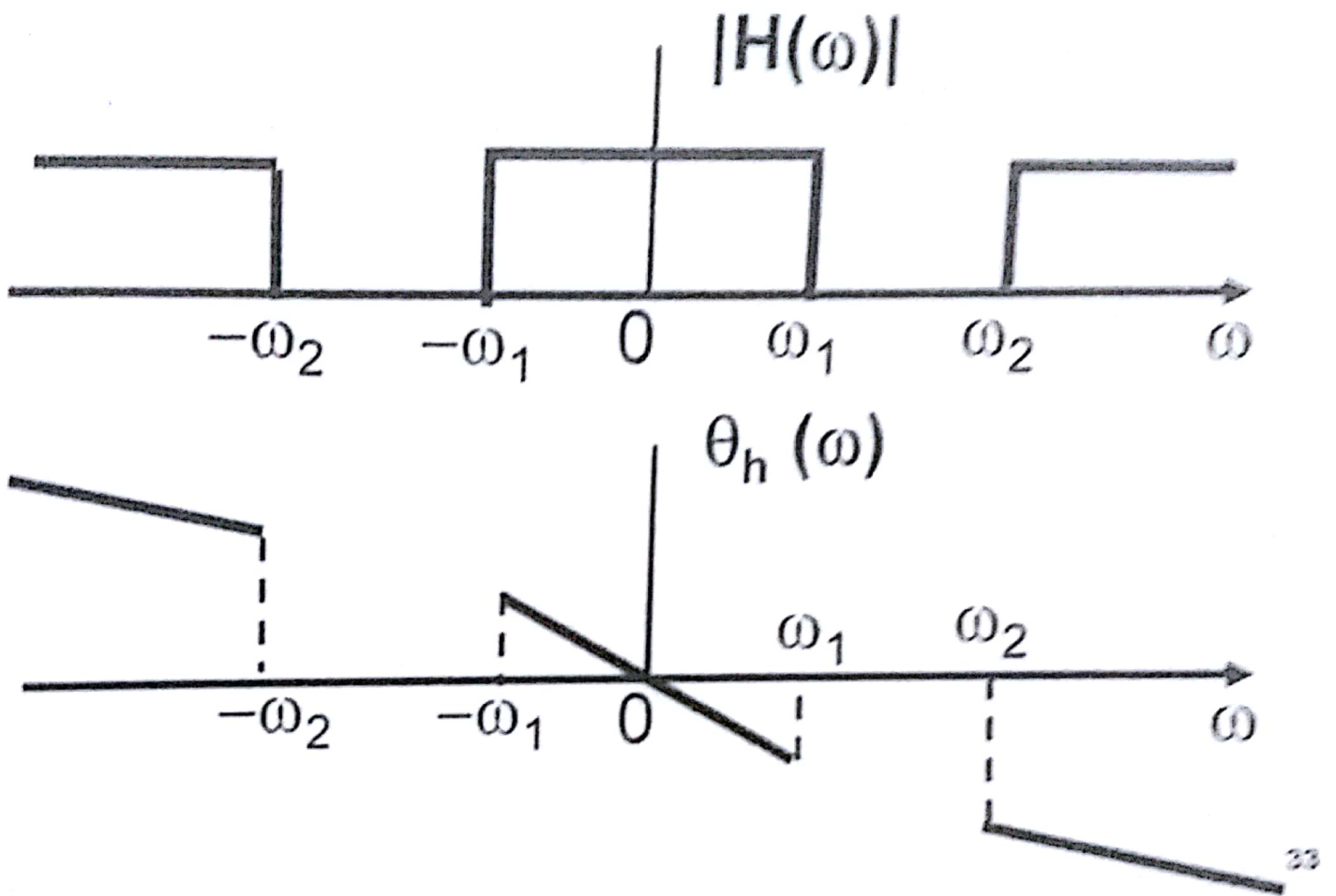
Ideal High-Pass Filter



Ideal Band-Pass Filter



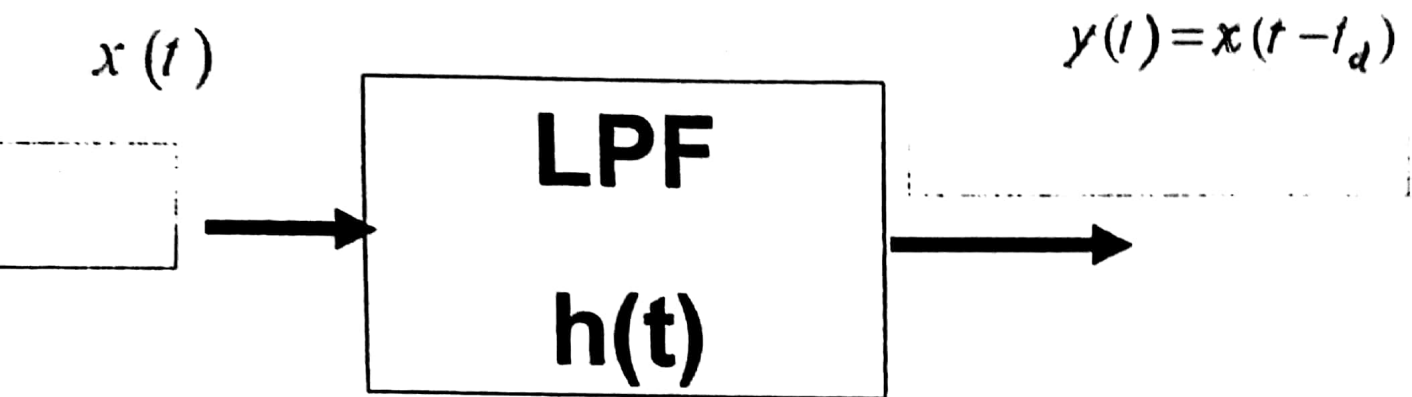
Ideal Band-Stop Filter



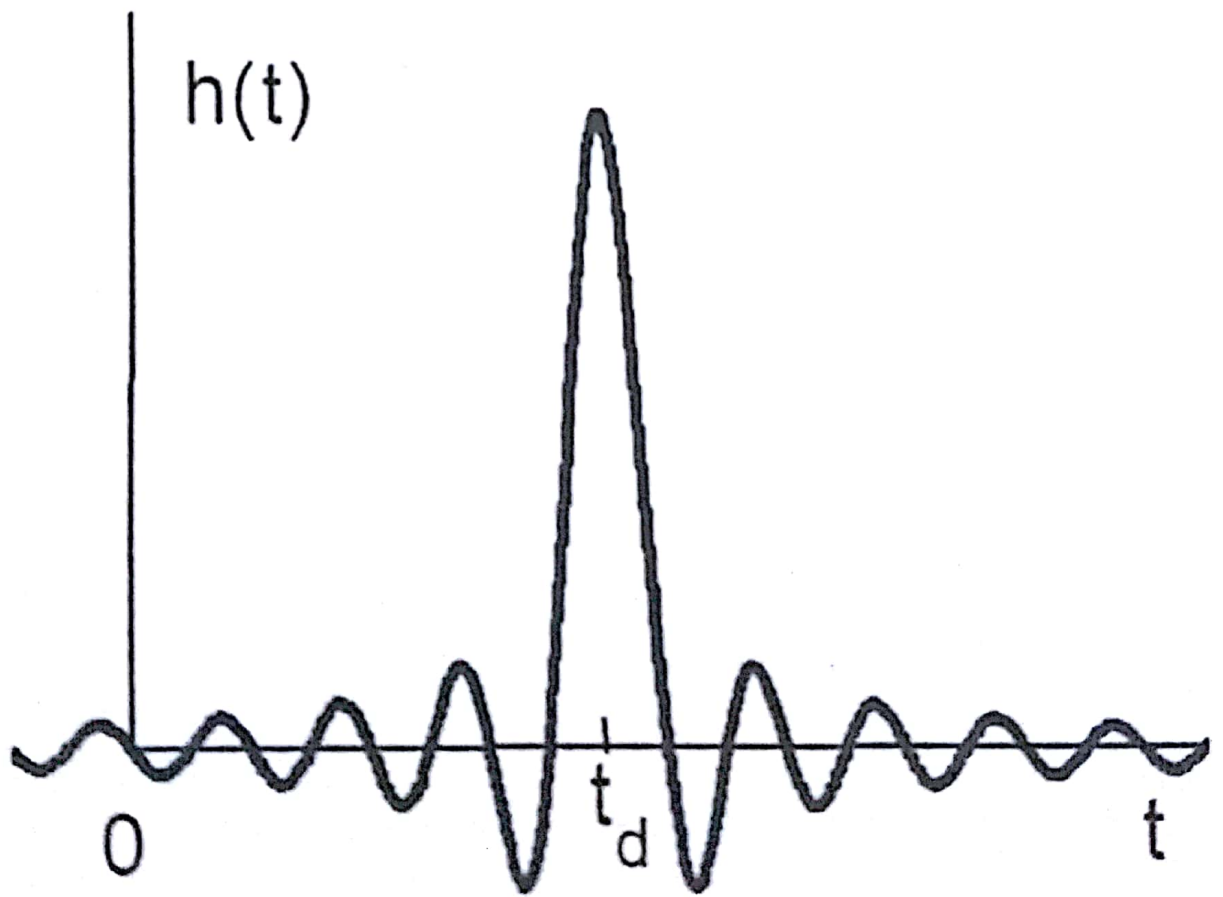
$$H_{HPF}(f) = \begin{cases} e^{-j2\pi f t_d}, & \text{for } |f| \geq f_c \\ 0, & \text{otherwise} \end{cases}$$

$$H_{BPF}(f) = \begin{cases} e^{-j2\pi f t_d}, & \text{for } f_1 \leq |f| \leq f_2 \\ 0, & \text{otherwise} \end{cases}$$

Ideal versus Practical Low-Pass Filter



$$h(t) = 2f_c \operatorname{sinc} [2f_c (t - t_d)]$$



- For a physical realizable system, $h(t)$ must be causal; that is,

$$h(t) = 0, \forall t < 0$$

- One practical approach is to cut off the tail of $h(t)$ for $t < 0$

$$\hat{h}(t) = h(t) u(t)$$

- If t_d is sufficient large

$$\hat{h}(t) \approx h(t)$$

Filter or System Bandwidth

- The bandwidth of an ideal low-pass filter

$$Bw = f_c$$

- The bandwidth of an ideal band-pass filter

$$Bw = f_h - f_l = f_2 - f_1$$

- No bandwidth for high-pass and band-stop filters.
- For practical filters, a common definition of filter bandwidth is the 3-dB bandwidth.

Signal Bandwidth

- The bandwidth of a signal can be defined as the range of frequencies in which most of the energy or power lies.
- It can also be defined in terms of the 3-dB bandwidth.
- The signal bandwidth is also called the essential bandwidth of the signal

Signal Energy and Energy Spectral Density

- The signal energy can be determined from its Fourier transform using Parseval's theorem

$$E_g = \int_{-\infty}^{+\infty} |g(t)|^2 dt = \int_{-\infty}^{\infty} |G(f)|^2 df$$

Example

- Verify Parseval's theorem for the signal

$$1) \quad g(t) = e^{-\alpha} u(t); \quad G(f) = \frac{1}{j2\pi f + \alpha} \Rightarrow E_g = 1/2\alpha$$

$$2) \quad x(t) = \text{sinc}(2\omega t)$$

Energy Spectral Density (ESD)

$$E_g = \int_{-\infty}^{\infty} \Psi_g(f) df$$

$$\text{where } \Psi_g(f) = |G(f)|^2$$

$\Psi_g(f)$ is called the **energy spectral density (ESD)**

- For previous example

$$\Psi_g(f) = \frac{1}{(2\pi f)^2 + a^2}$$

Example

- Estimate the essential bandwidth B of the signal

$$g(t) = e^{-at} u(t)$$

if the essential bandwidth is required to contain 95% of the signal energy.

$$\frac{0.95}{2a} = \int_{-F}^F \frac{1}{(2\pi f)^2 + a^2} df \Rightarrow B = 2.02a \text{ Hz}$$
$$= 12.7a \text{ rad/s}$$

Correlation of Energy Signals

- There are applications where it is necessary to compare one reference signal with one or more signals to determine the similarity between the pair from which some information will be extracted.
- This comparison can be done by computing the *correlation* between these signals.

Cross-correlation

- A measure of similarity between a pair of energy signals, $x(t)$ and $y(t)$ is given by the ***cross-correlation*** function expressed as

$$\psi_{xy}(\tau) = \int_{-\infty}^{\infty} x(t) y(t - \tau) dt$$

Cross-correlation cont.

- If we wish to make $y(t)$ the reference signal, then the corresponding cross-correlation function is given by

$$\psi_{yx}(\tau) = \int_{-\infty}^{\infty} y(t) x(t - \tau) dt$$

Autocorrelation function

- In the special case where $y(t) = x(t)$, we have the *autocorrelation* of $x(t)$ which is defined as

$$\psi_x(\tau) = \int_{-\infty}^{\infty} x(t) x(t - \tau) dt$$

Properties of Crosscorrelation and Autocorrelation functions

$$\psi_{xy}(\tau) = \psi_{yx}(-\tau)$$

$$\psi_x(\tau) = \psi_x(-\tau)$$

$$\psi_x(0) = E_x$$

- Autocorrelation function and the energy spectral density

$$\psi_x(t) \leftrightarrow |X(f)|^2 = \Psi_x(f)$$

- ESD of the Input and the Output

$$\Psi_y(f) = |H(f)|^2 \Psi_x(f)$$

Signal Power and Power Spectral Density

- For a real power signal $g(t)$

$$P_g = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{+T/2} g^2(t) dt$$

- The time-averaged autocorrelation function of $g(t)$ is defined as

$$\mathfrak{R}_g(\tau) = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{+T/2} g(t) g(t - \tau) dt$$

Autocorrelation of periodic signal

- If $g(t)$ is periodic with period T

$$\mathcal{R}_g(\tau) = \frac{1}{T} \int_{-T/2}^{+T/2} g(t) g(t - \tau) dt$$

- The power spectral density (PSD) of $g(t)$, $S_g(f)$, is the Fourier transform of $\mathfrak{R}_g(\tau)$

$$S_g(f) = \int_{-\infty}^{\infty} \mathfrak{R}_g(\tau) e^{-j2\pi f \tau} d\tau$$

$$\mathfrak{R}_g(\tau) = \int_{-\infty}^{\infty} S_g(f) e^{j2\pi f \tau} df$$

$$\mathfrak{R}_g(0) = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{+T/2} g^2(t) dt = \int_{-\infty}^{\infty} S_g(f) df = P_g$$

PSD cont.

- Input and output spectral densities

$$S_y(f) = |H(f)|^2 S_x(f)$$

Example

- Find the autocorrelation function and ESD of

$$x(t) = e^{-at} u(t)$$

$$\begin{aligned}\Psi_x(\tau) &= \int_{-\infty}^{\infty} e^{-at} u(t) e^{-a(t-\tau)} u(t-\tau) dt \\ &= e^{a\tau} \int_{\tau}^{\infty} e^{-2at} dt = \frac{1}{2a} e^{-a|\tau|}\end{aligned}$$

- Hence

$$\Psi_x(f) = \frac{1}{(2\pi f)^2 + a^2} = |X(f)|^2 !$$

Suggested problems

- 3.1.3, 3.1.5, 3.1.7
- 3.2.3, 3.2.5
- 3.3.1, 3.3.2, 3.3.6
- 3.4.1
- 3.5.3, 3.5.4
- 3.6.1
- 3.7.4, 3.7.5
- 3.8.1, 3.8.4



EE325: Chapter 4 (Lec. #1)

Amplitude Modulations & Demodulations

M. A. Smadi

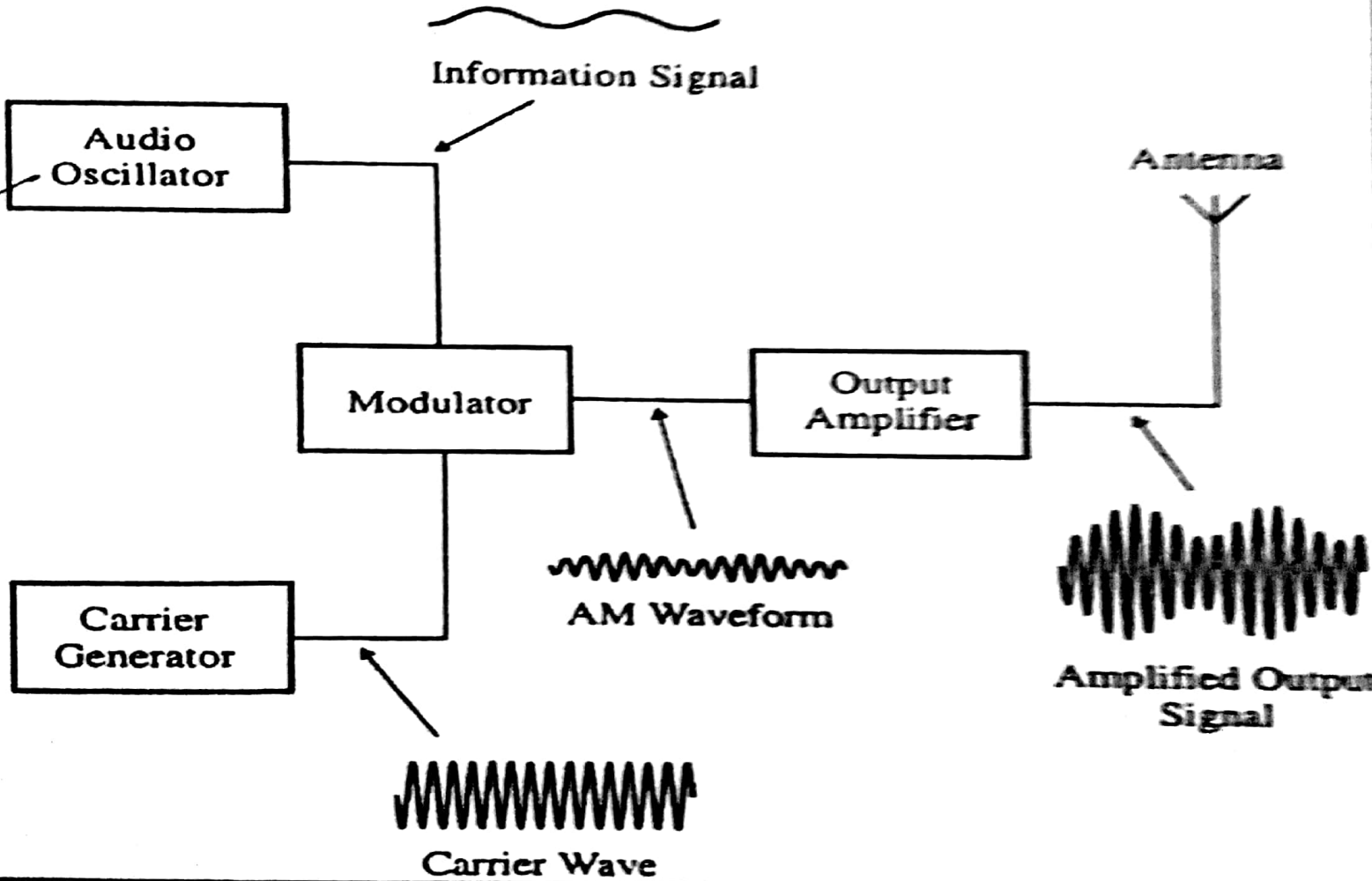
Outlines

- Introduction
- Base-band and Carrier Communication
- Amplitude Modulation (AM):DSB-Large Carrier
- Amplitude Modulation: Double sideband- Suppressed Carrier (DSBSC)
- Quadrature amplitude Modulation (QAM)
- Single Sideband Modulation (SSB)
- Vestigial Sideband (VSB)
- Frequency mixing
- Superhetrodyne AM radio.
- Frequency division multiplexing (FDM).

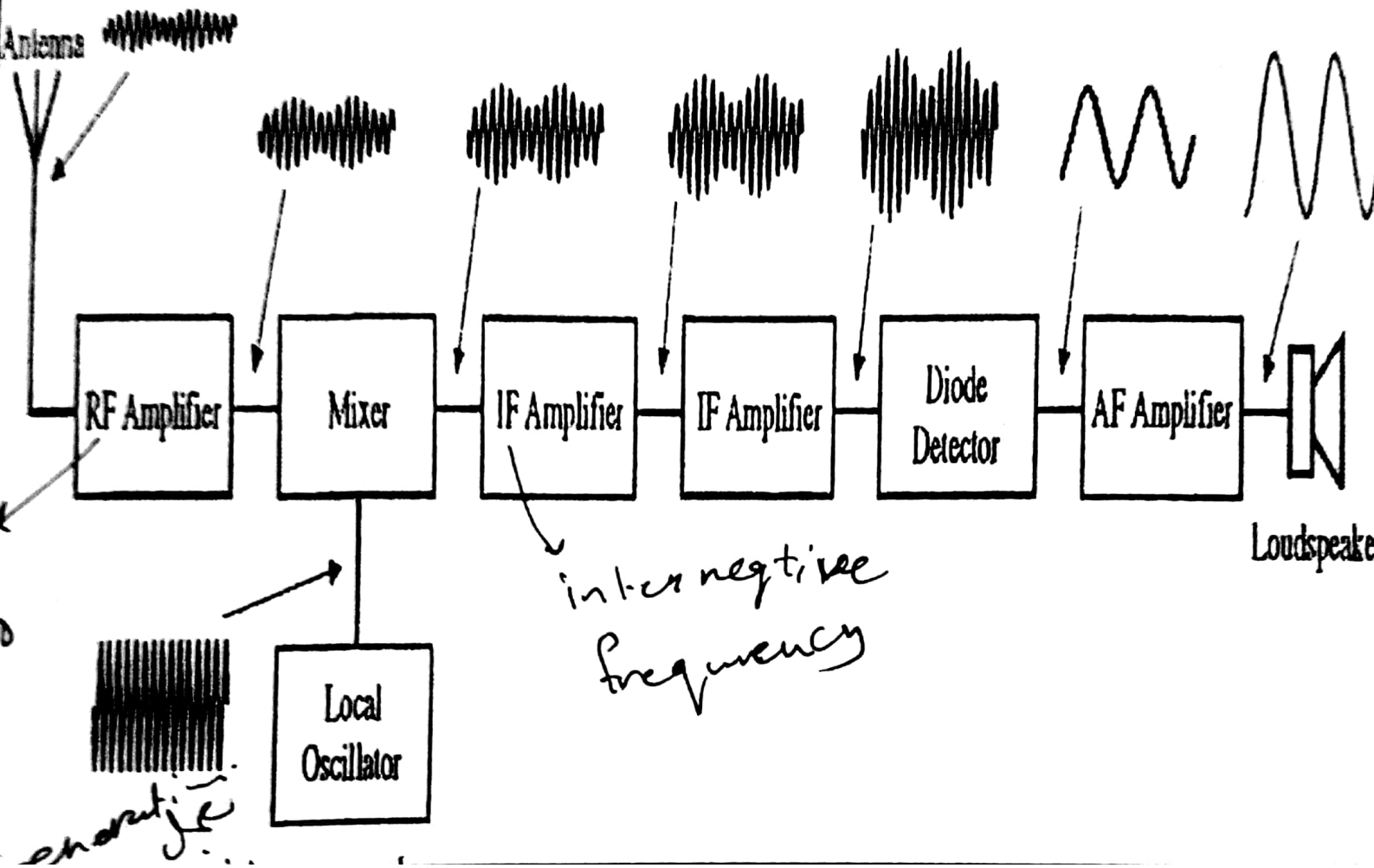
Introduction

- Modulation is a process that causes a shift in the range of frequencies of a message signal.
- A communication that does not use modulation is called **baseband communication**
- A communication that uses modulation is called **carrier communication**

Example of AM transmitter



Example of AM (radio) Receiver



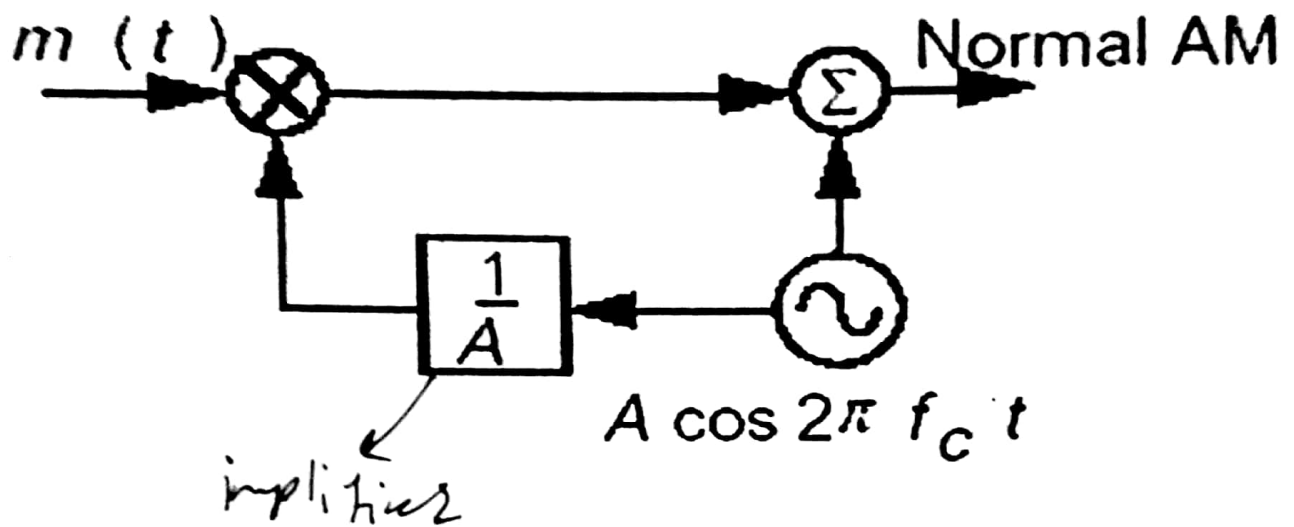
Baseband and Carrier Communication

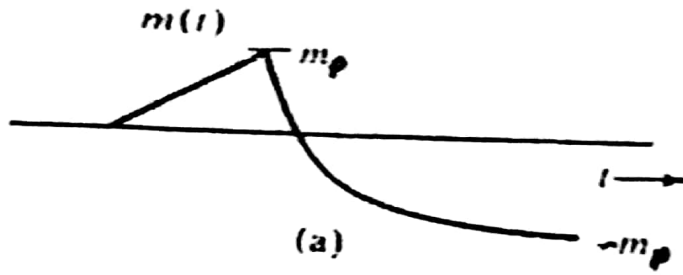
- Baseband Com.: is message signal (information signal) delivered by the information source. It is usually low frequency signal.
- Communication that uses modulation to shift the frequency spectrum of message signal is known as carrier communication.
 - Amplitude modulation (AM)
 - Frequency modulation (FM)
 - Phase modulation (PM)

Amplitude Modulation (AM)

← Double Sideband Large Carrier (DSB-LC)

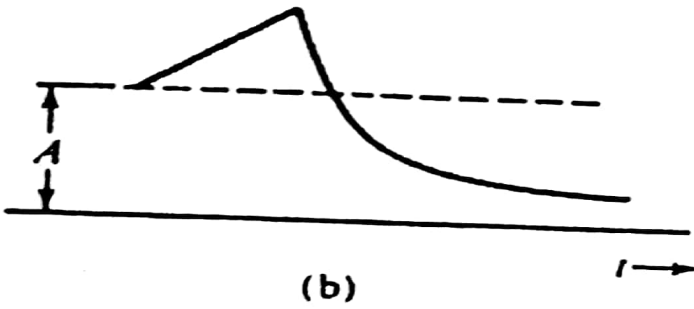
$$s_{AM}(t) = [m(t) + \underline{A}] \cos 2\pi f_c t$$



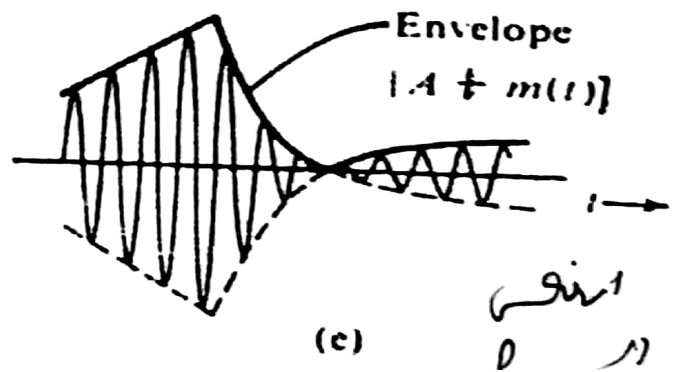
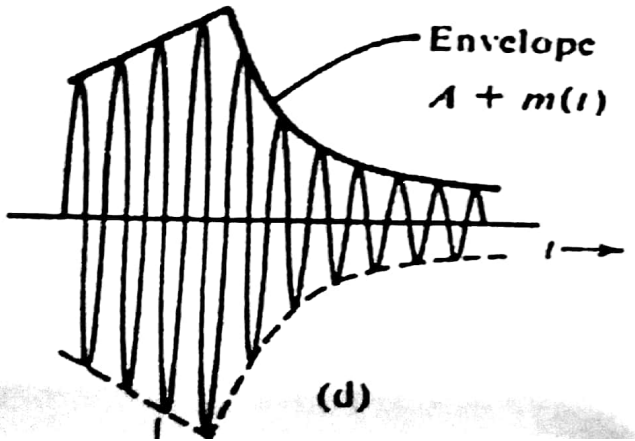
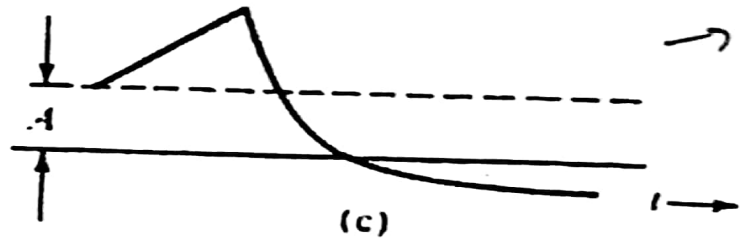


Sym.

$A + m(t) > 0$ for all t

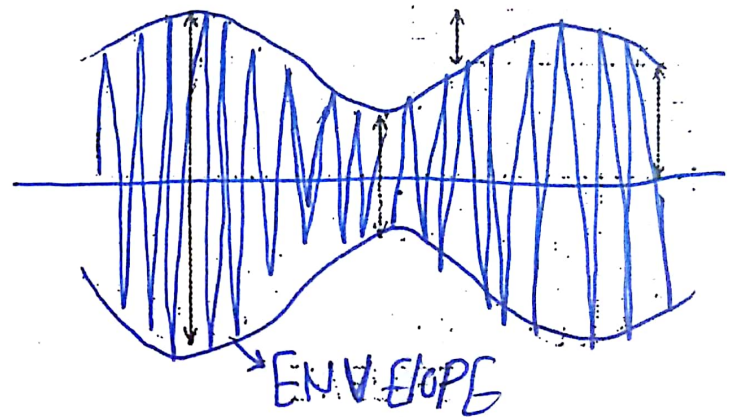
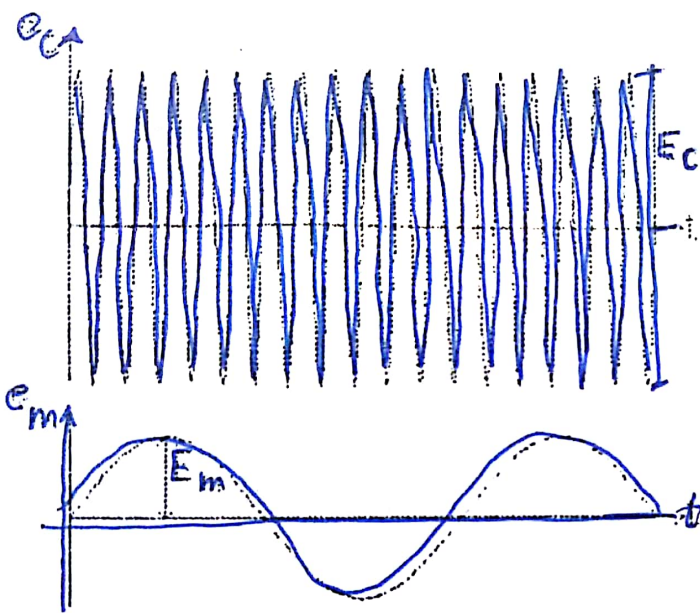


$A + m(t) > 0$ for all t



0 1

Another example of AM Waveform (single tone modulation)



$$c(t) = E_c \sin 2\pi f_c t$$

$$m(t) = E_m \sin 2\pi f_m t$$

$$s(t) = [E_c + m(t)] \sin 2\pi f_c t$$

Modulation Index

- The amount of modulation in AM signal is given by its modulation index:

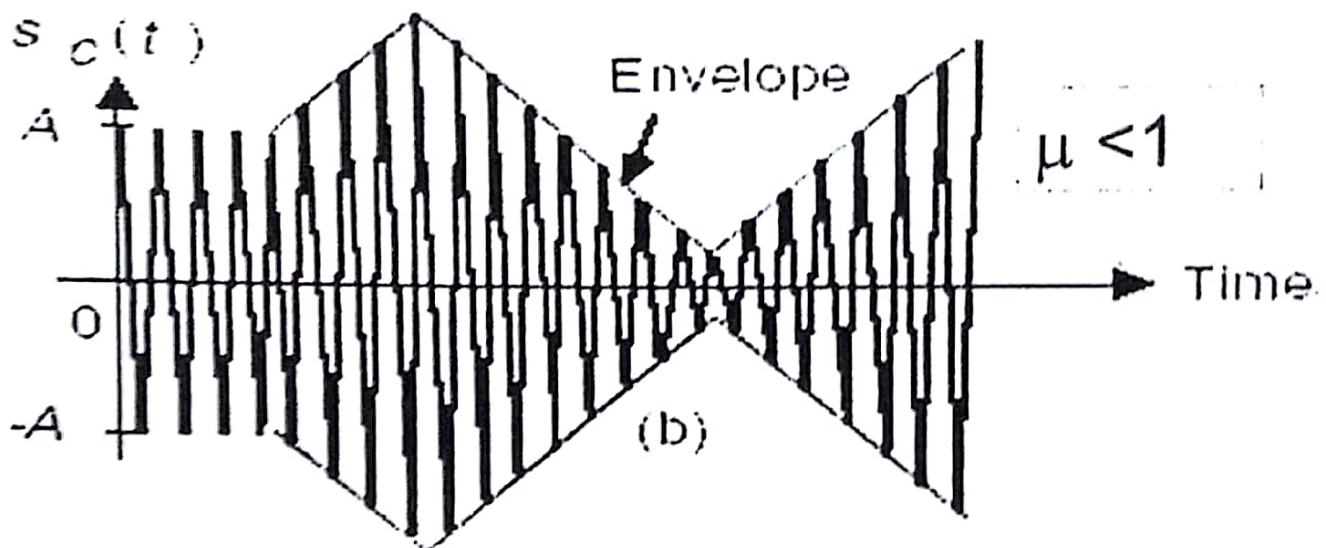
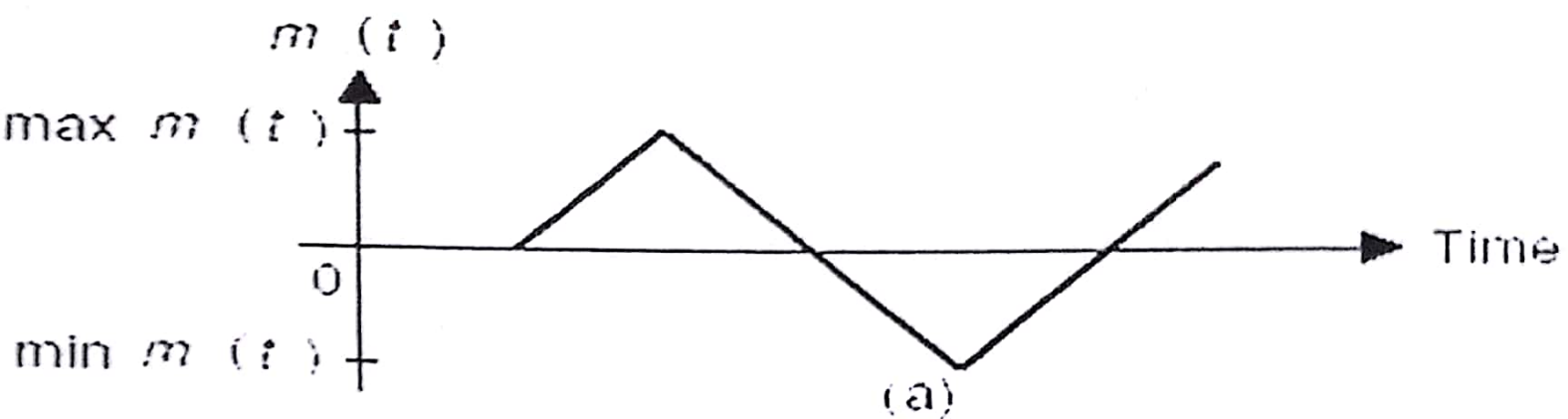
$$\mu = \frac{m_p}{A} = \frac{E_{\max} - E_{\min}}{E_{\max} + E_{\min}}, \quad m_p = \min |m(t)|$$

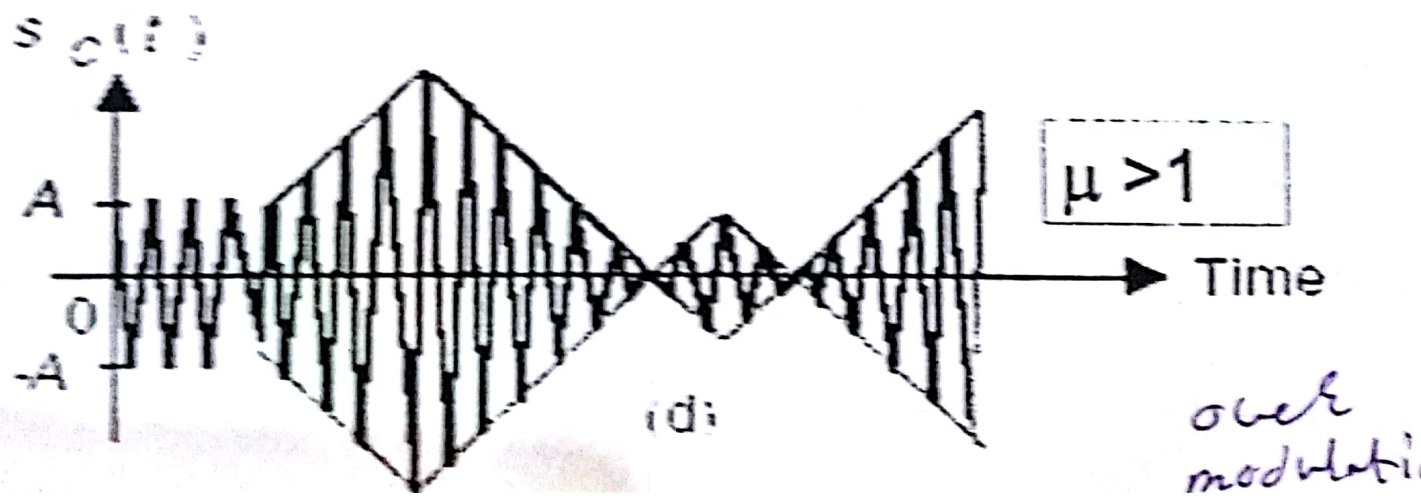
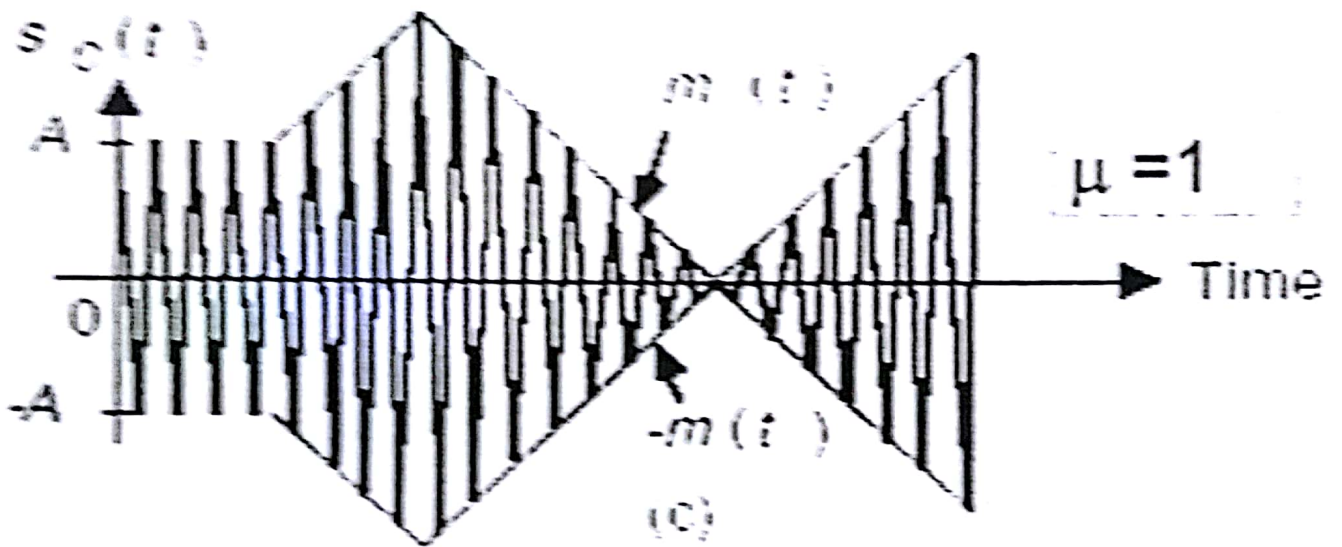
$$E_{\max} = A + m_p, \quad E_{\min} = A - m_p$$

When $m_p = A$, $\mu = 1$ or 100% modulation.

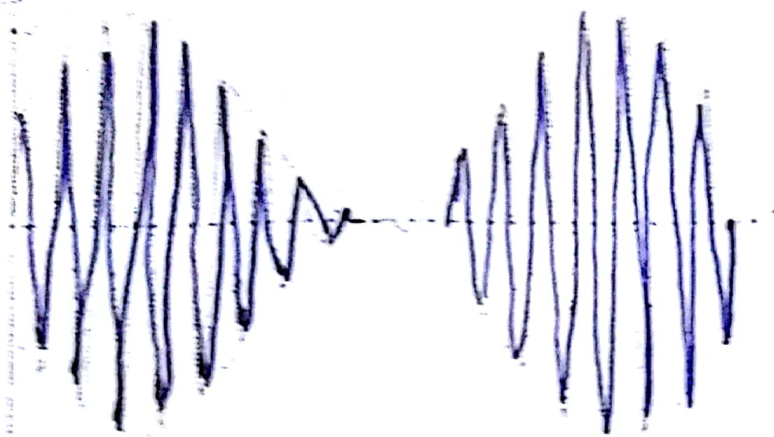
Over-modulation, i.e., $m_p > A$ ($\mu > 1$), should be avoided because it will create distortions.

Effect of Modulation Index

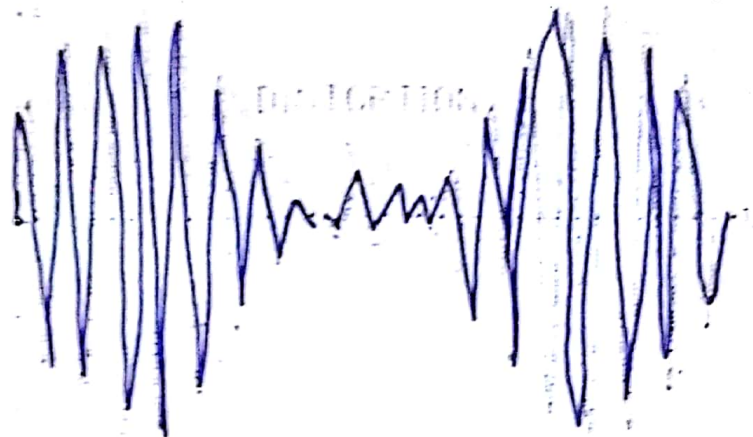




Effects of Modulation Index



$\mu = 1$



$\mu > 1$

over mod

Sideband and Carrier Power

$$s_{AM}(t) = [m(t) + A] \cos 2\pi f_c t$$

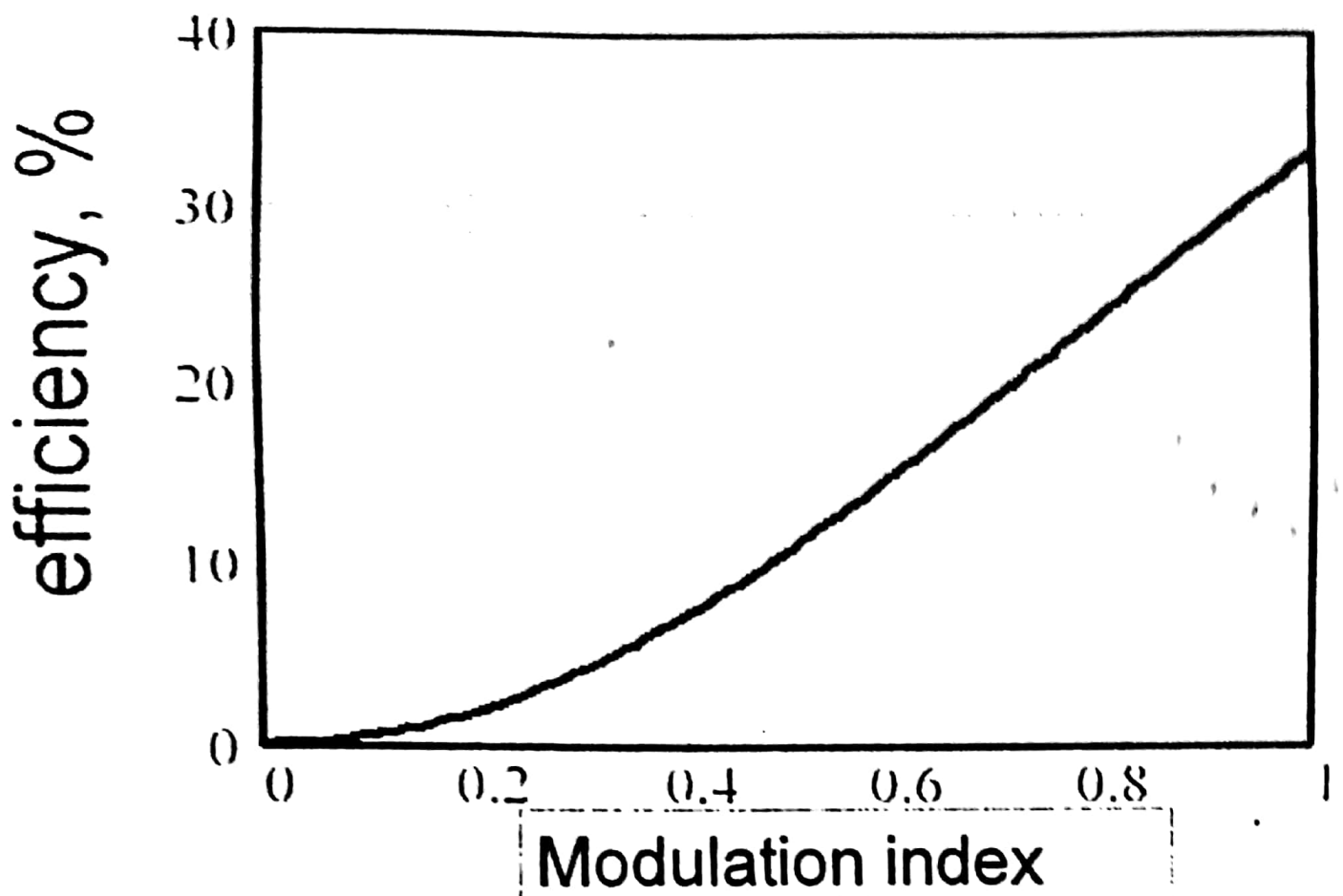
- Carrier Power: $P_c = \frac{A^2}{2}$
- Sideband Power: $P_s = \frac{P_m}{2}$
- Total power: $P_{tot} = P_c + P_s$

- Power efficiency: $\eta = \frac{P_s}{P_c + P_s} = \frac{P_m}{A^2 + P_m}$

- For single tone modulation; $m(t) = \mu A \cos(2\pi f_m t)$

$$P_m = \frac{(\mu A)^2}{2} \Rightarrow \eta = \frac{\mu^2}{2 + \mu^2}$$

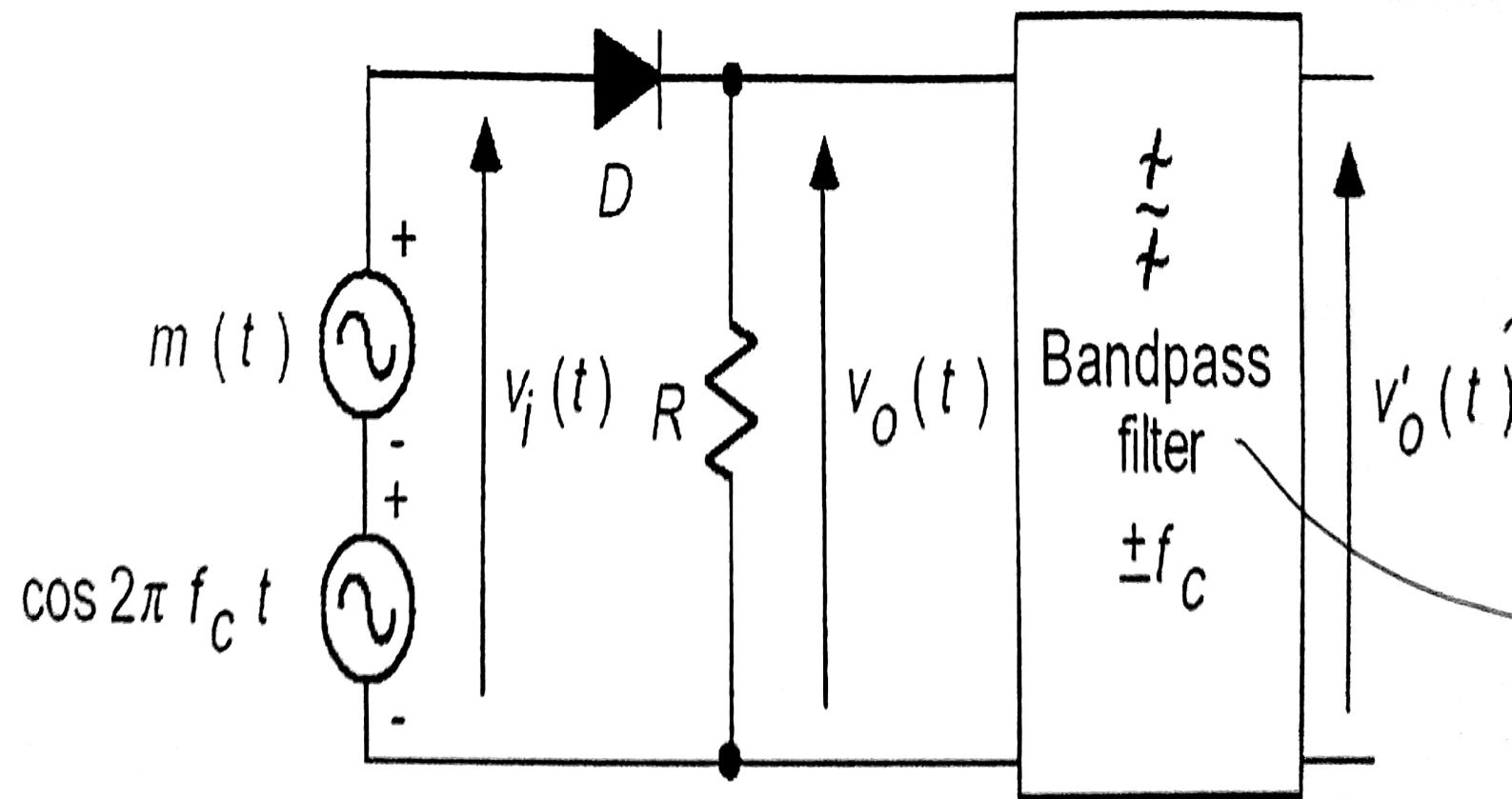
Power efficiency of AM



Example

- Conventional AM signal with a sinusoidal message has the following parameters:
 $A=10$, $\mu=0.5$, $f_c=1\text{MHz}$, and $f_m=1\text{kHz}$
1. Find time-domain expression $s_{AM}(t)$
 2. Find its Fourier transform
 3. Sketch its spectrum
 4. Find the signal power, carrier power and the power efficiency
 5. Find the AM signal bandwidth

Generation of AM Signals diode as NLE or as switch



Square-law modulator

$$v_o(t) = av_i(t) + bv_i^2(t)$$

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مربعی

$$v_o(t) = [aA + 2Abm(t)] \cos 2\pi f_c t$$

DSB with

$f_c \geq 3B$ B: $m(t)$ BW; To avoid overlap the spectrum of

$m^2(t)$ and $M(f - f_c)$

$m(t)$
BW = ?

Switching modulator

- Assume $|m(t)| \leq A$, and diode an ideal switch

$$s_{AM}(t) = [m(t) + A \cos 2\pi f_c t] \phi(t),$$

$$\phi(t) = \frac{1}{2} + \frac{2}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos[2\pi f_c (2n-1)t]$$

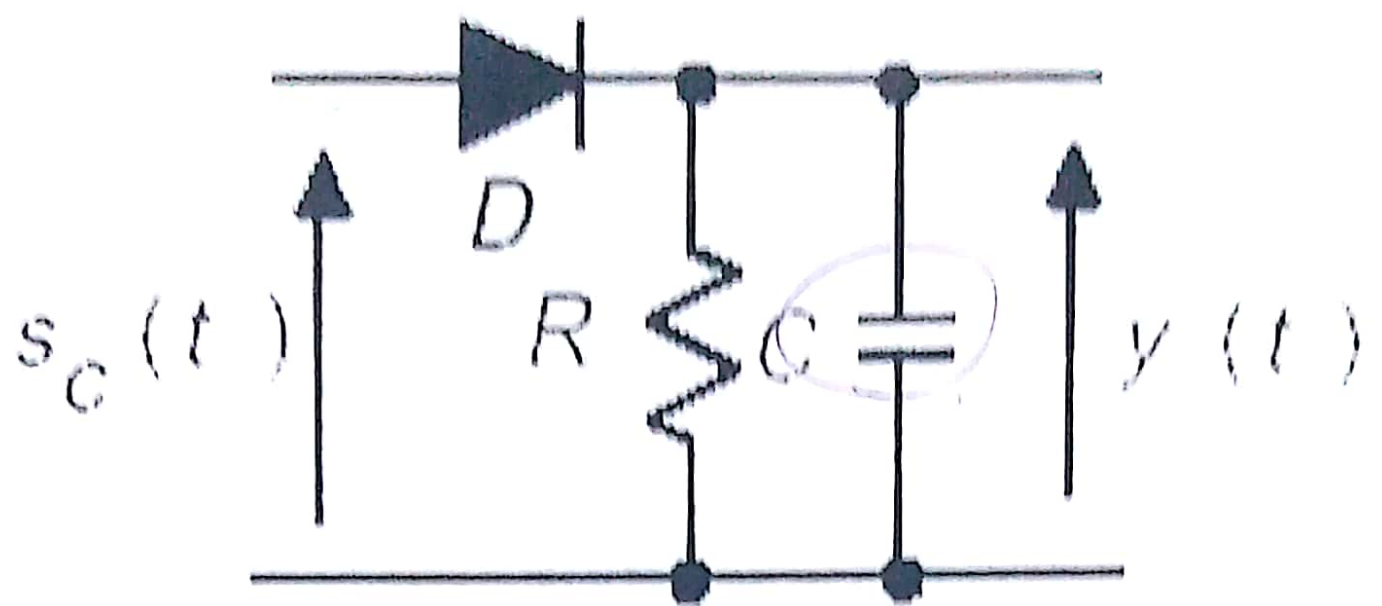
- $\phi(t)$: Square train pulse; then pass it BPF($\pm f_c$)

$$v_o(t) = \left[\frac{A}{2} + \frac{2}{\pi} m(t) \right] \cos 2\pi f_c t$$

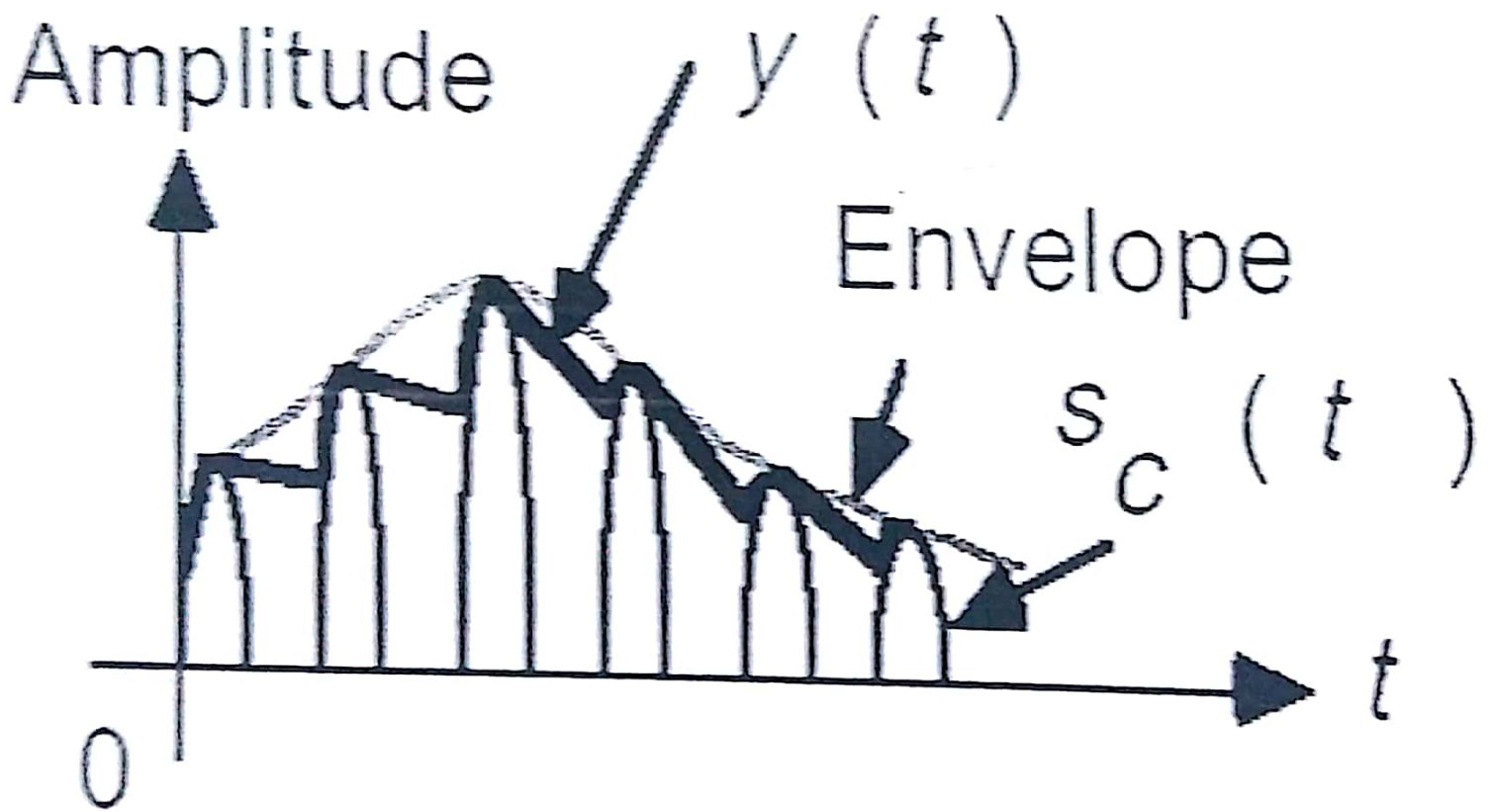
Demodulation of AM signals

- AM signals can be demodulated by
 - Envelope detector
 - Rectifier detector
 - Coherent (synchronous) detector.

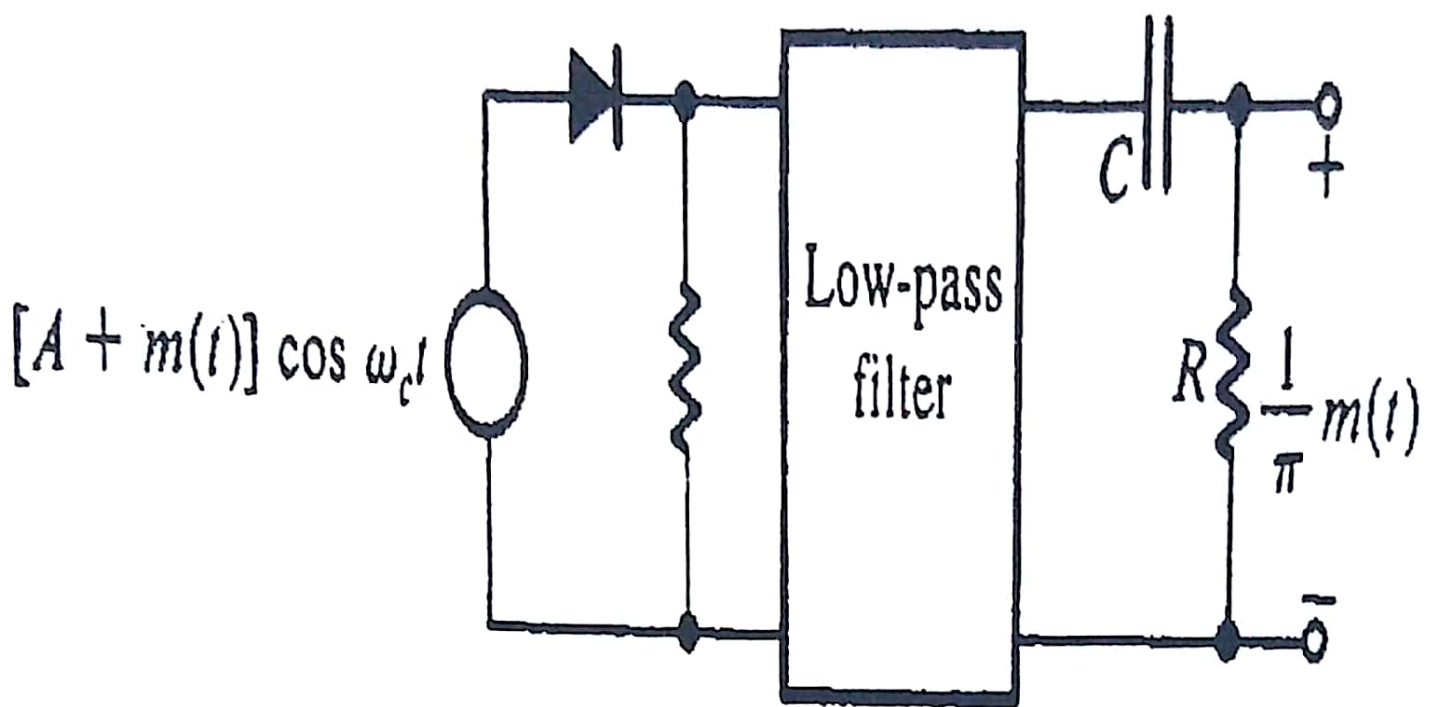
Envelope Detector

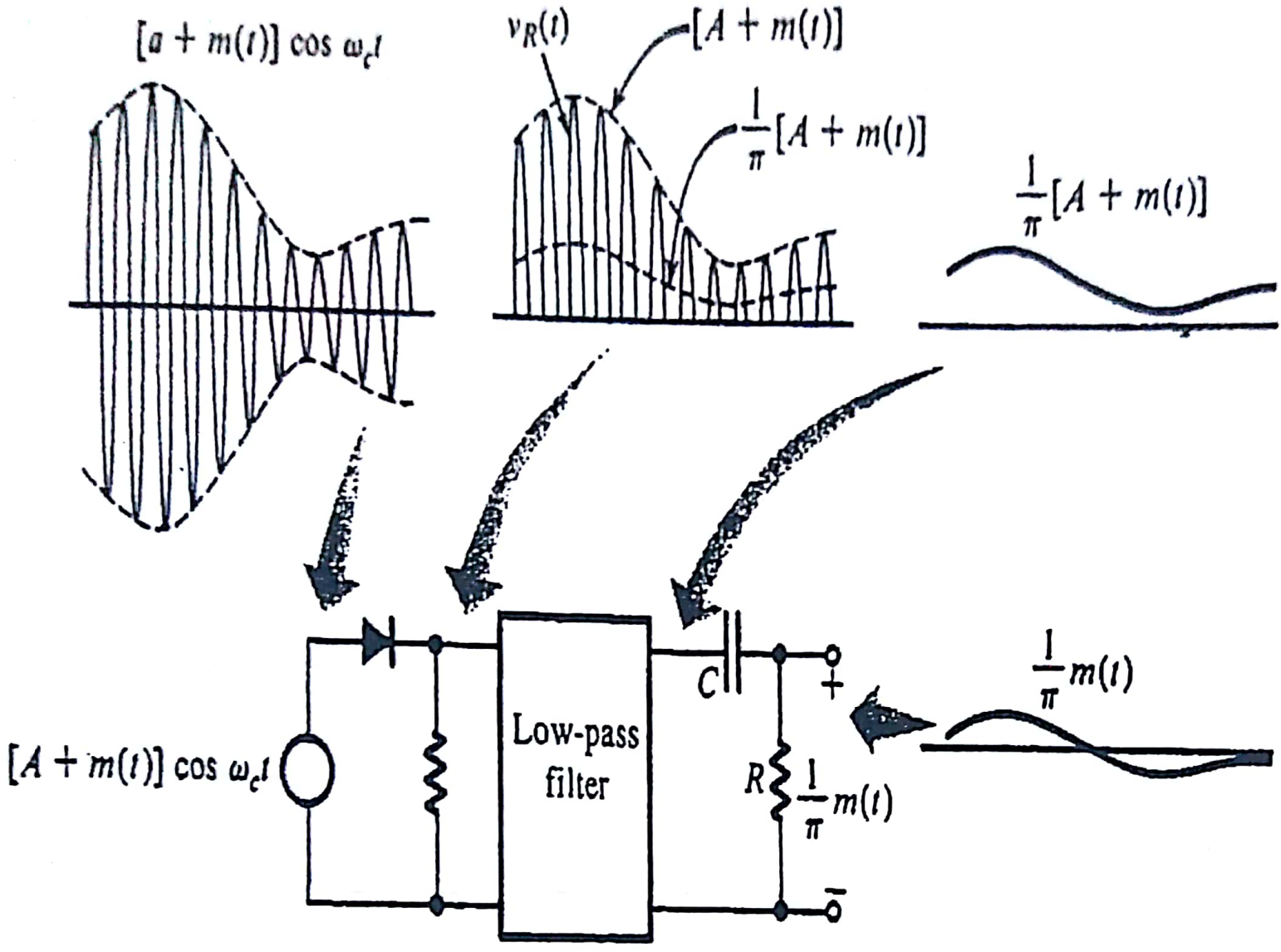


Envelope Detector (Cont.)



Rectifier Detector





Rec. Detector cont.

- Hence,

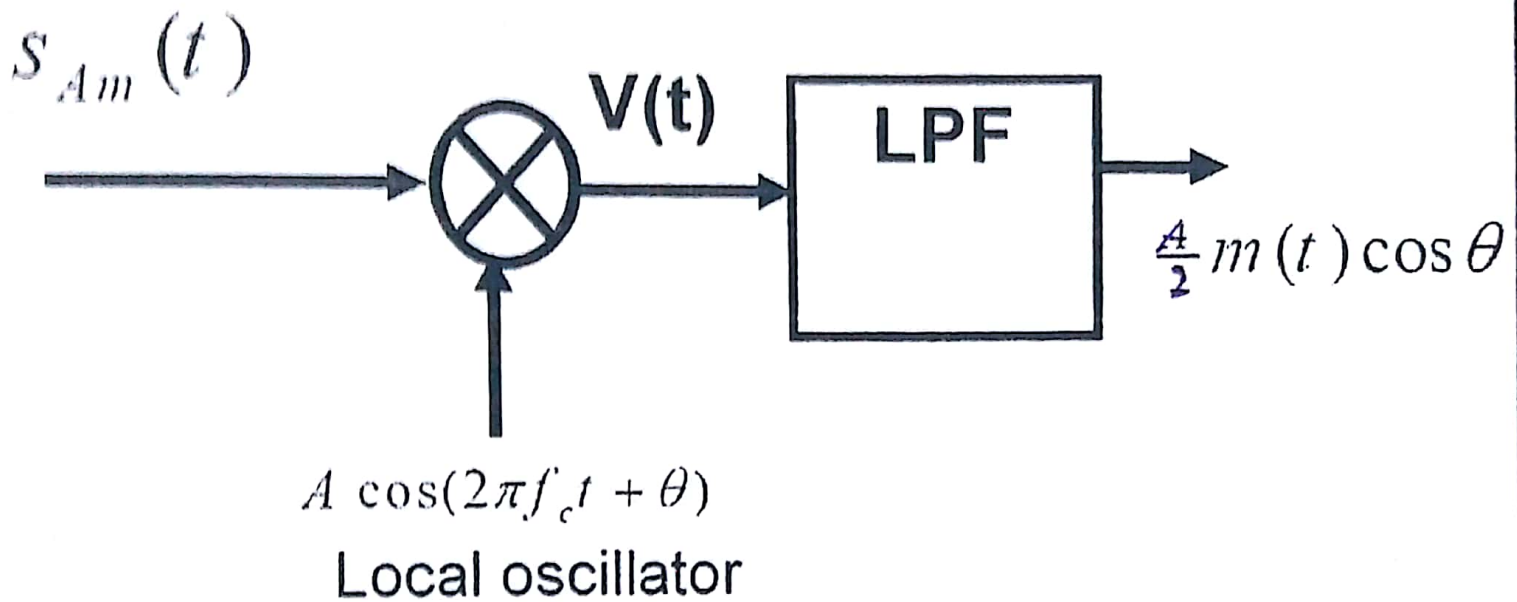
$$v_R(t) = [A + m(t)] \cos 2\pi f_c t \phi(t),$$

$$\phi(t) = \frac{1}{2} + \frac{2}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos[2\pi f_c (2n-1)t]$$

- Or

$$v_R(t) = \frac{1}{\pi} [A + m(t)] + \text{high freq. terms}$$

Coherent detector



Advantages/Disadvantages of Conventional AM (DSB-LC)

- **Advantages**
 - Very simple demodulation (envelope detector)
 - “Linear” modulation
- **Disadvantages**
 - Low power efficiency
 - Transmission bandwidth twice the message bandwidth.

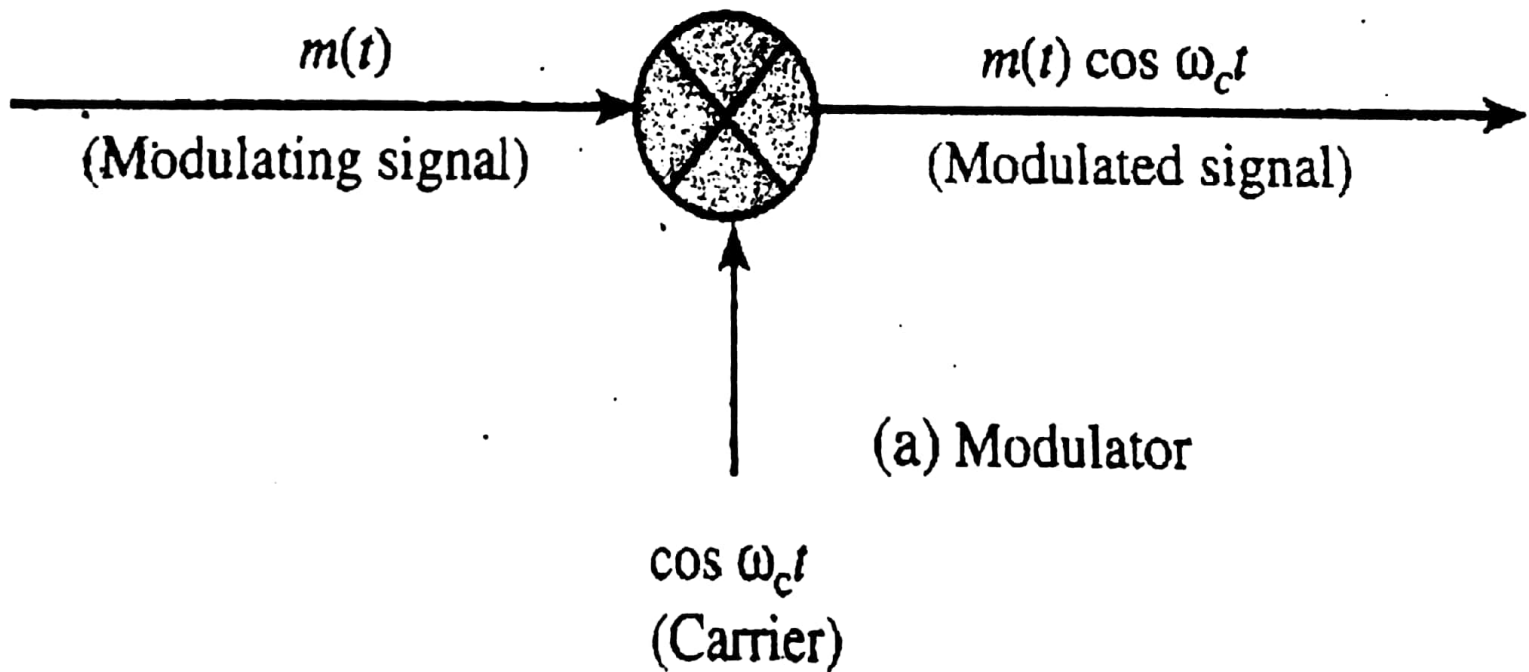


EE325: Chapter 4 (Lec. #2)

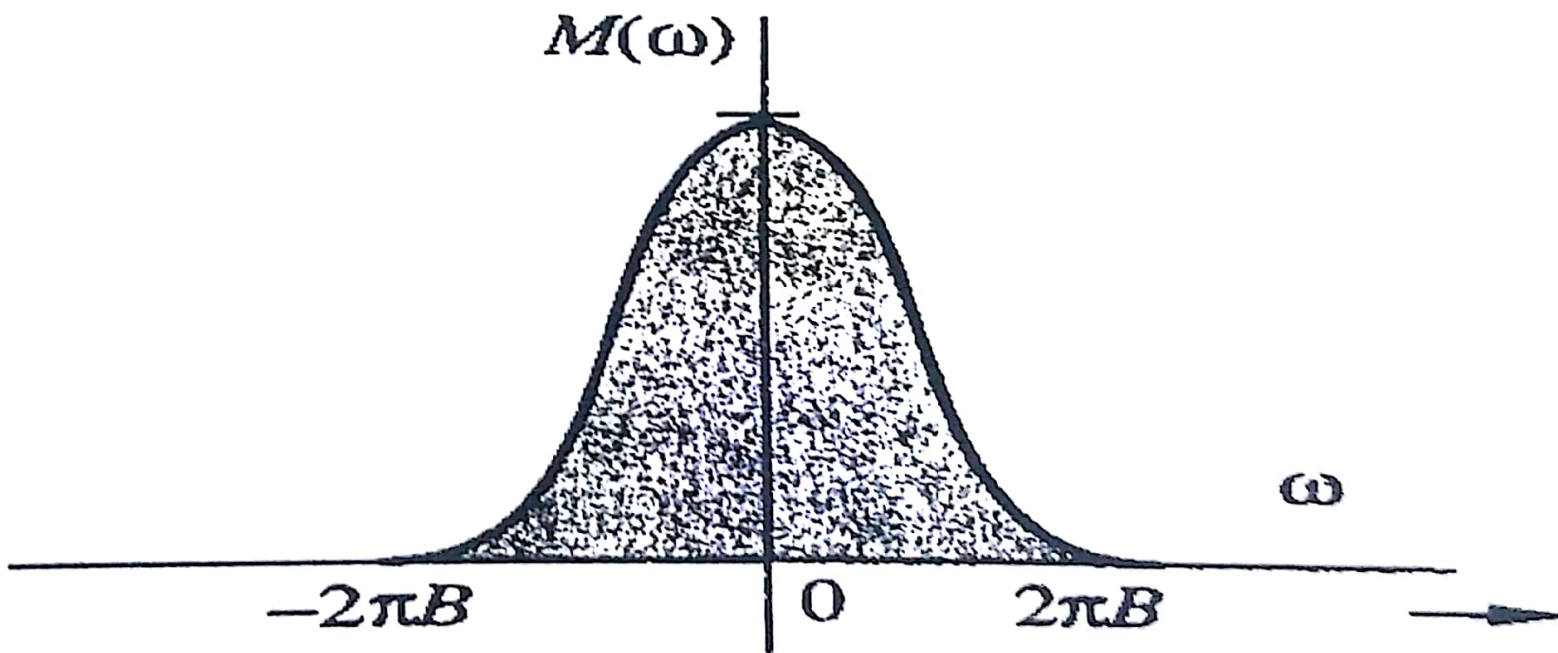
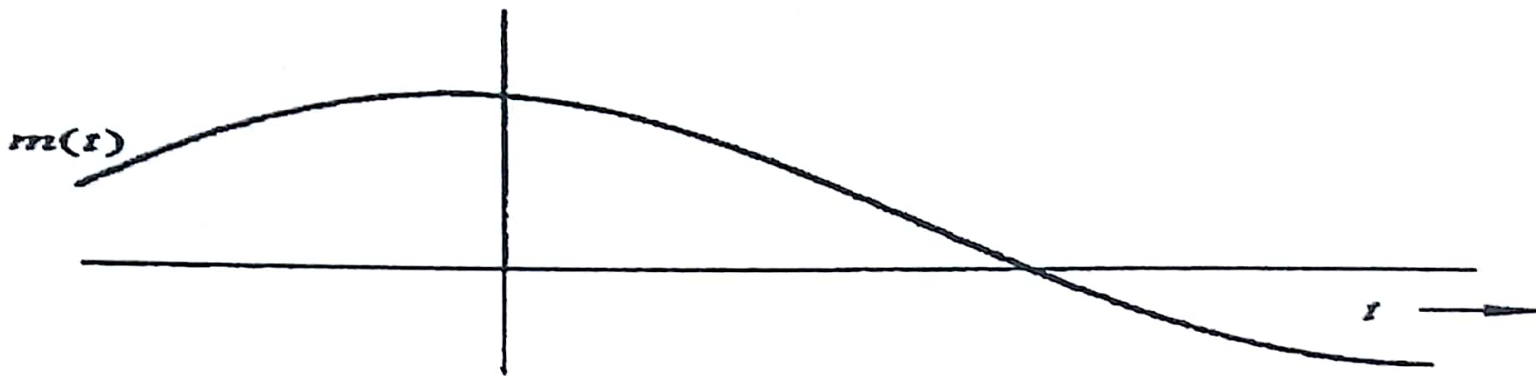
Amplitude Modulations & Demodulations

M. A. Smadi

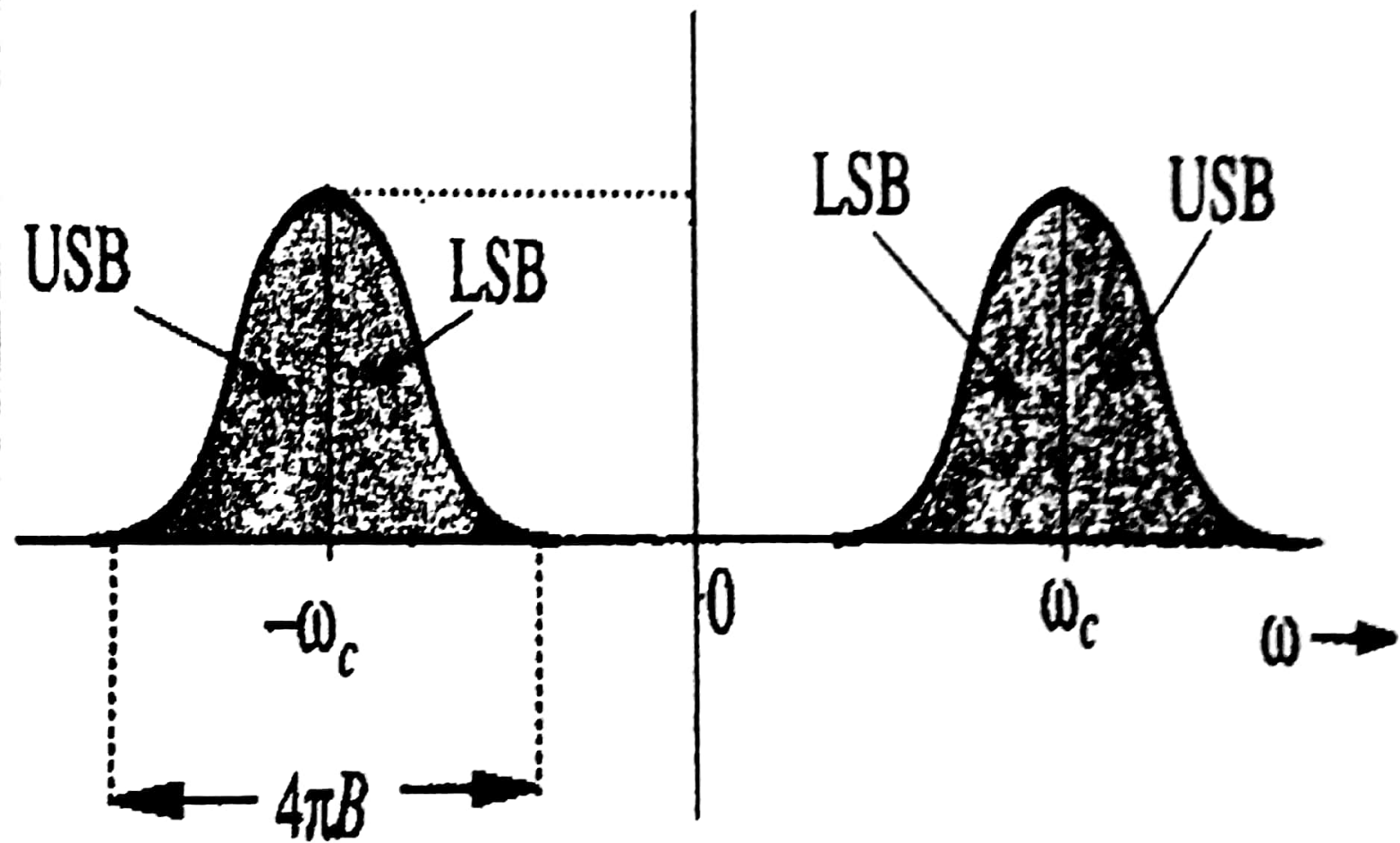
Double-sideband suppressed carrier DSBSC



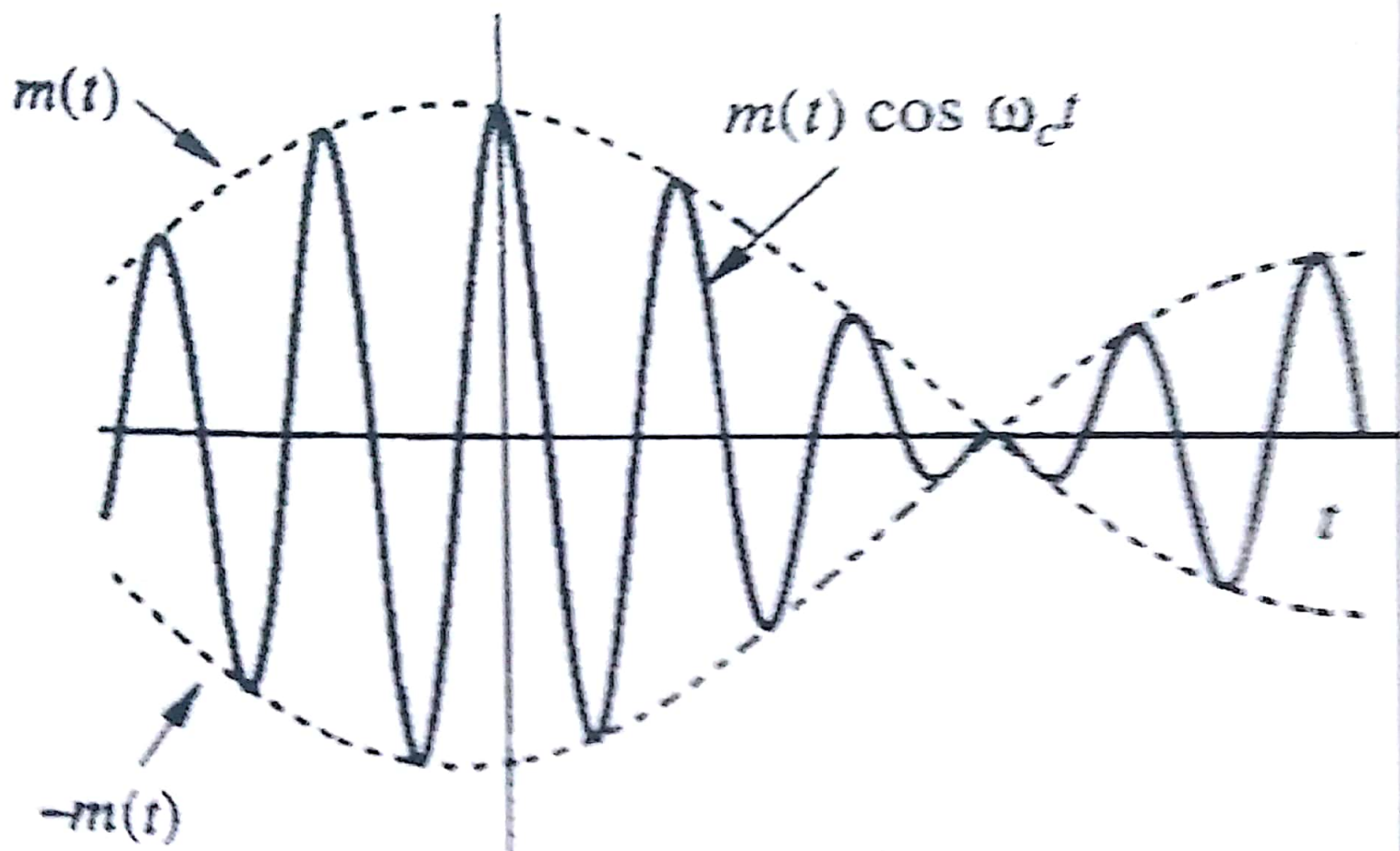
The modulating signal $m(t)$



DSBSC signal: $m(t) \cos(\omega_c t)$

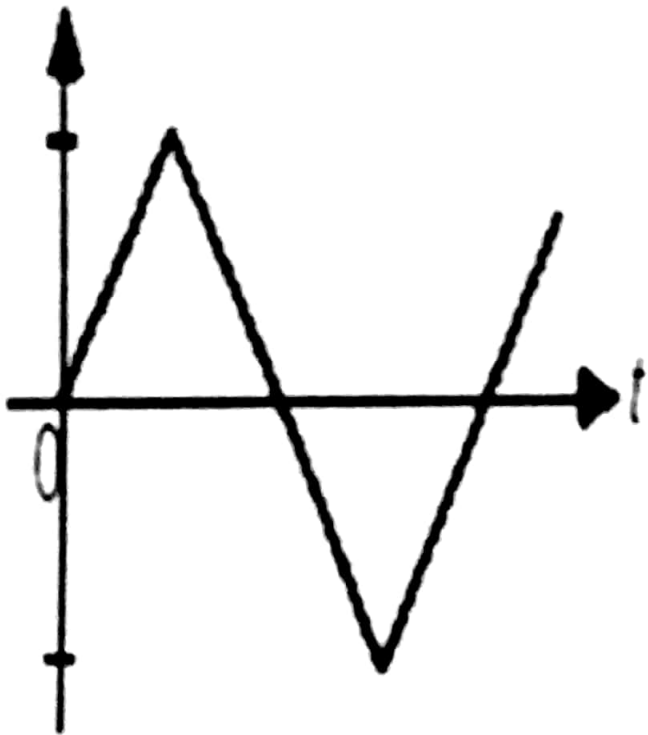


Modulated signal $m(t) \cos(\omega_c t)$

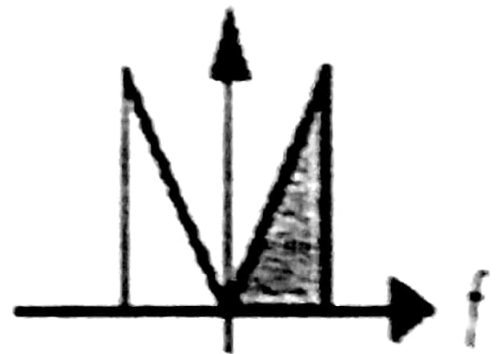


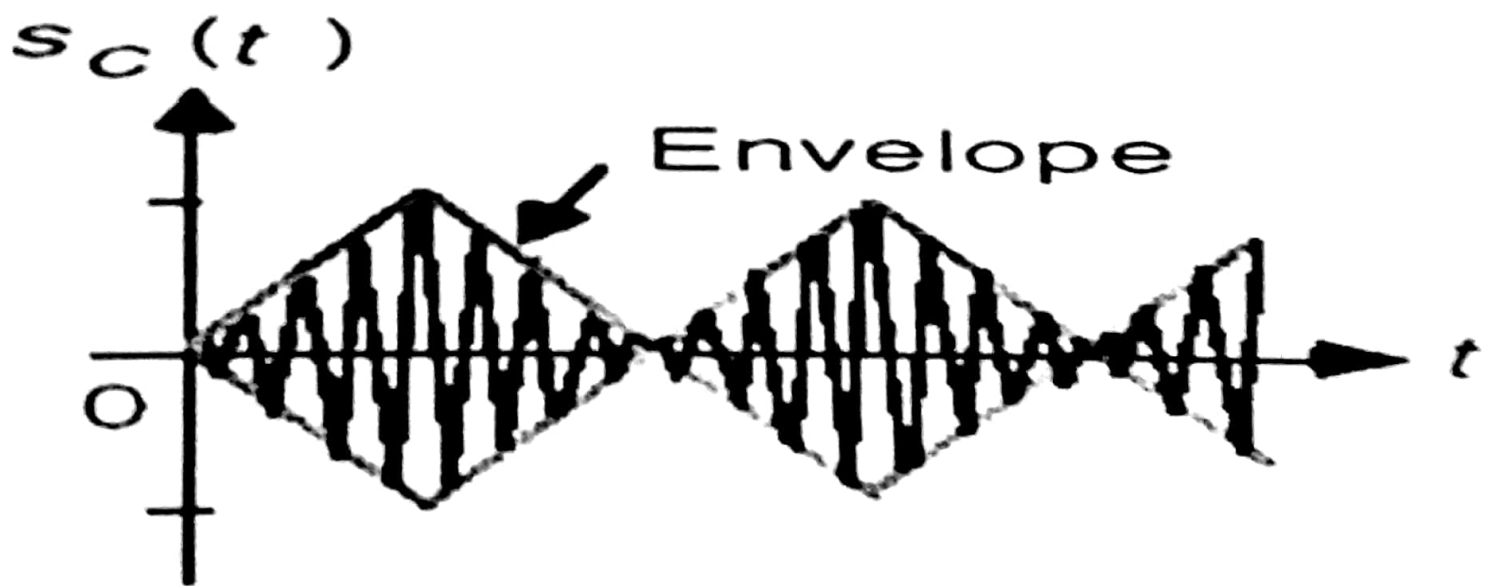
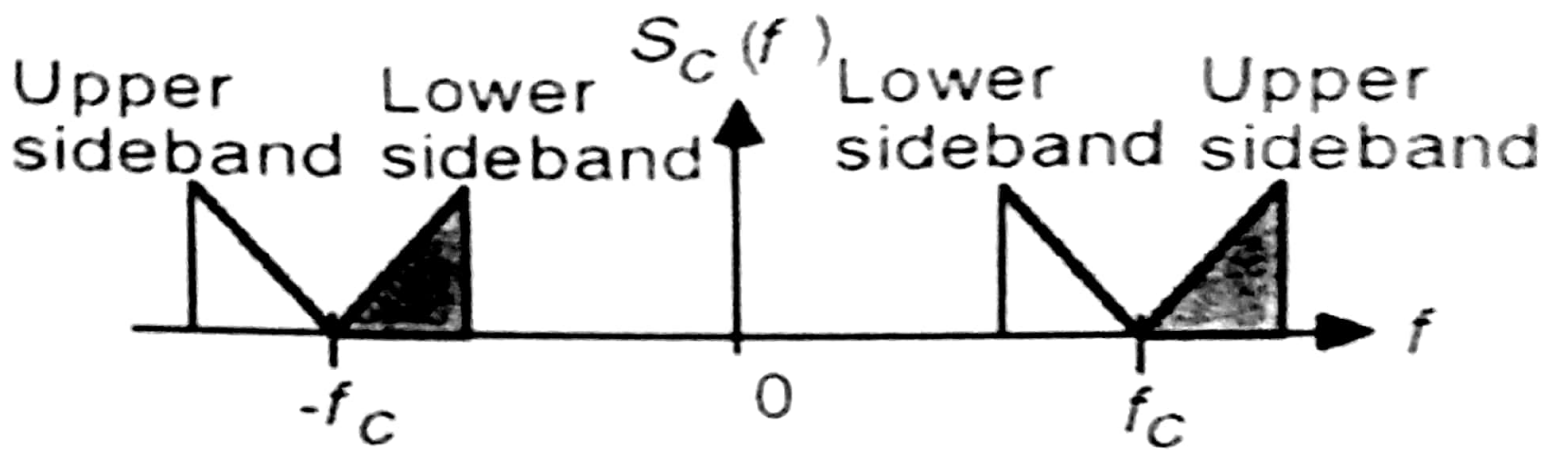
Example.

$m(t)$



$M(f)$

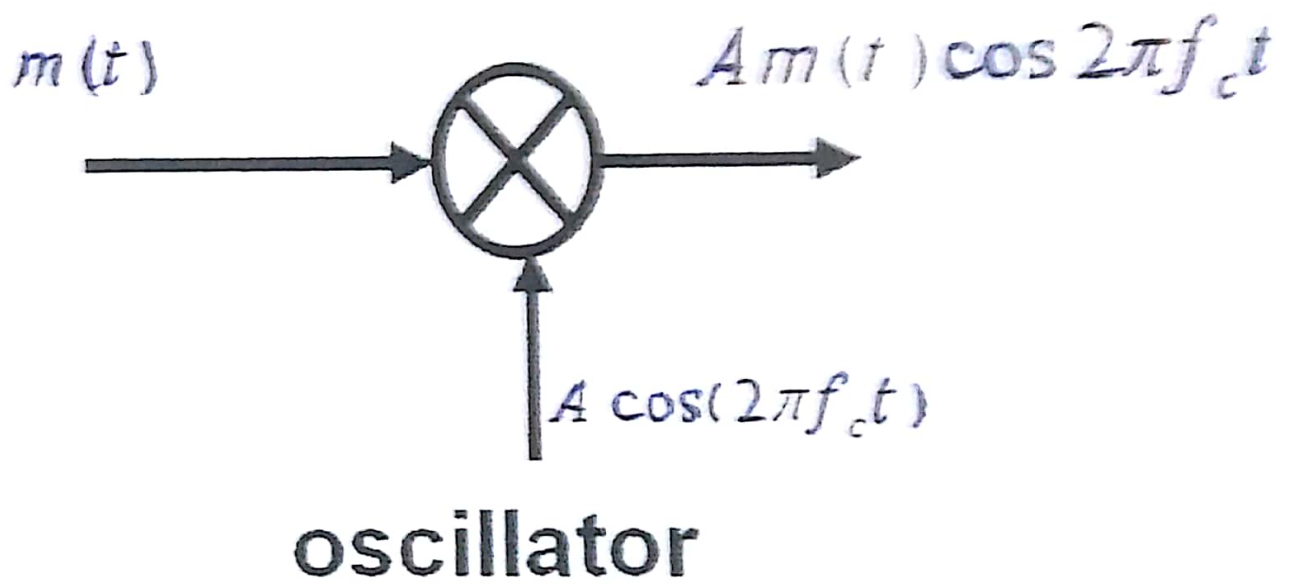




DSBSC Modulators

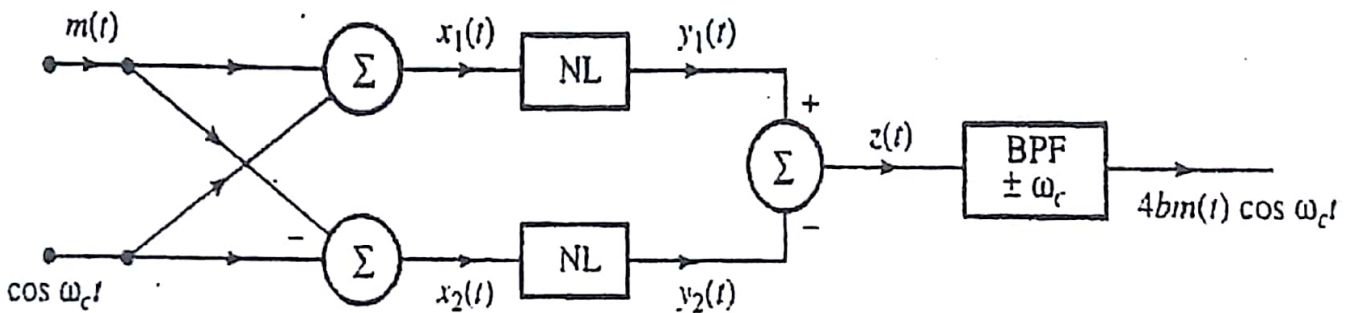
- DSBSC signal can be generated using several types of modulators:
 - Multiplier Modulators
 - Nonlinear Modulators
 - Switching Modulators

Multiplier modulator



Nonlinear Modulators

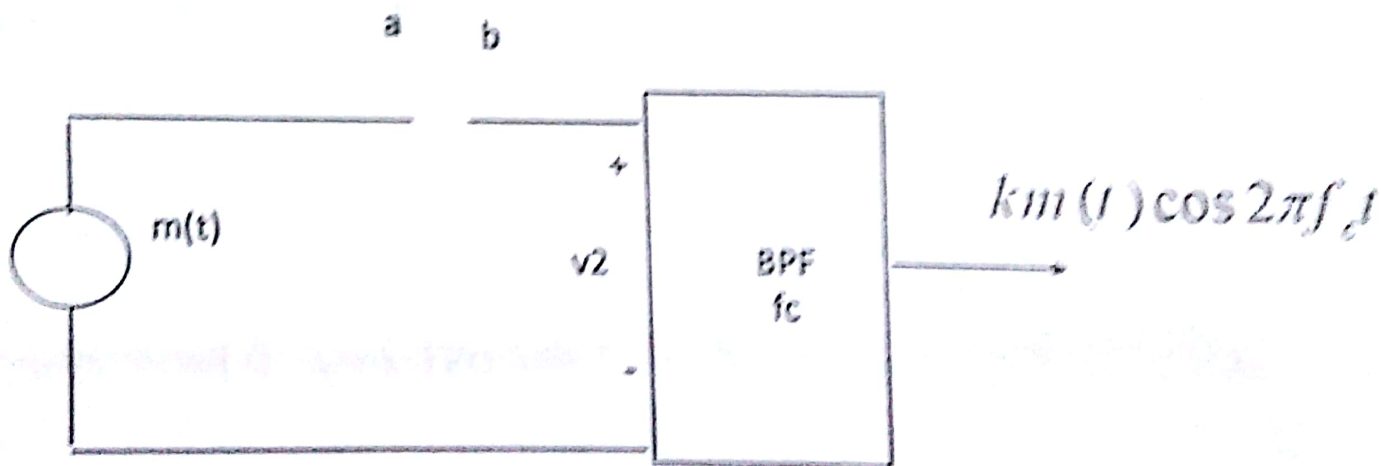
Single Balanced Mod.



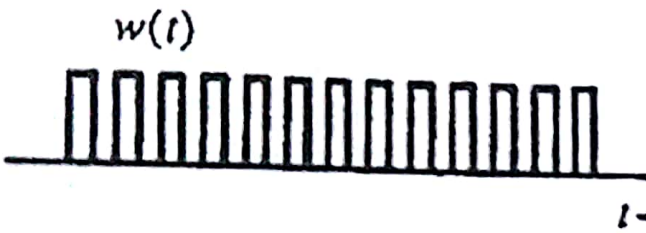
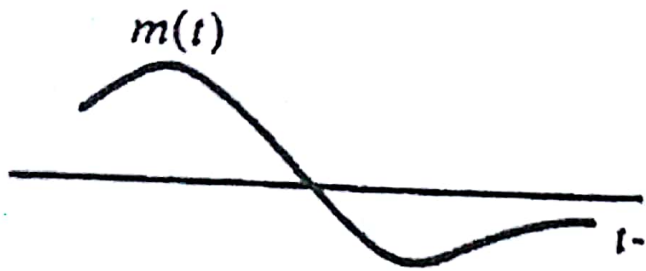
$$y(t) = ax(t) + bx^2(t)$$

$$z(t) = 2am(t) + \underline{4bm(t) \cos(2\pi f_c t)}$$

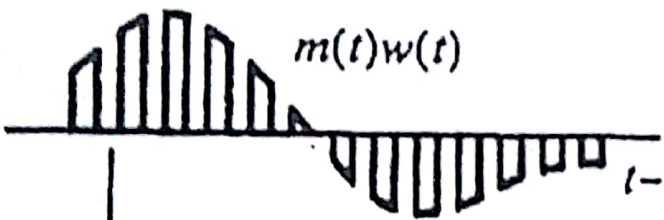
Switching Modulators

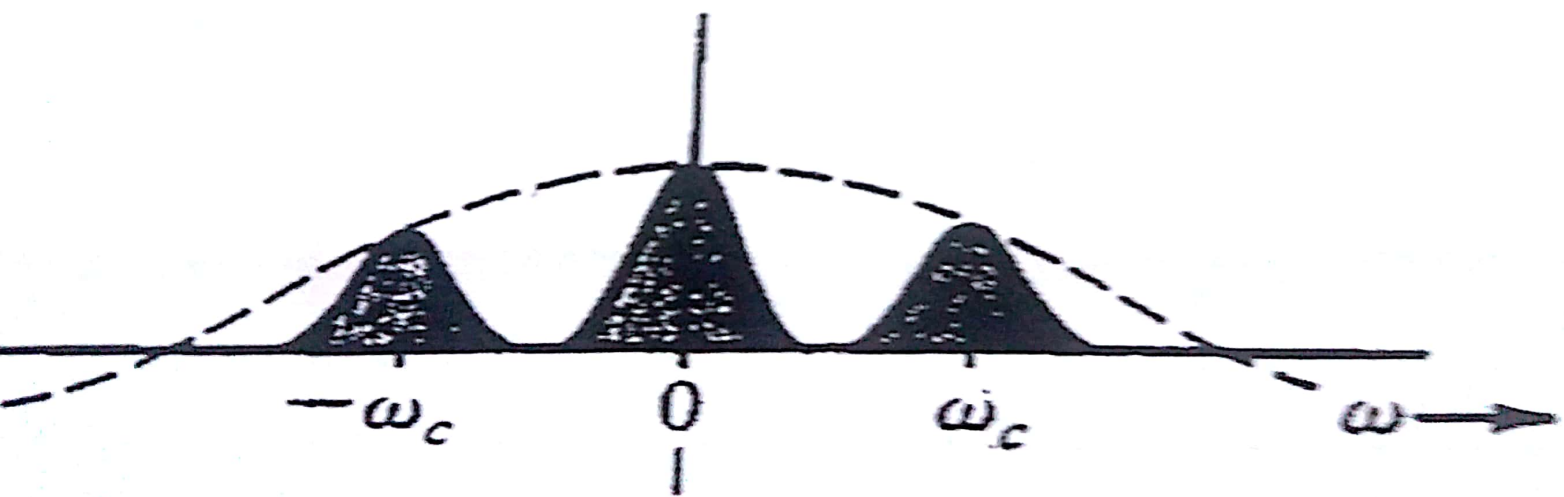
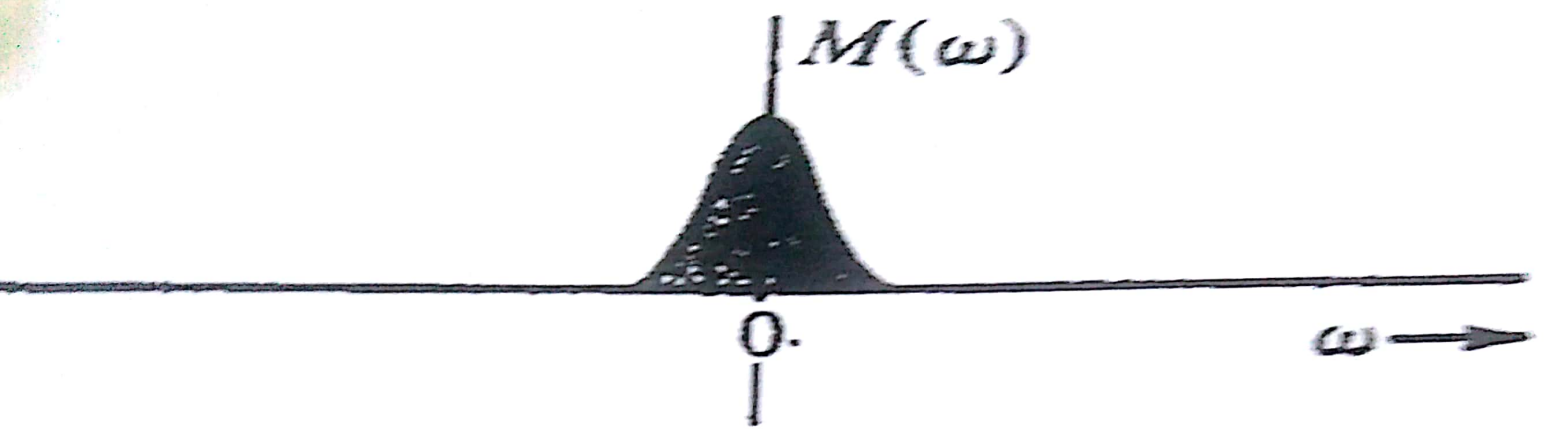


Switching Modulators

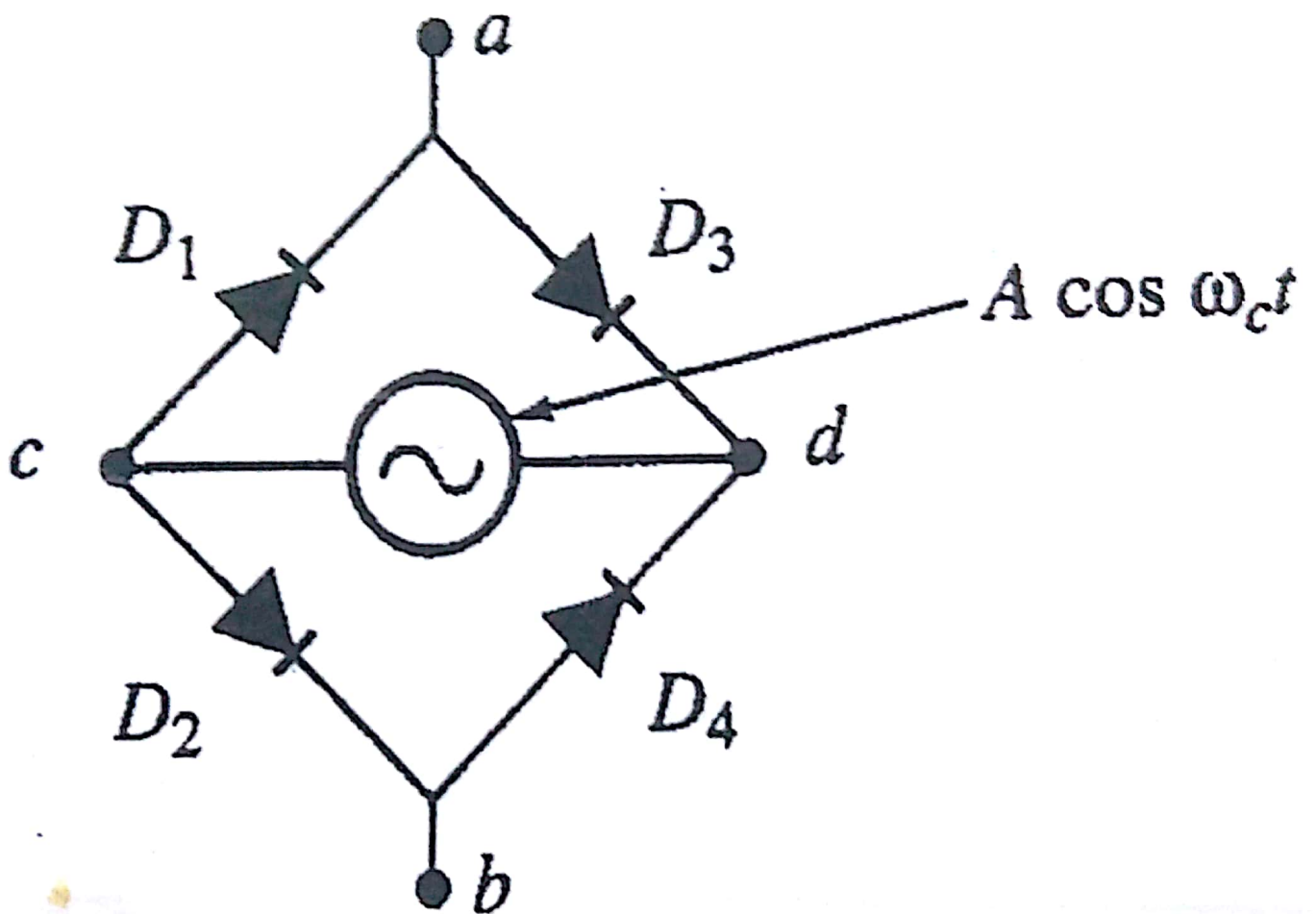


$$w(t) = \frac{1}{2} + \frac{2}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos[2\pi f_c (2n-1)t]$$

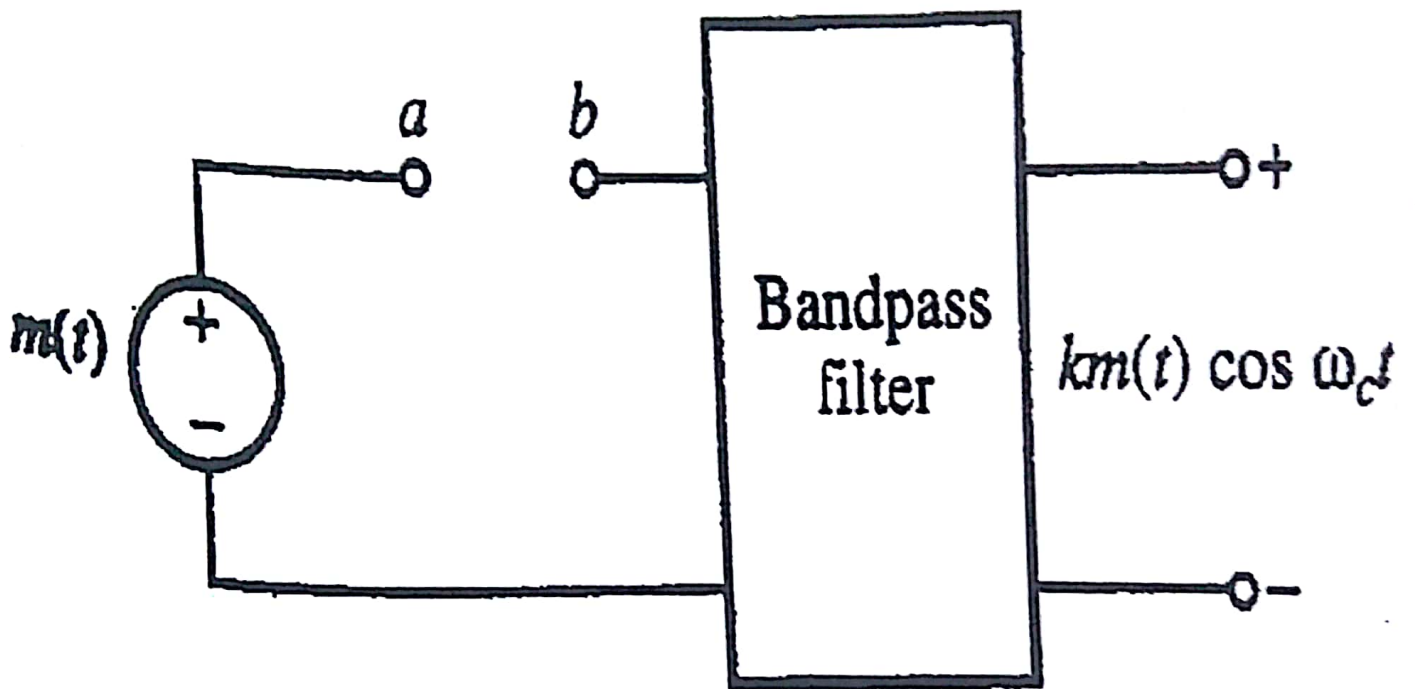




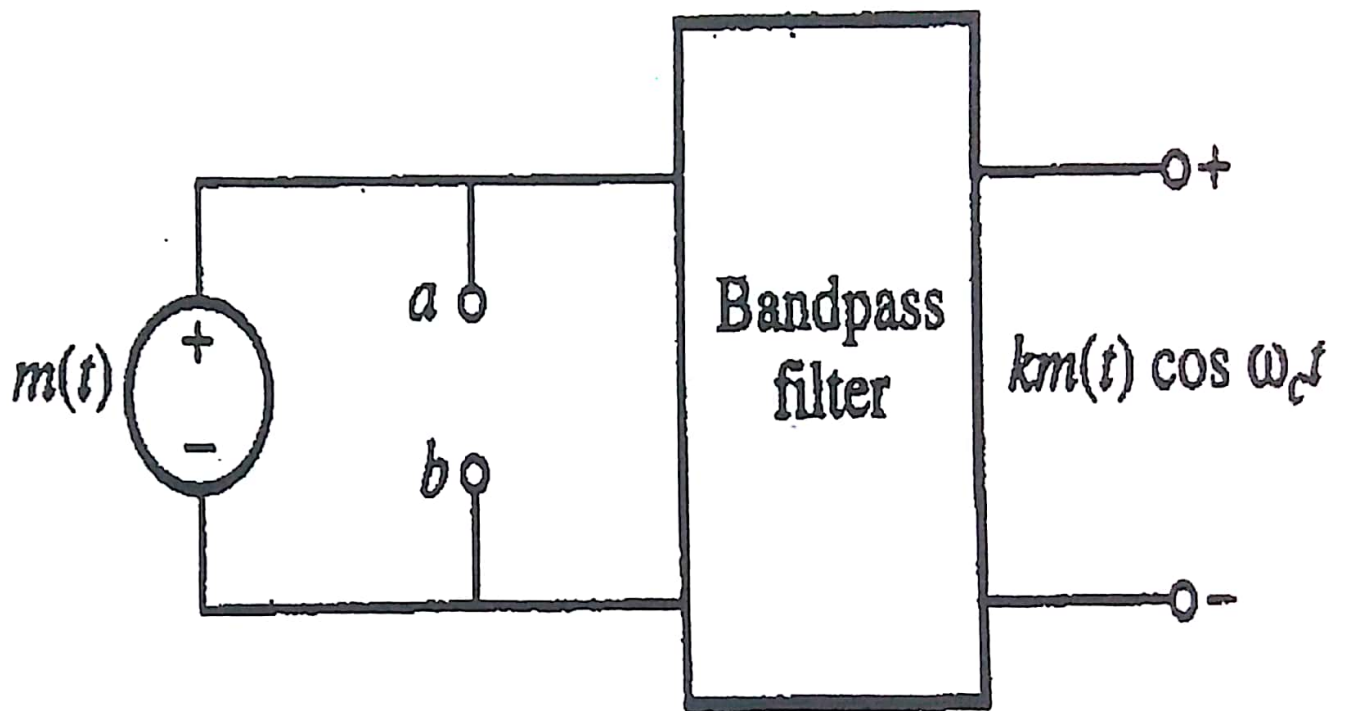
Diode-bridge mod. (switch)



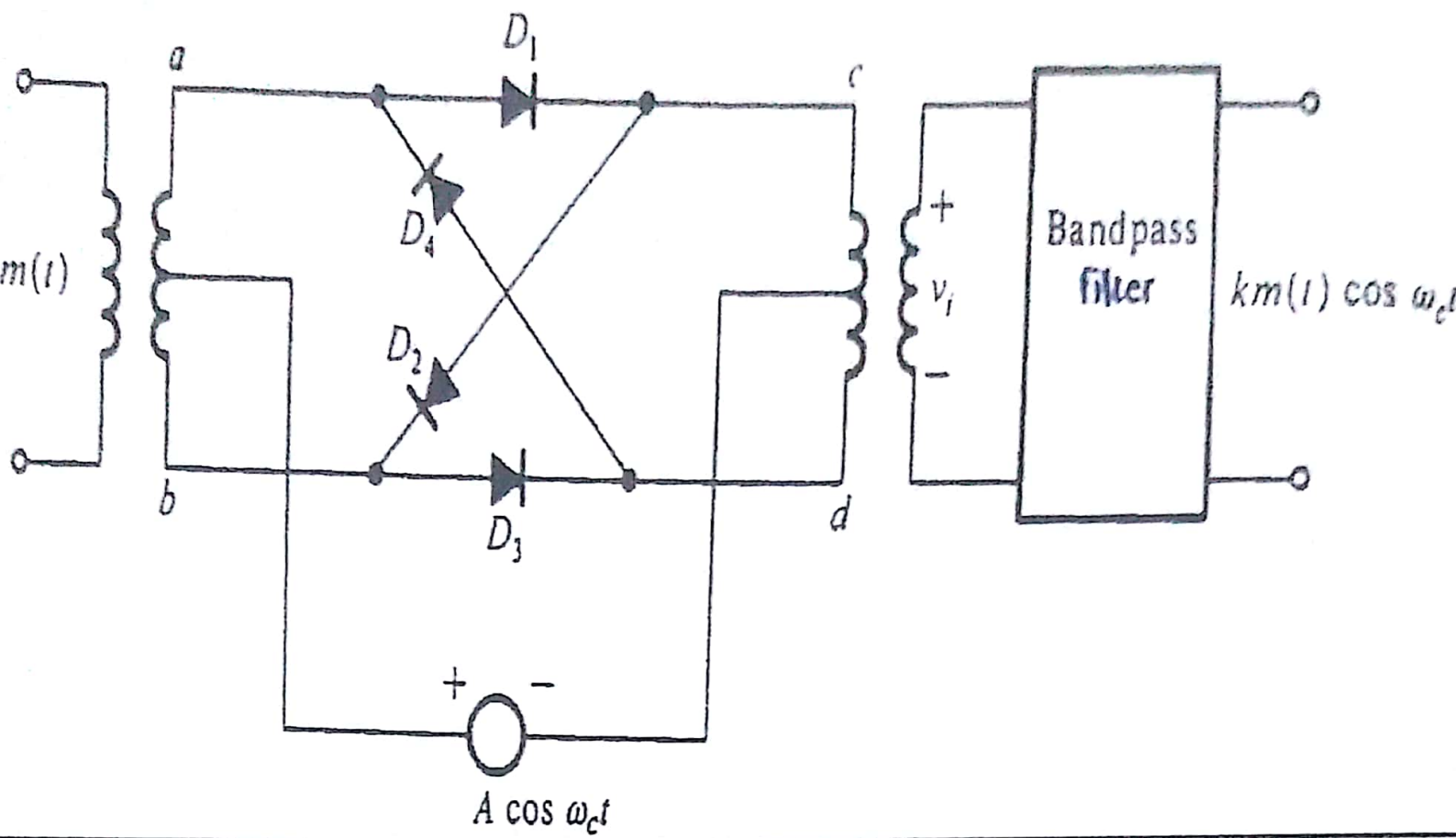
Series-bridge diode modulator



Shunt-bridge diode modulator



Ring Modulator

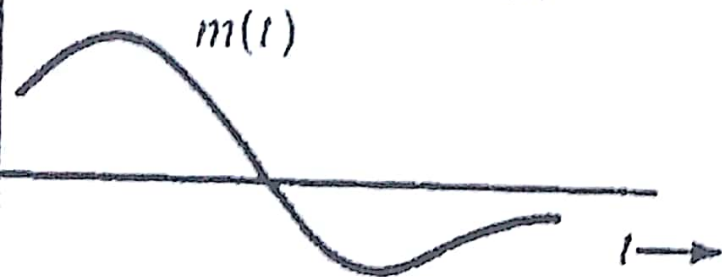


Ring modulator Double Balanced Mod.

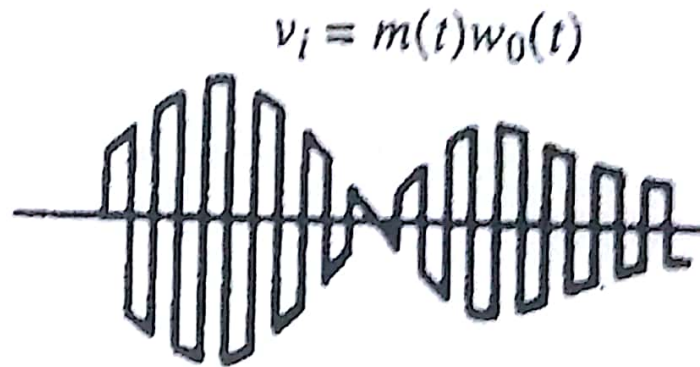


(b)

$$w_0(t) = 2w(t) - 1 = \frac{4}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos[2\pi f_c (2n-1)t]$$

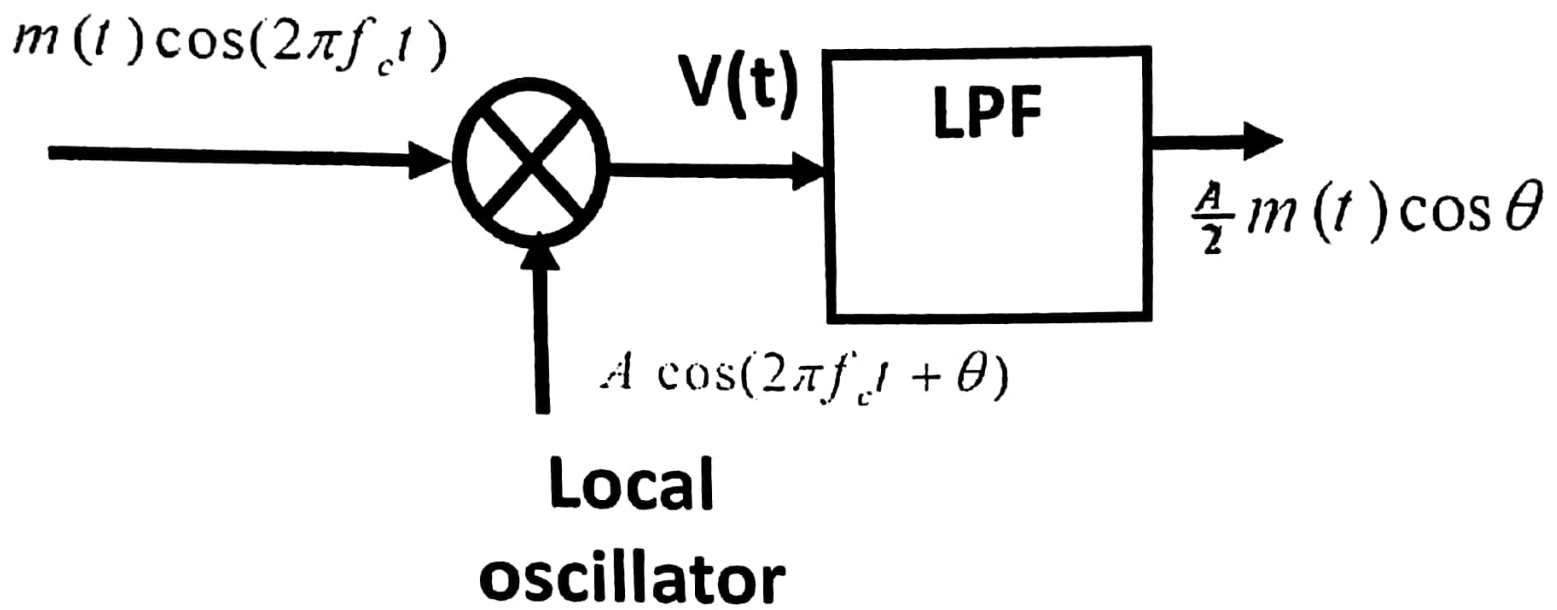


(c)

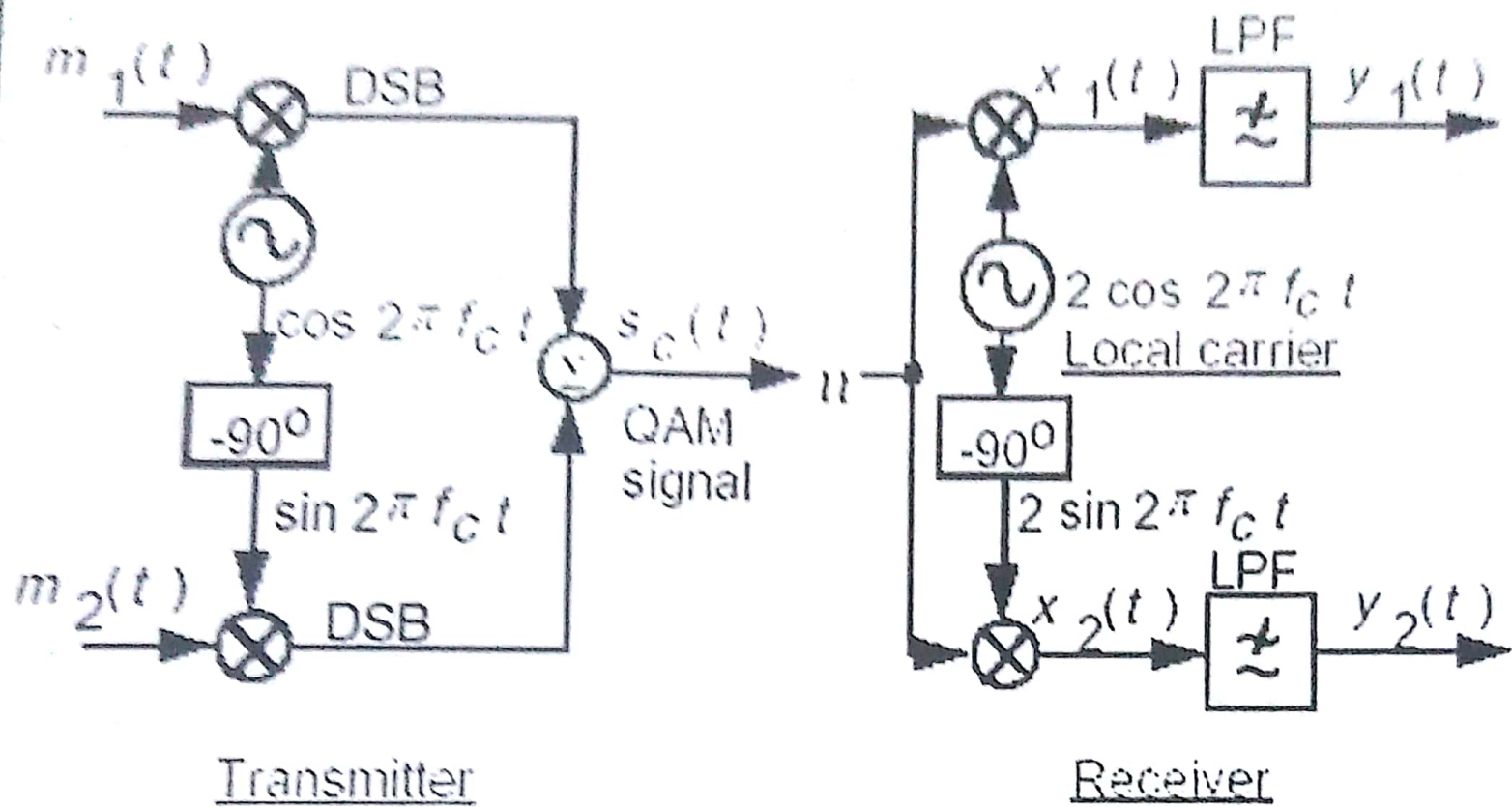


(d)

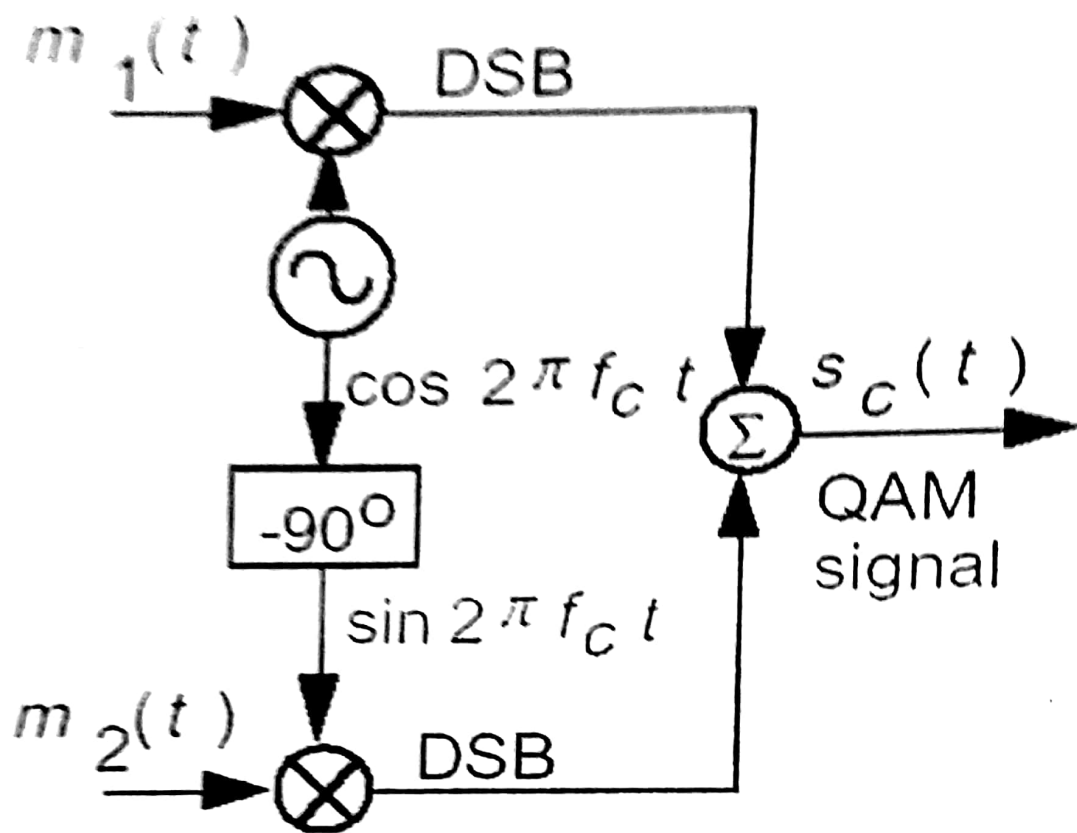
Demodulation of DSBSC



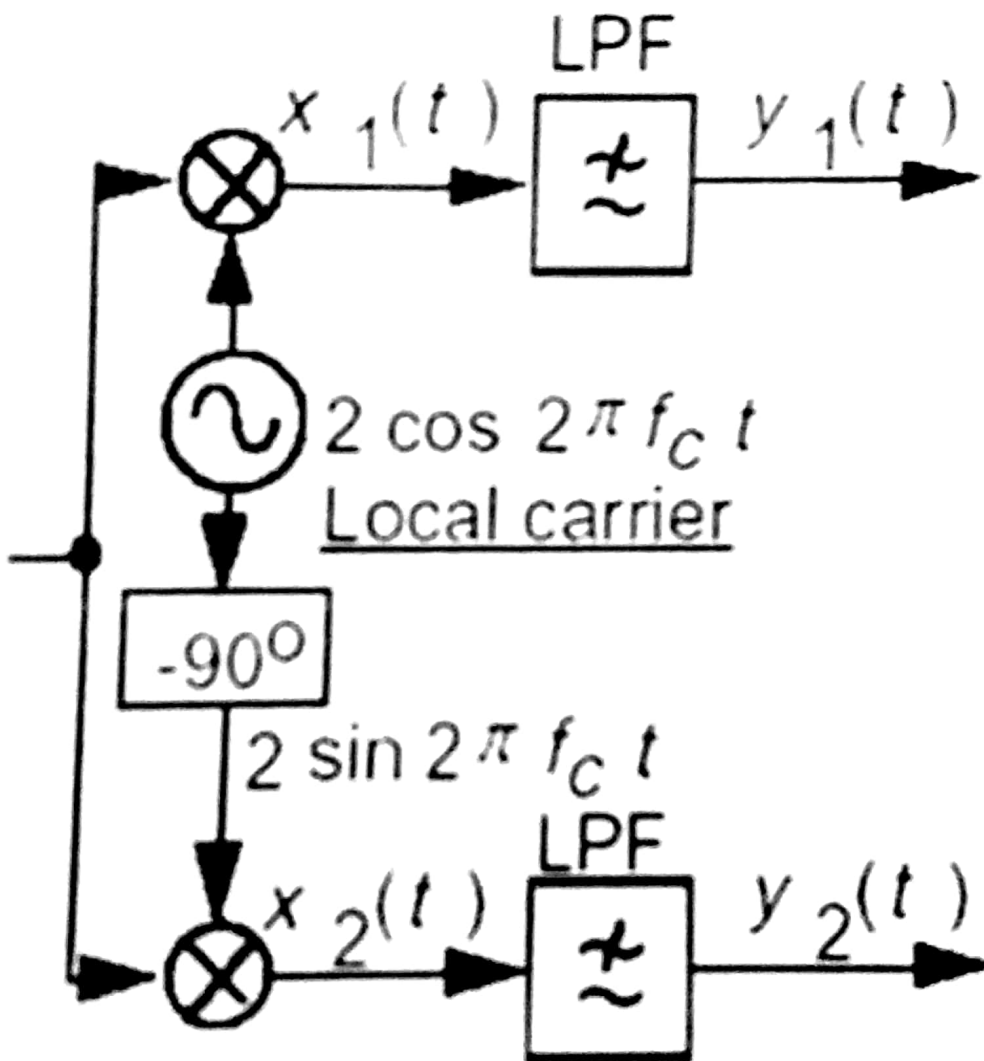
Quadrature Amplitude Modulation (QAM)



Transmitter



Receiver



QAM cont.

- Quadrature multiplexing is used in color television to multiplex the signals which carry the information about colors.

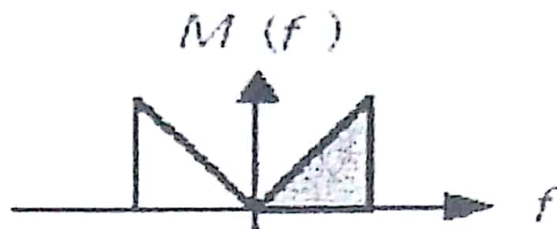


EE325: Chapter 4 (Lec. #3)

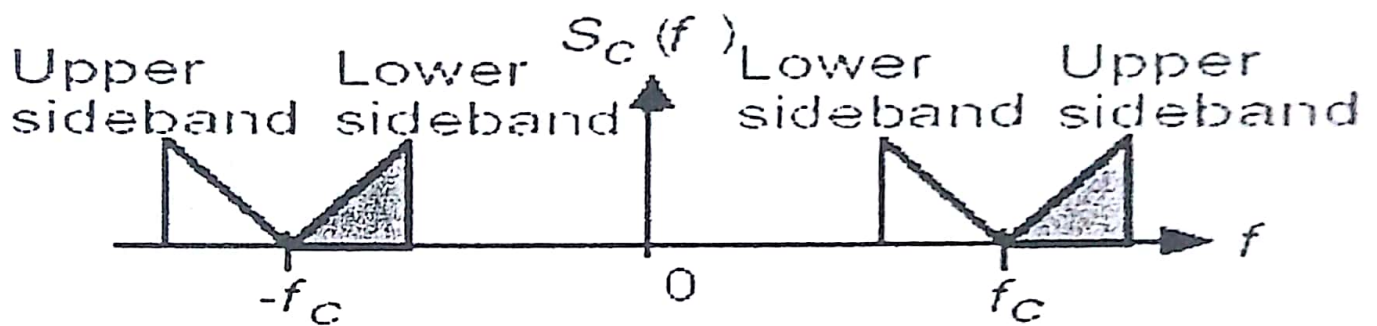
Amplitude Modulations & Demodulations

M. A. Smadi

Single Sideband (SSB)



(a)



SSB time representation

$$S_{SSB}(t) = m(t) \cos 2\pi f_c t \mp m_h(t) \sin 2\pi f_c t,$$

- : USB

+ : LSB

$$m_h(t) = m(t) * \frac{1}{\pi t} \quad \square \text{ Hilbert transform of } m(t)$$

OR

$$M_h(f) = -jM(f) \operatorname{sgn}(f) = M(f)H(f)$$

$$H(f) = 1 \begin{cases} -\pi/2, & f > 0 \\ \pi/2, & f < 0 \end{cases} \Rightarrow \text{Ideal phase shifter by } \pi/2$$

SSB representation

Note,

$$M_+(f) = M(f) \chi_1(f) = M(f) \frac{1}{2} [1 + \text{sgn}(f)] = \frac{1}{2} [M(f) + jM_h(f)]$$

$$M_-(f) = M(f) \chi_2(-f) = M(f) \frac{1}{2} [1 - \text{sgn}(f)] = \frac{1}{2} [M(f) - jM_h(f)]$$

Hence,

$$\begin{aligned} S_{\text{USB}}(f) &= M_+(f - fc) + M_-(f + fc) \\ &= \frac{1}{2} [M(f - fc) + M(f + fc)] - \frac{1}{2j} [M_h(f - fc) - M_h(f + fc)] \end{aligned}$$

AND,

$$S_{\text{LSB}}(t) = m(t) \cos 2\pi f_c t - m_h(t) \sin 2\pi f_c t$$

Example

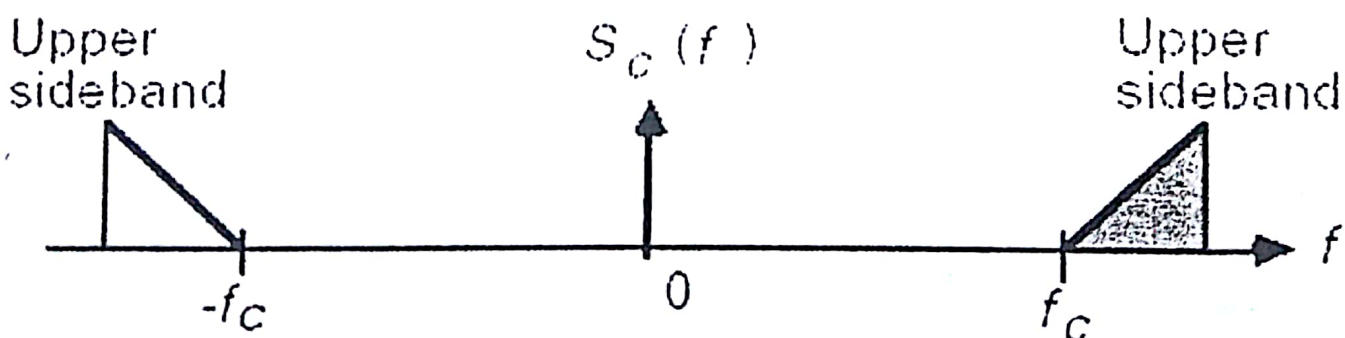
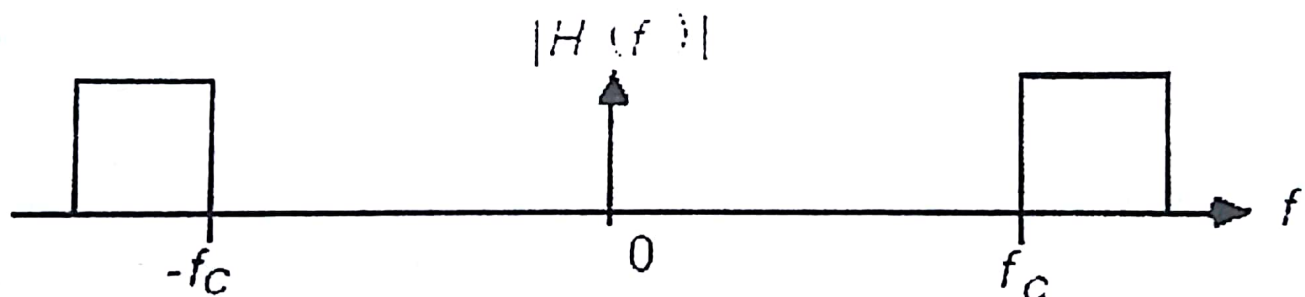
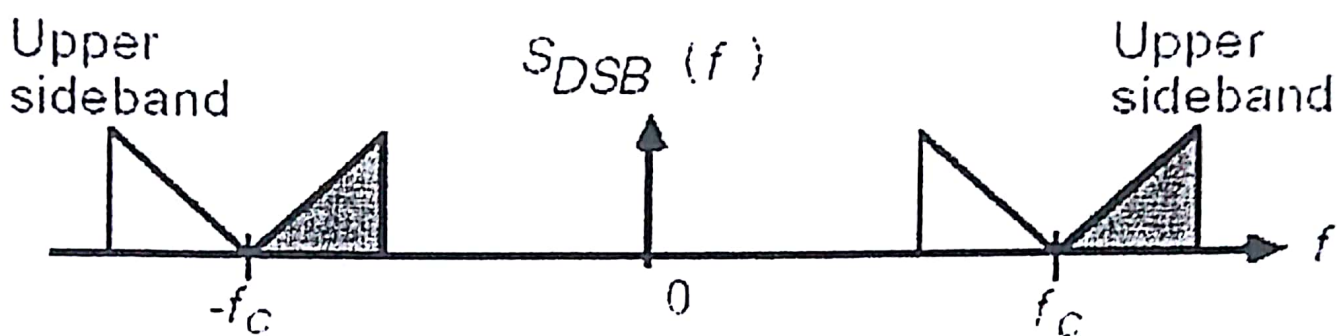
- Single tone message signal: $m(t) = \cos 2\pi f_m t$
- Then $m_h(t) = \cos(2\pi f_m t - \pi/2) = \sin(2\pi f_m t)$
- Hence,

$$S_{SSB}(t) = \cos(2\pi f_m t) \cos(2\pi f_c t) \mp \sin(2\pi f_m t) \sin(2\pi f_c t) = \cos(2\pi [f_c \pm f_m] t)$$

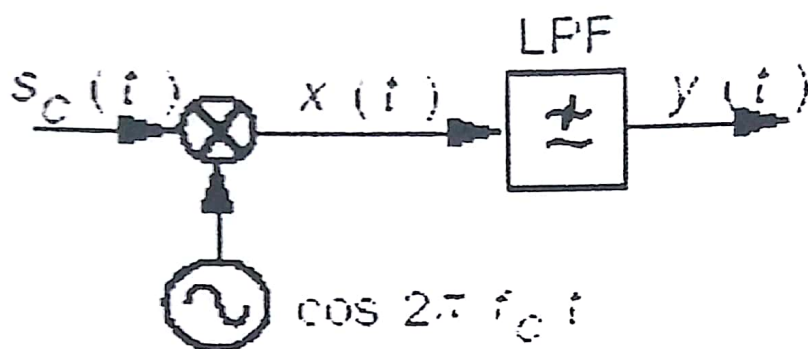
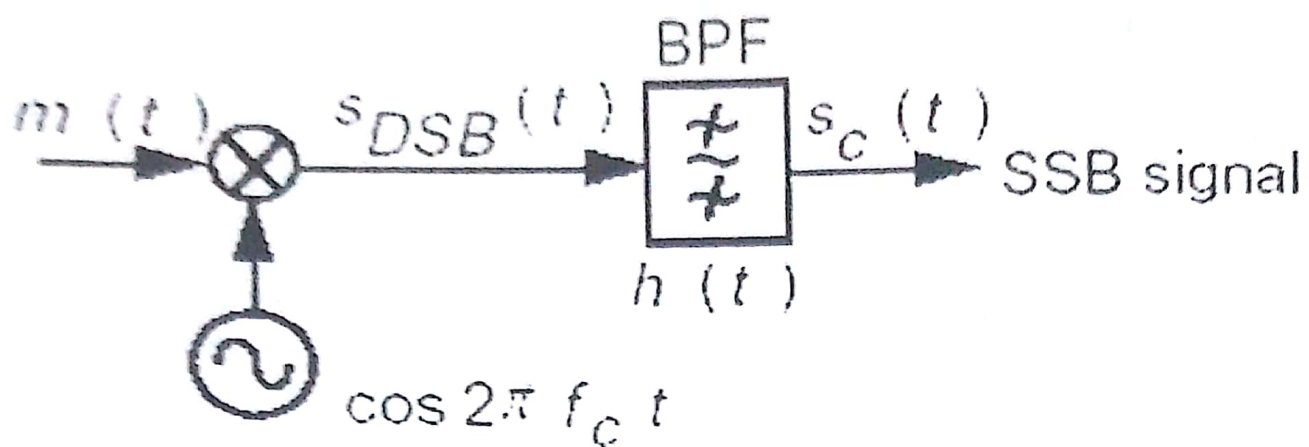
$$S_{USB}(f) = \frac{1}{2} [\delta(f_c + f_m) + \delta(-f_c - f_m)]$$

$$S_{LSB}(f) = \frac{1}{2} [\delta(f_c - f_m) + \delta(-f_c + f_m)]$$

Selective filtering method

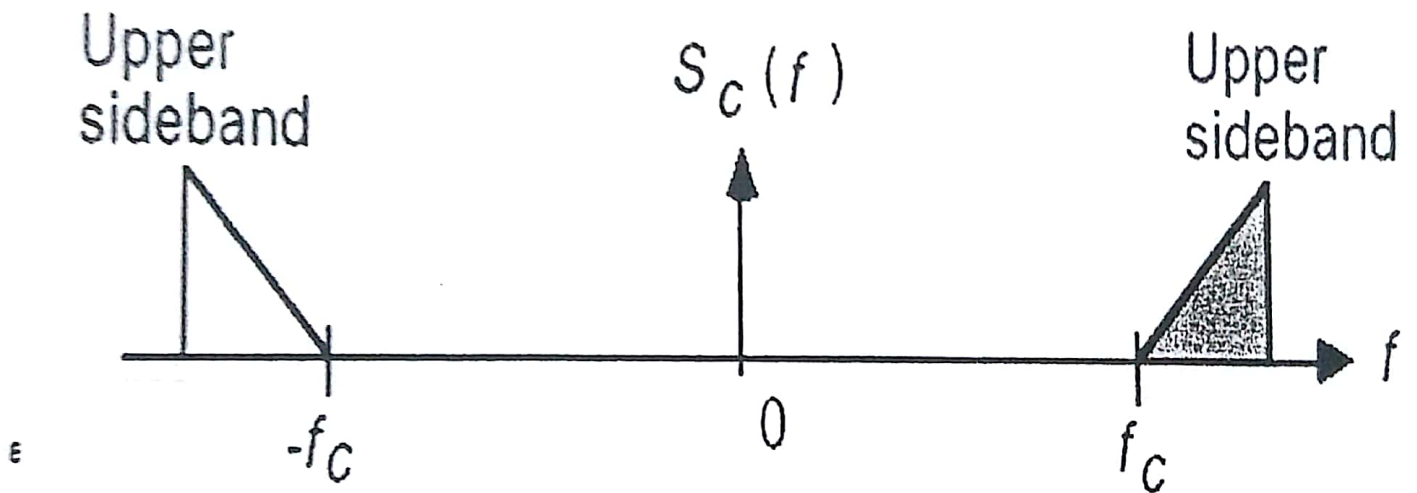
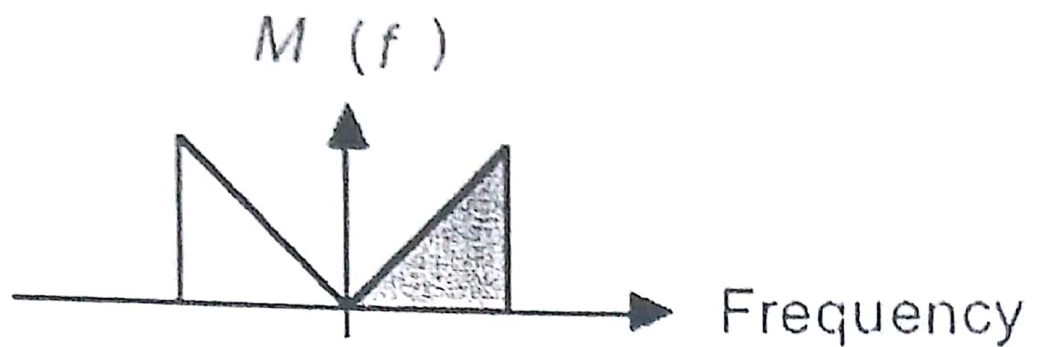


Selective filtering method (Cont.)

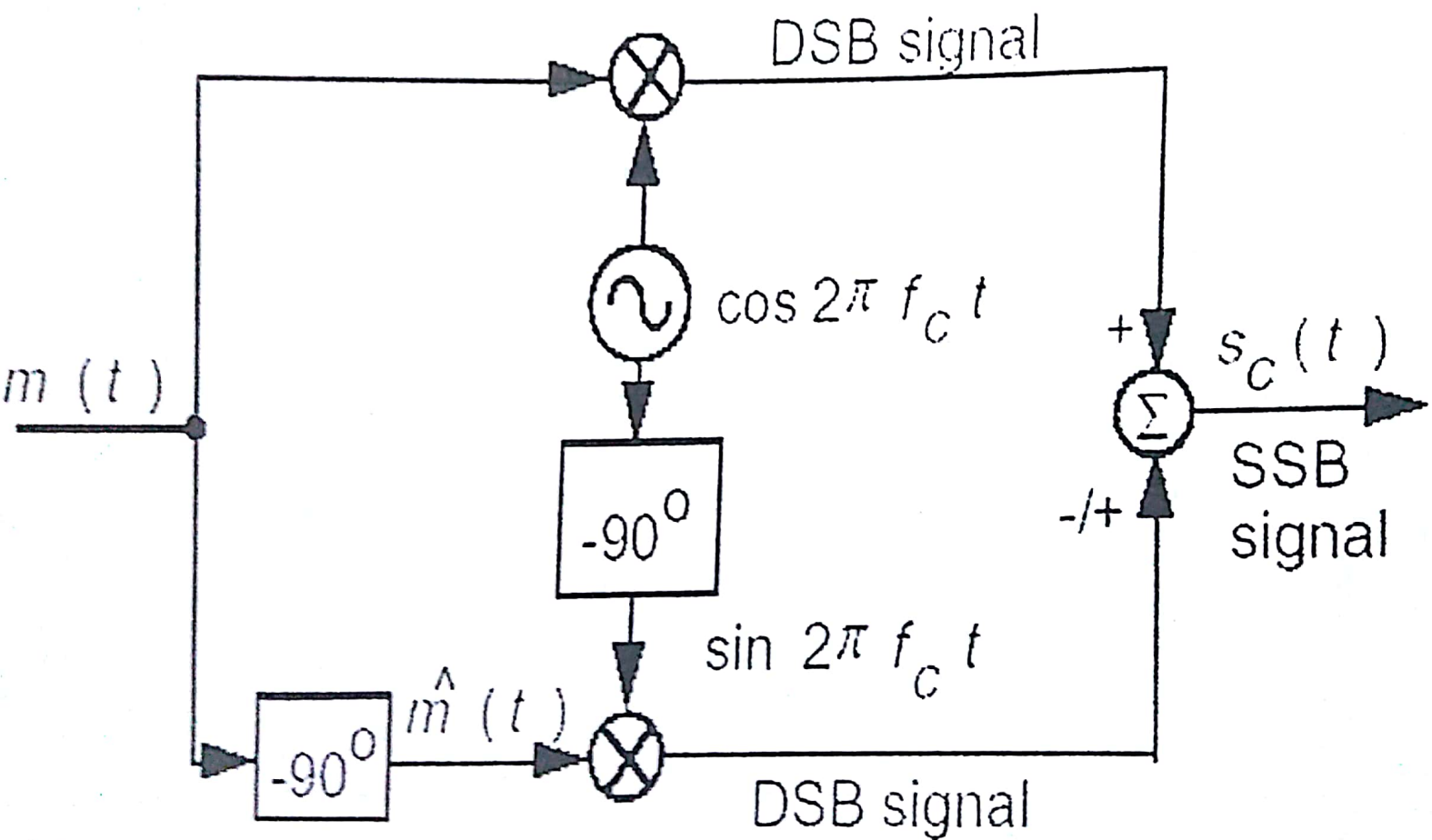


Local carrier

Phase-Shift Method



Phase-Shift Method



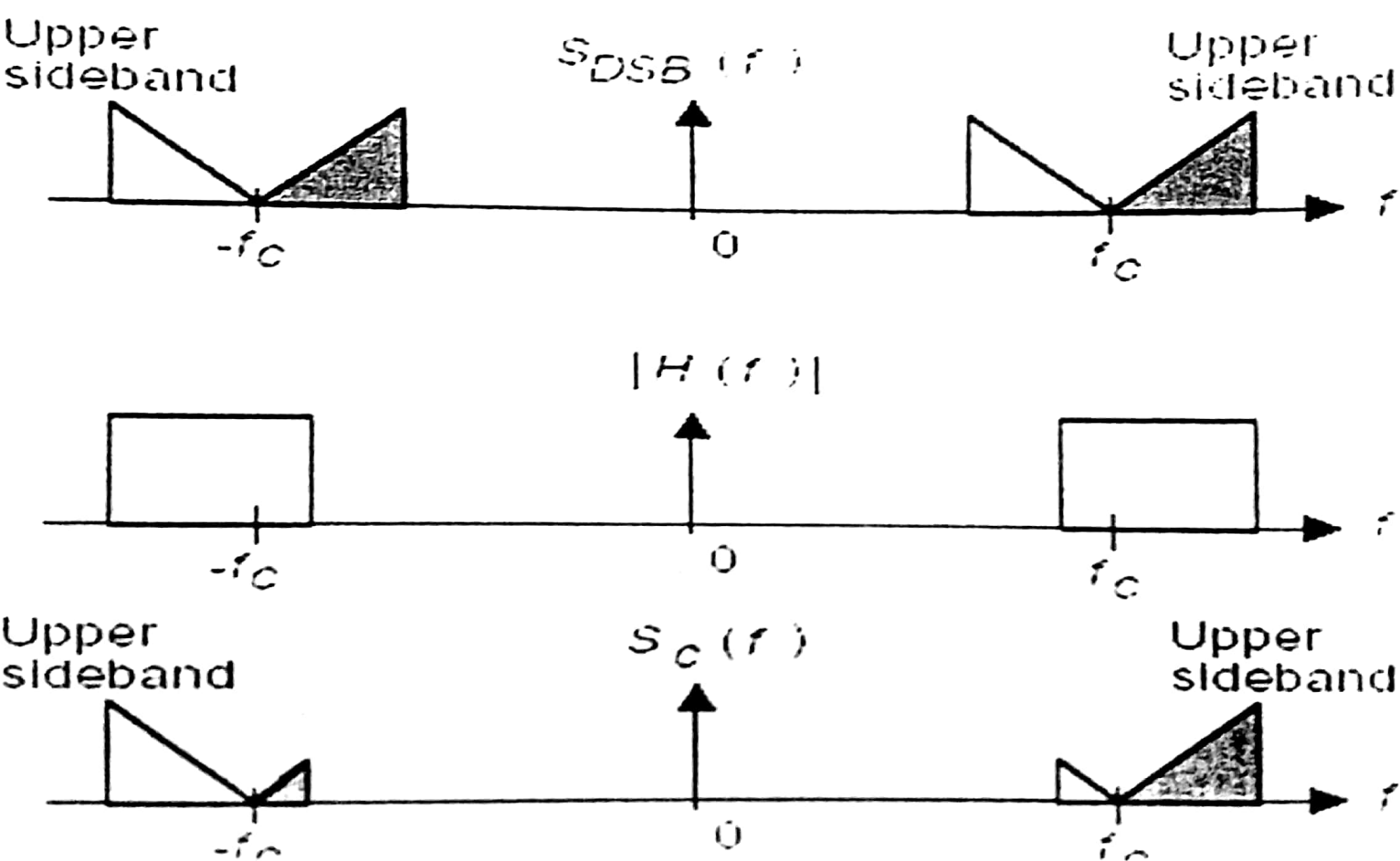
Phase–Shift Method (Cont.)

- **Advantages:**
 - Does not deploy bandpass filter.
 - Suitable for message signals with frequency content down to dc.
- **Disadvantage:**
 - Practical realization of a wideband 90° phase shift circuit is difficult.

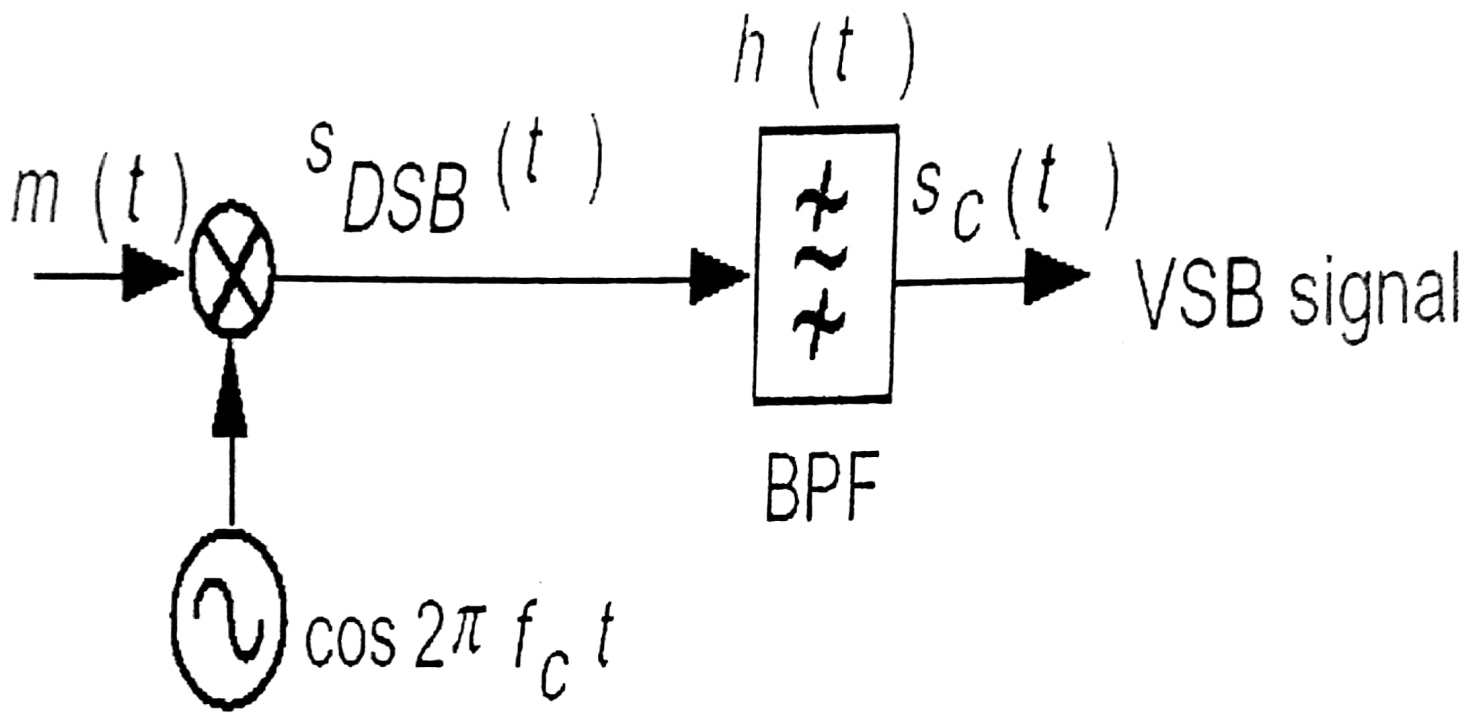
Demodulation of SSB Signals

- Demodulation of SSB signals can be accomplished by using a synchronous detector as used in the demodulation of normal AM and DSBSC signals.
- If we want to use an envelope detector, it can be shown that we must insert a pilot carrier signal $A\cos(2\pi f_c t)$ to the SSB signal, where $A \gg m(t)$ and $A \gg m_h(t)$
- The pilot signal carries most of the transmission power which becomes inefficient.

Vestigial-Sideband Modulation (VSB)

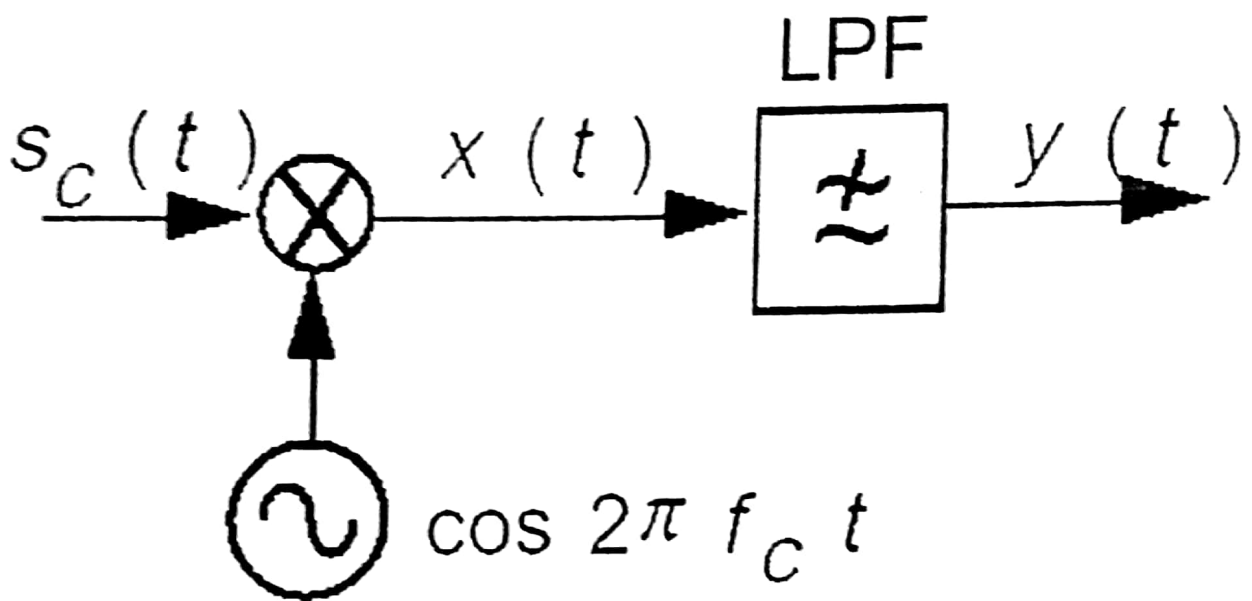


VSB modulator



Demodulation of VSB

- Demodulation of VSB signals can be accomplished by using a synchronous detector.



Local carrier

Transfer function of LPF in VSB receiver

$$S_{VSD}(f) = [M(f + f_c) + M(f - f_c)]H_{BPF}(f)$$

$$X(f) = [S_{VSD}(f + f_c) + S_{VSD}(f - f_c)]$$

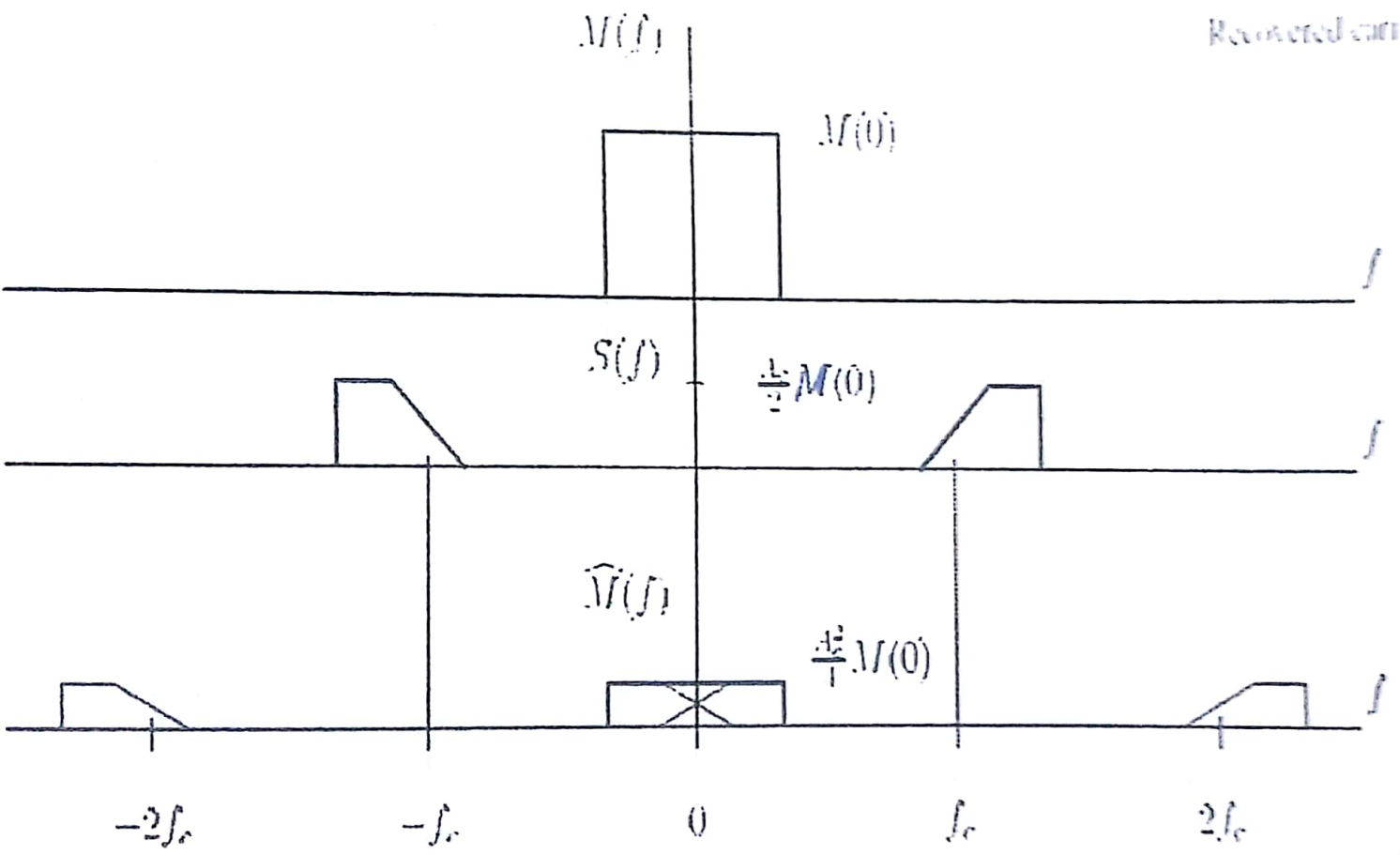
$$Y(f) = M(f) = X(f)H_{LPF}(f)$$

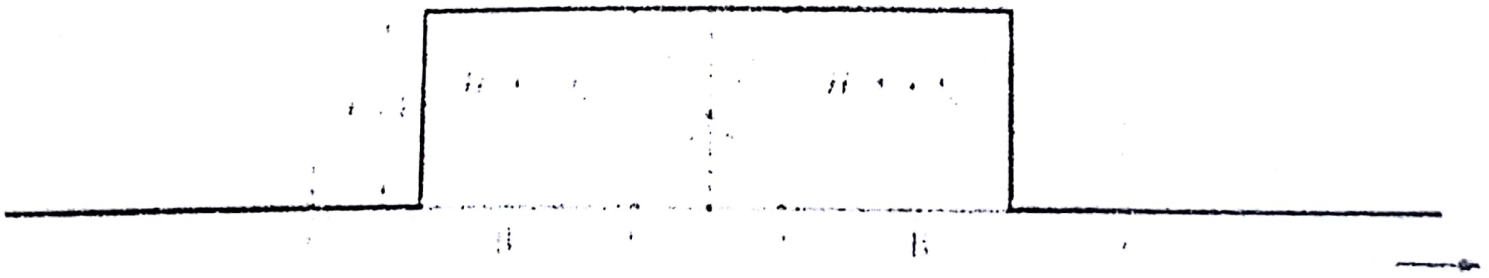
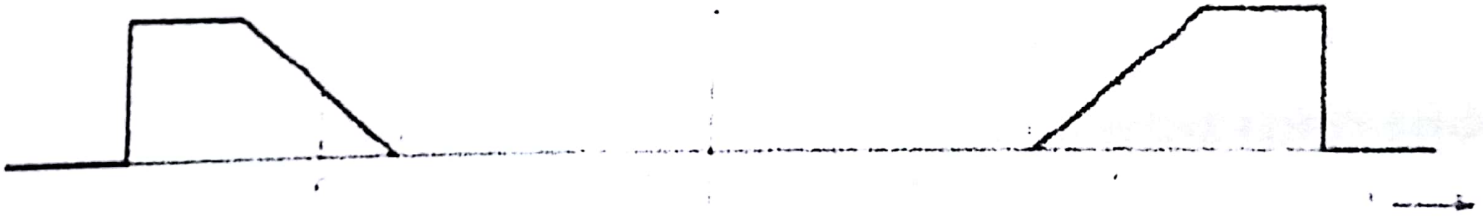
$$= M(f)[H_{BPF}(f + f_c) + H_{BPF}(f - f_c)]H_{LPF}(f)$$

Hence,

$$H_{LPF}(f) = \frac{1}{H_{BPF}(f - f_c) + H_{BPF}(f + f_c)}, |f| \leq B$$

Recovered carrier





VSB+C

- VSB modulated signals can also be detected by an envelope detector.
- As in the demodulation of a SSB signal, we need to send a pilot carrier signal, resulting an inefficient use of available transmitted power.

Comparison of conventional AM, DSB-SC, SSB and VSB.

- Conventional AM: simple to modulate and to demodulate, but low power efficiency (50% max) and double the bandwidth
- DSB-SC: high power efficiency, more complex to modulate & demodulate, double the bandwidth
- SSB: high power efficiency, the same (message) bandwidth, more difficult to modulate & demodulate.
- VSB: lower power efficiency & larger bandwidth but easier to implement.



EE325: Chapter 4 (Lec. #4)

Amplitude Modulations & Demodulations

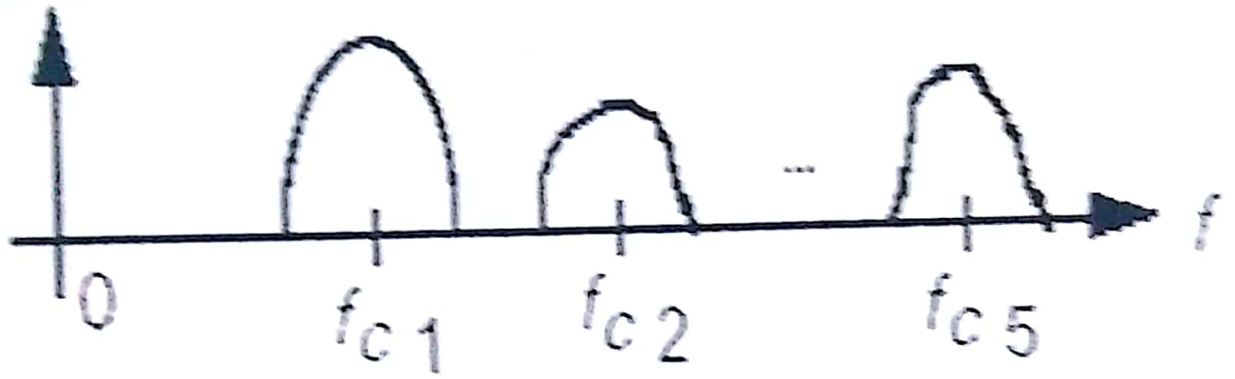
M. A. Smadi

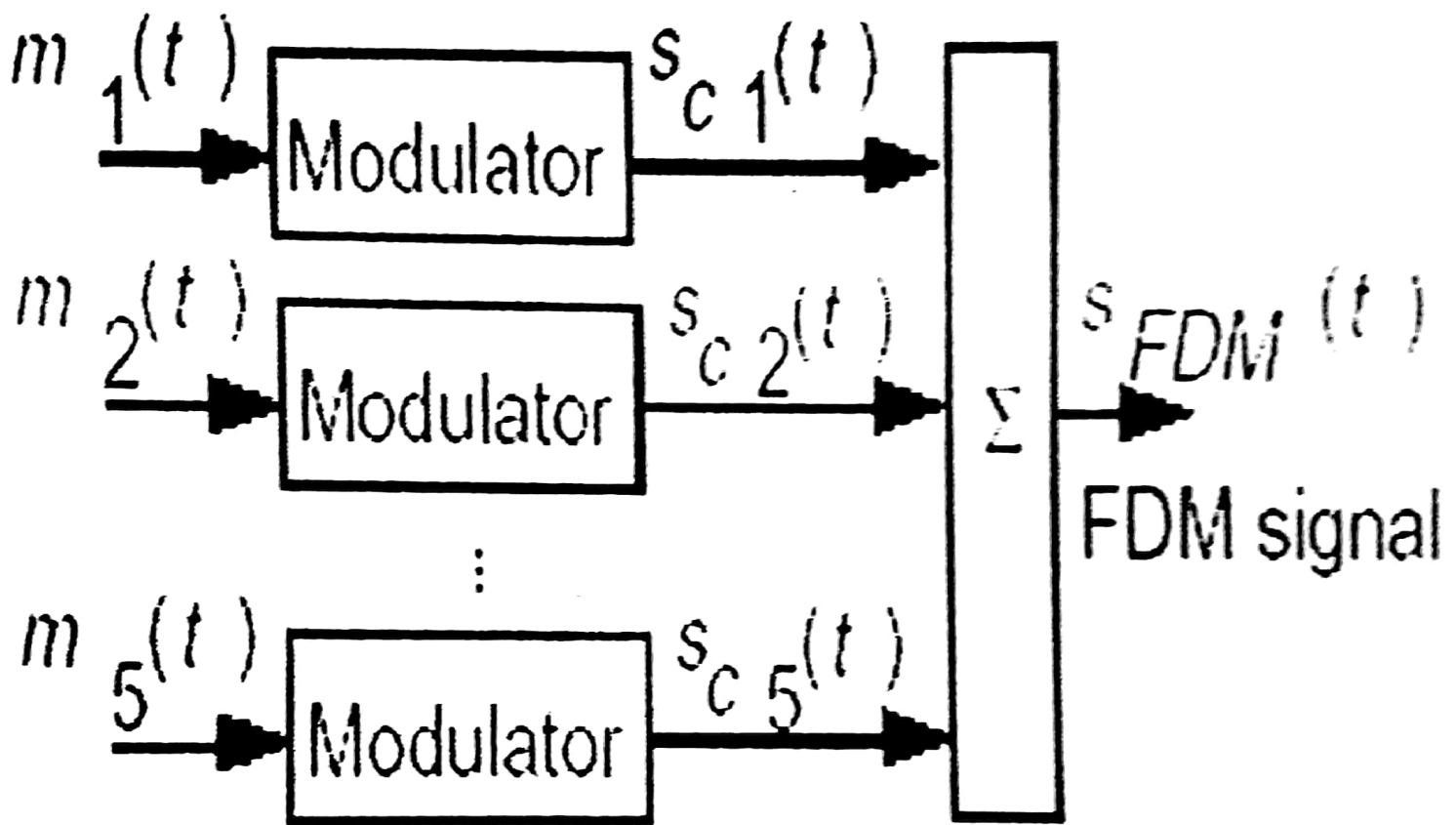
Multiplexing

- Multiplexing: combining a number of message signals into a composite signal to transmit them simultaneously over a wideband channel.
- Two commonly-used types: time-division multiplexing (TDM) and frequency division multiplexing (FDM).
- TDM: transmit different message signals in different time slots (mostly digital).
- FDM: transmit different message signals in different frequency slots (bands) using different carrier frequencies.

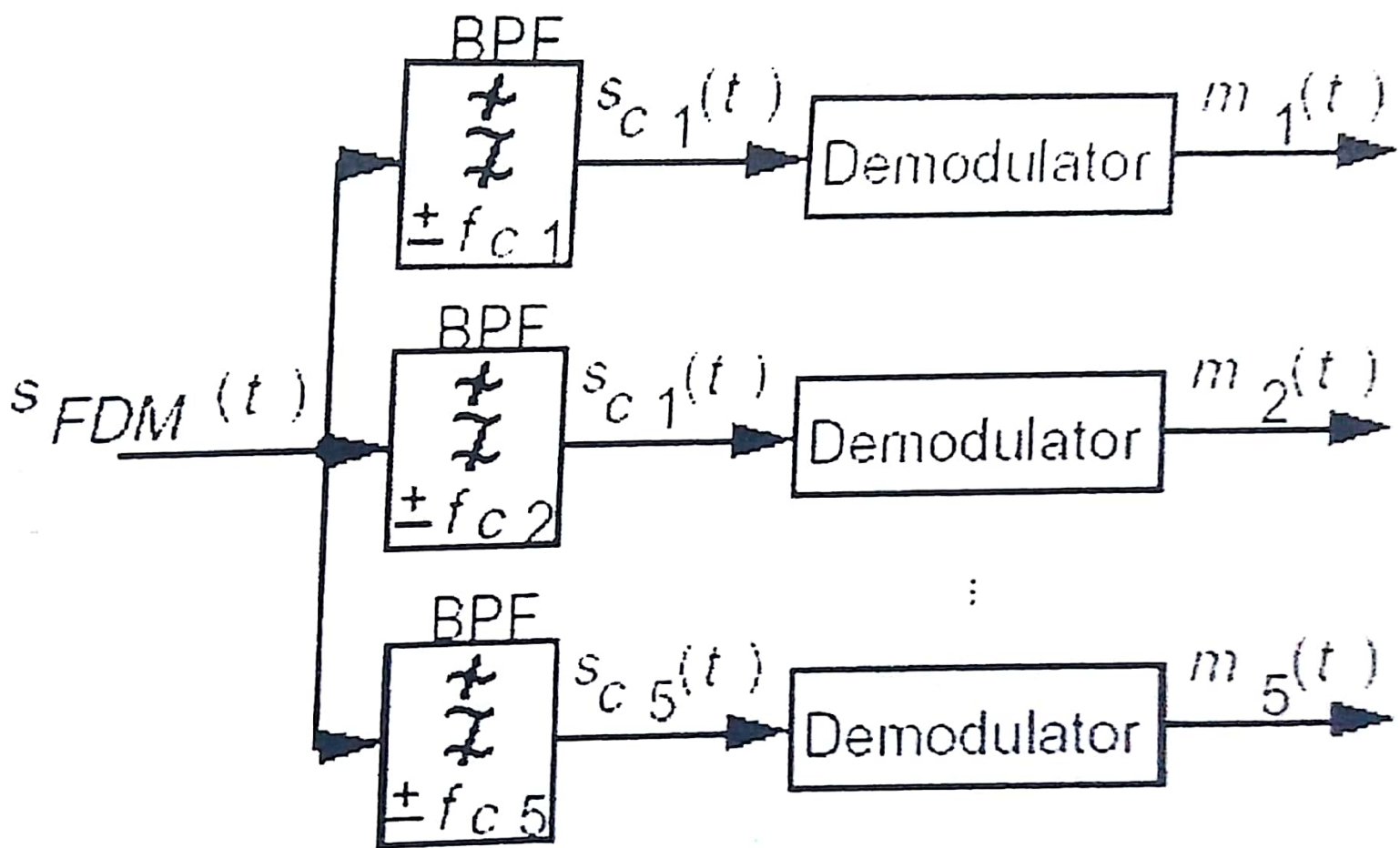
FDM

$|S_{FDM}(f)|$



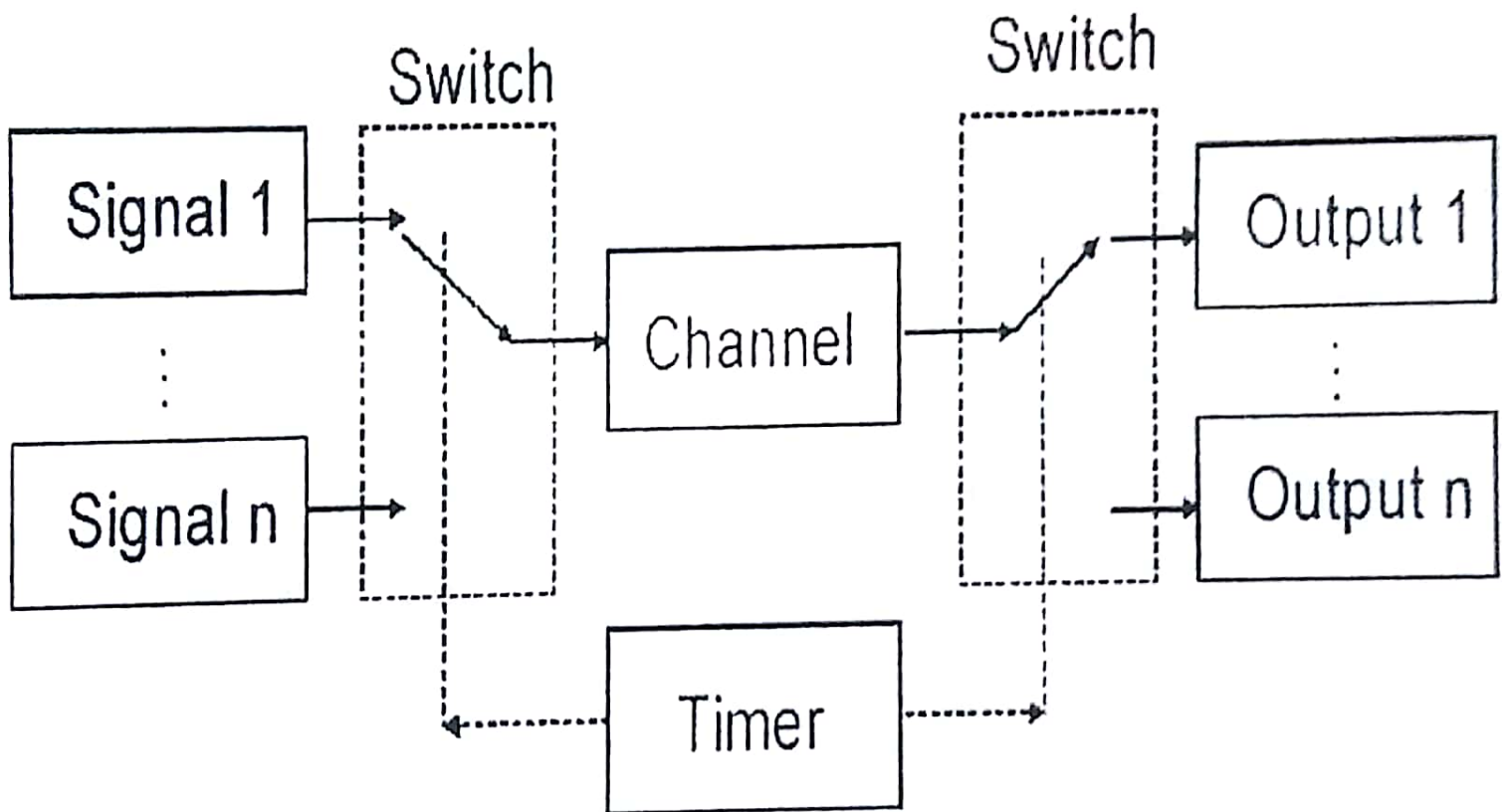


(a) Transmitter

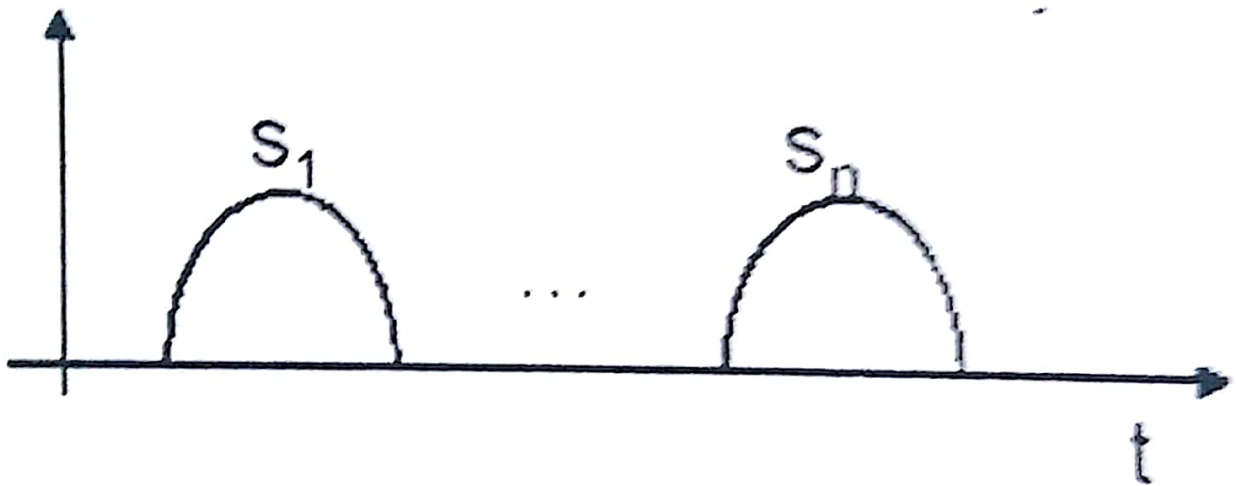


(b) Receiver

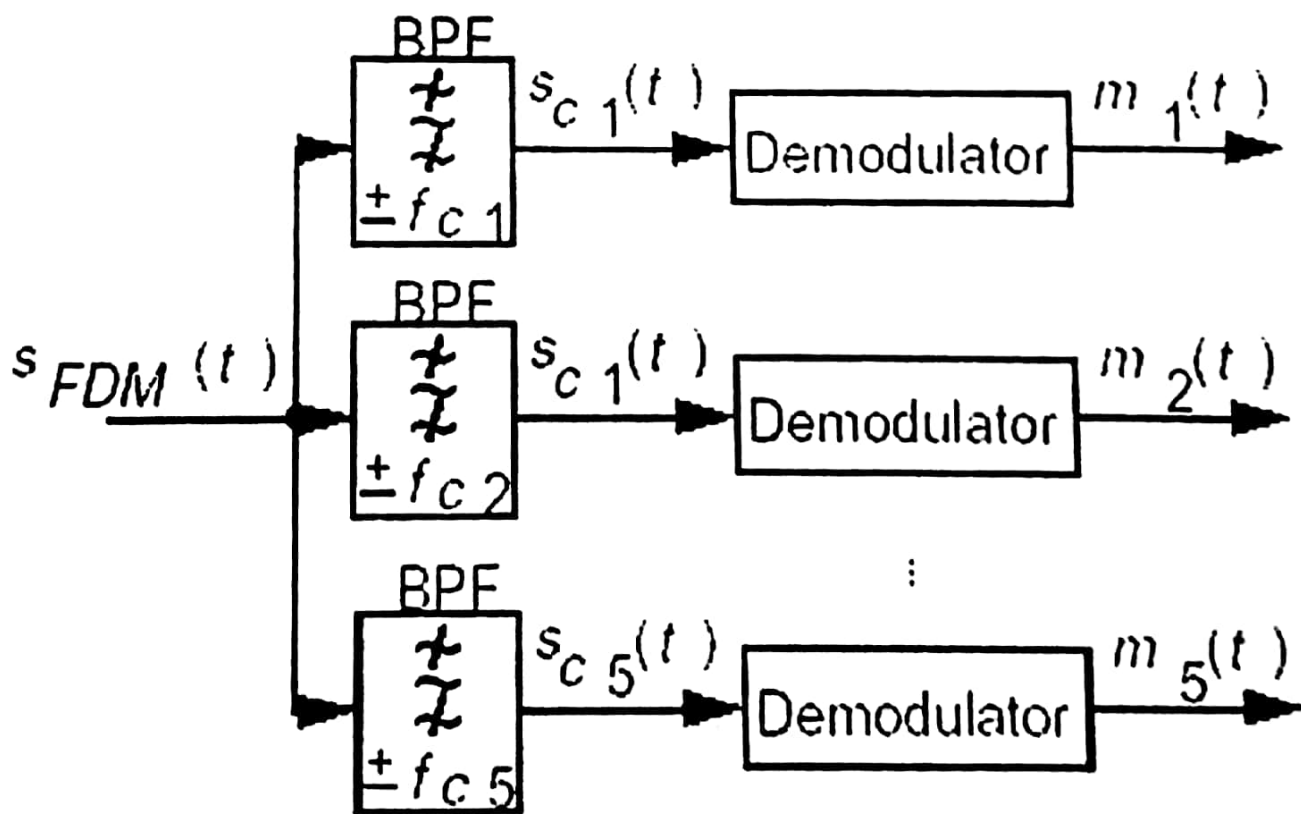
Time Division Multiplexing



TDM



AM receiver for many radio stations ?



(b) Receiver

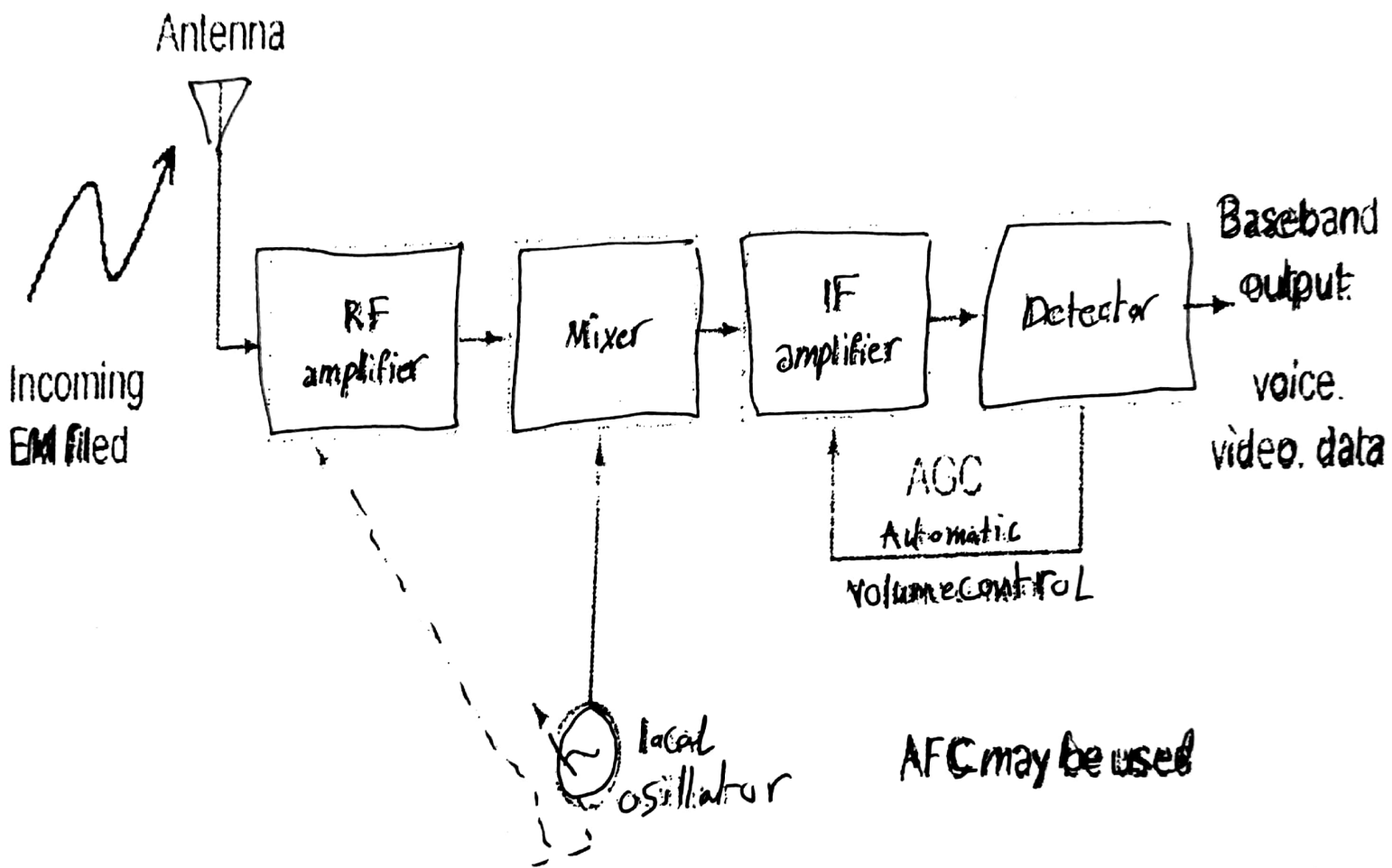
Frequency mixing

- It is desired in communication system to translate the spectrum of the modulated signal up word or down word in frequency to be centered around desired frequency

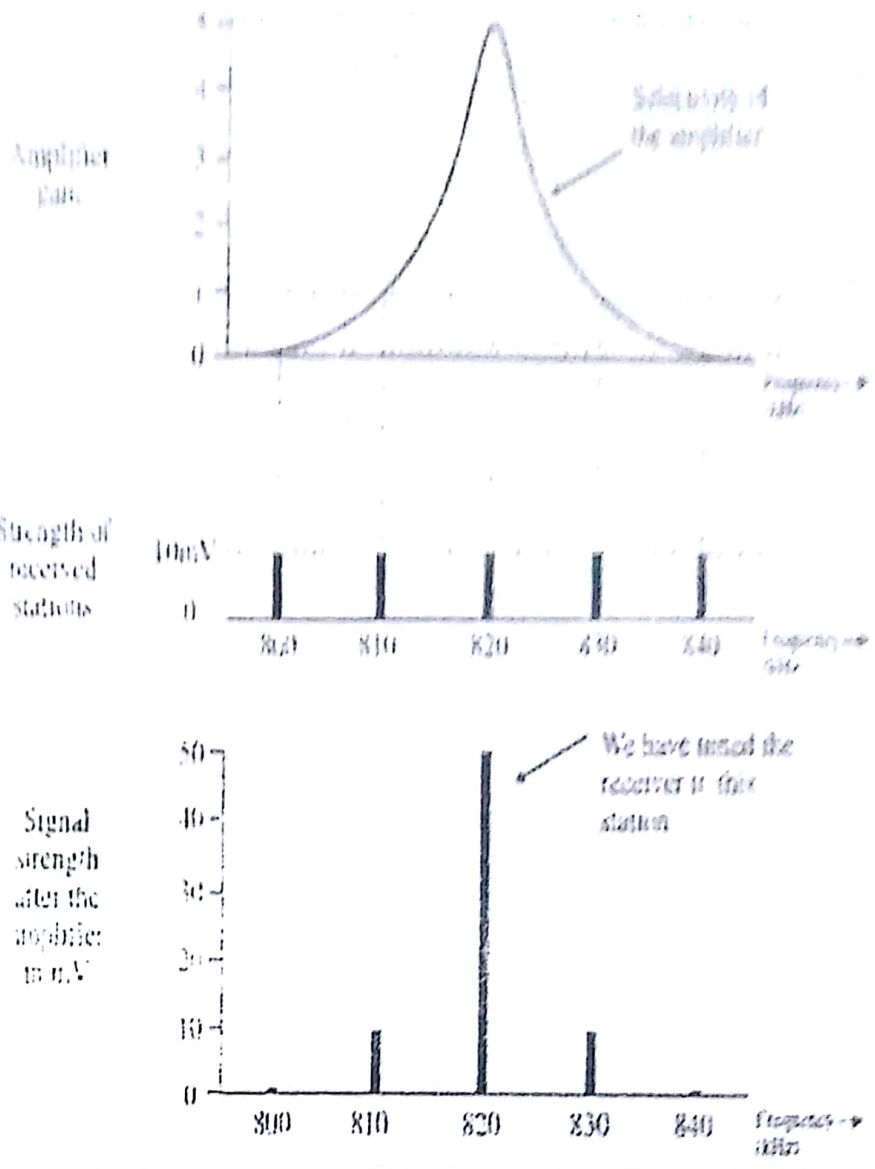
$$f_0 = |f_1 - f_c|$$

$$\Rightarrow f_0 = \begin{cases} f_1 - f_c : \text{up conversion} \\ f_c - f_1 : \text{down conversion} \end{cases}$$

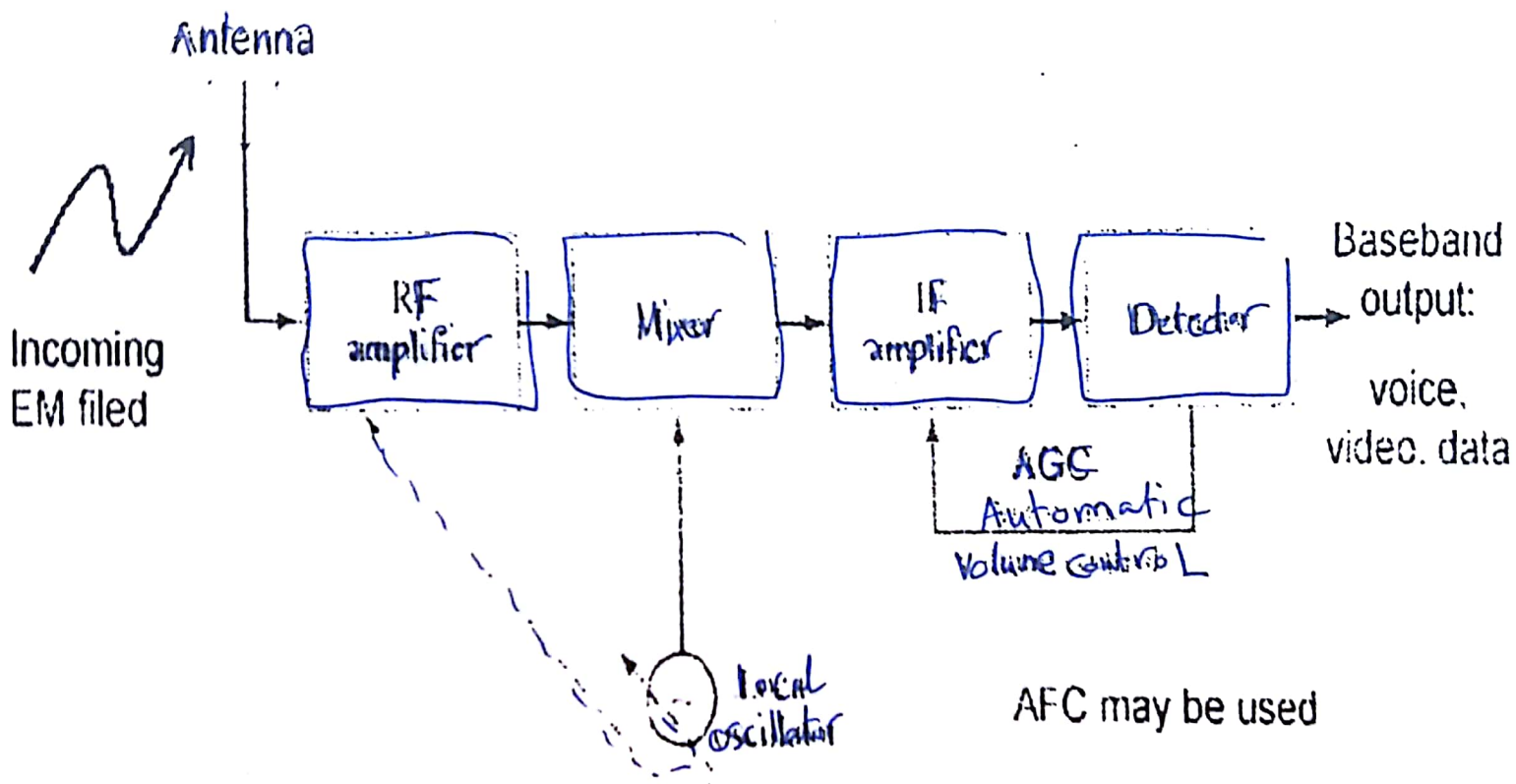
Superheterodyne AM Receiver

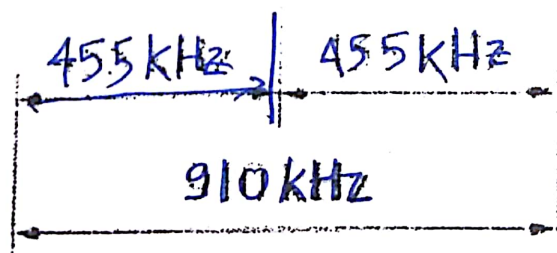
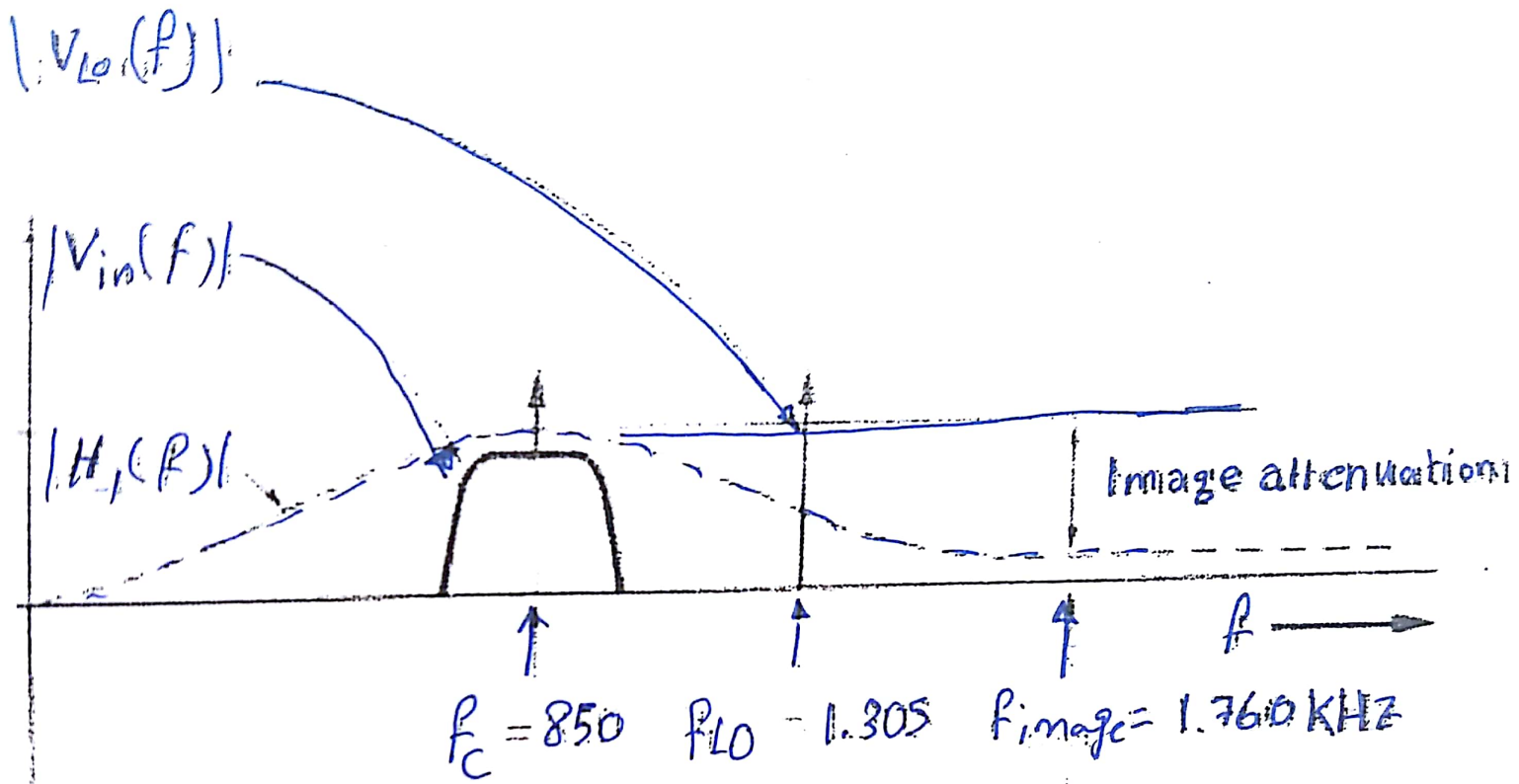


- The RF amplifier amplifies the incoming signal and start the process of selecting the wanted station and rejecting the unwanted ones.



The Mixer and the IF Amplifier



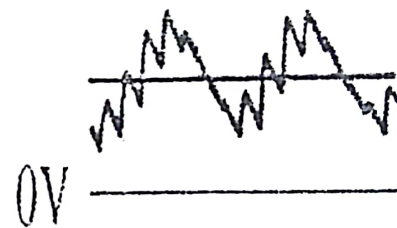


The input to the diode detector
from the last IF amplifier

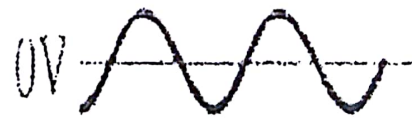


Output of diode detector includes:

- a DC level,
- the audio signal,
- ripple at IF frequency



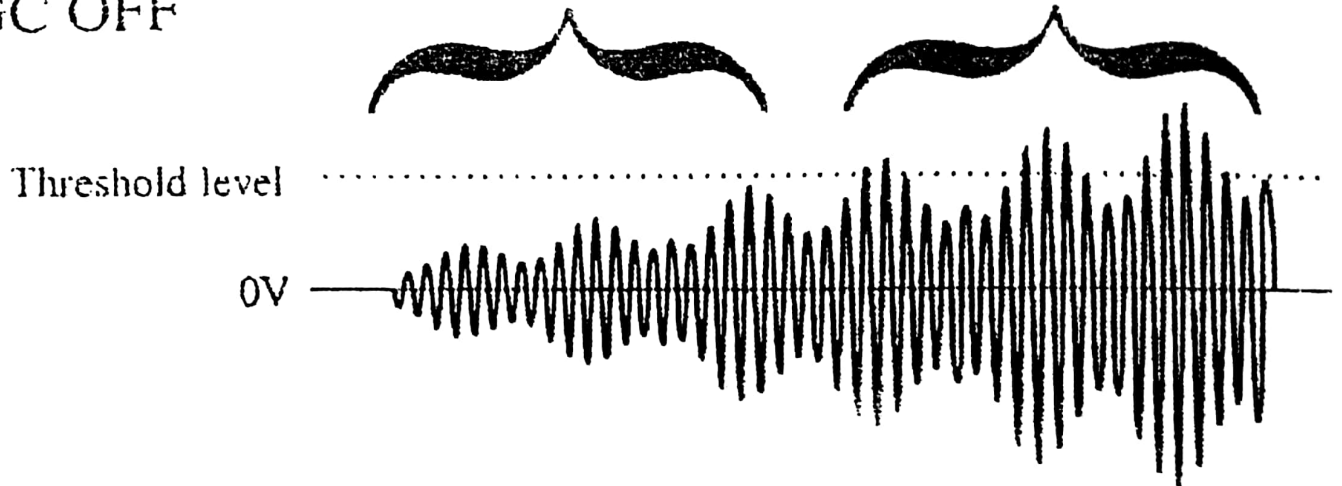
Output after filtering



AGC OFF

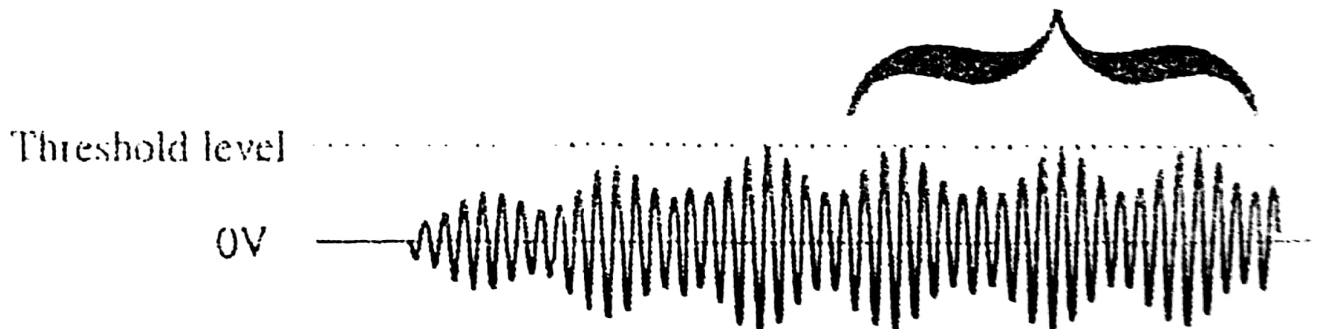
At low signal strength the AGC circuit has no effect

This part of the transmission will overload the receiver and cause distortion



AGC ON

The AGC has limited the amplification to prevent overload and distortion



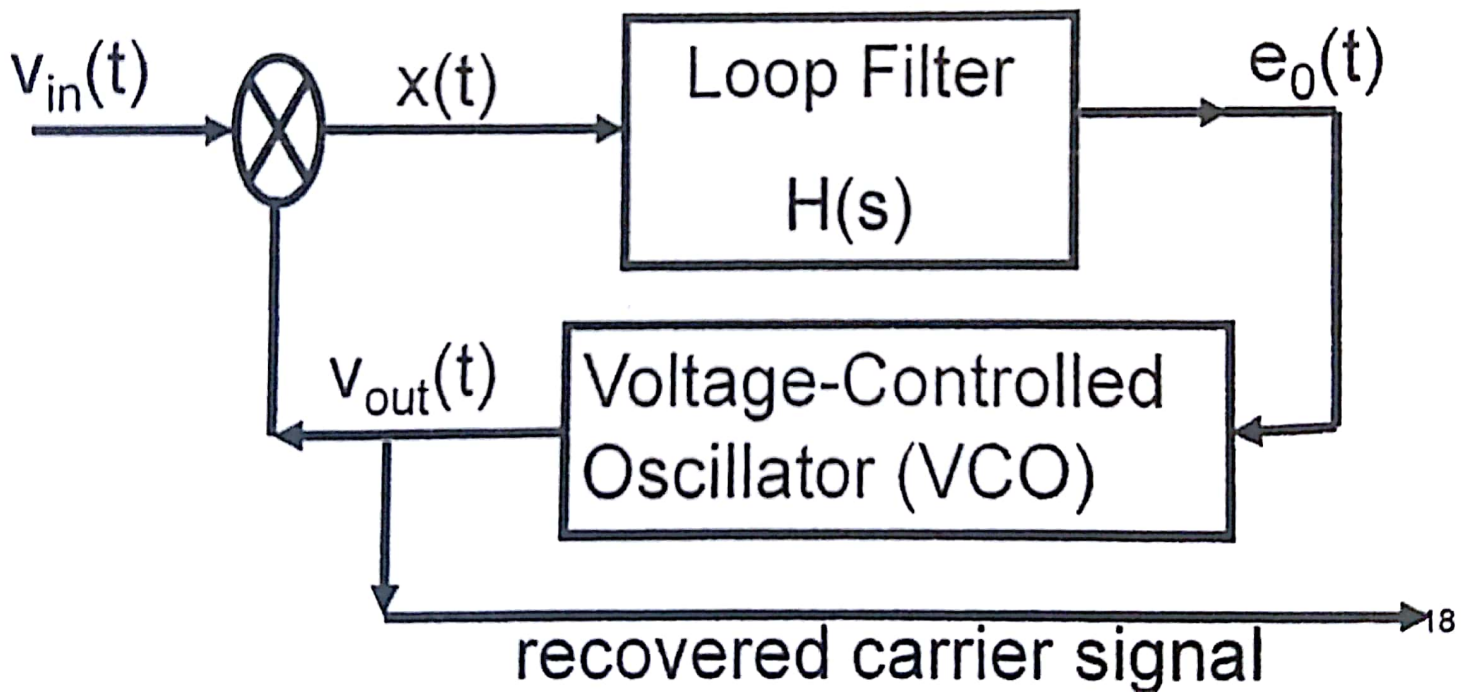
Carrier Acquisition

- To ensure identical carrier frequencies at the transmitter and the receiver, we can use **quartz crystal** oscillators, which are generally very stable.
- At very high carrier frequencies, the quartz-crystal performance may not be adequate, we can use the phased-locked loop (PLL)

Phased-Locked Loop (PLL)

- Phase-locked loop is one of the most commonly used circuit in both telecommunication and measurement engineering.
- PLL can be used to track the phase and the frequency of the carrier component of an incoming signal.

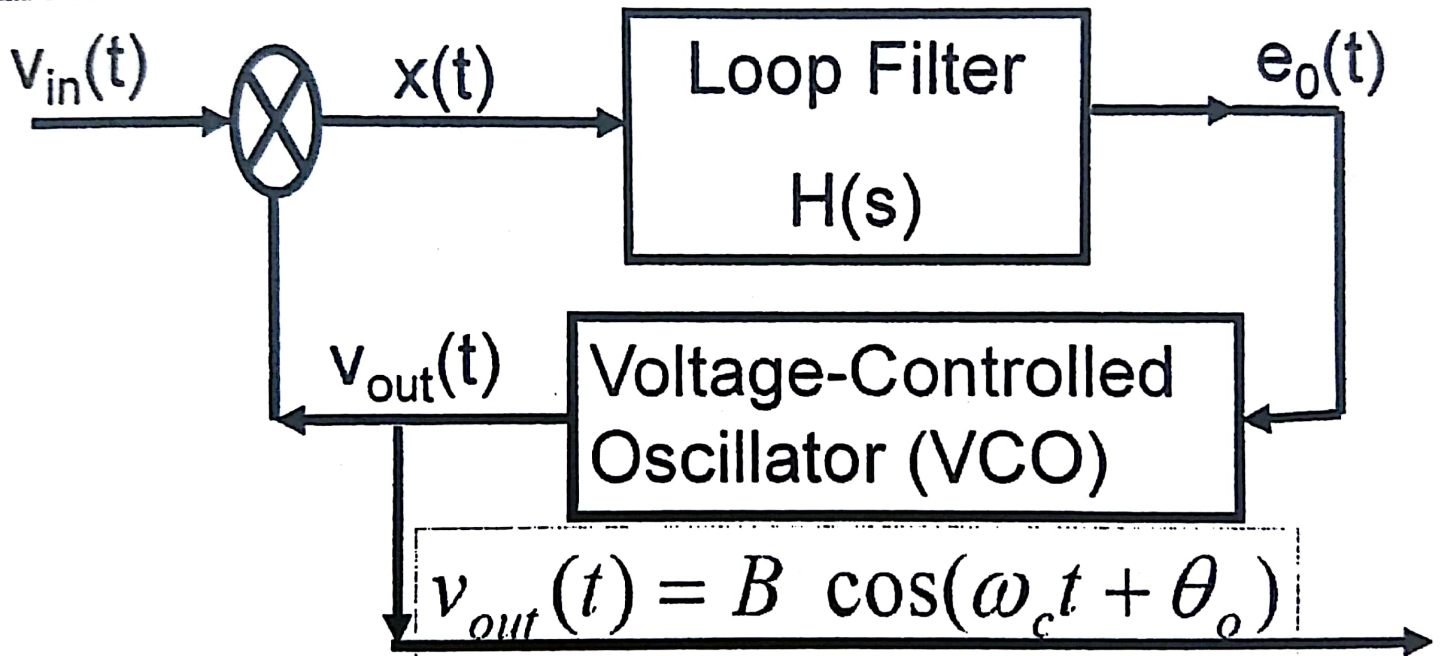
- A PLL has three basic components:
 1. A voltage controlled oscillator
 2. A multiplier
 3. A loop filter $H(s)$



- In every application, the PLL tracks the frequency and the phase of the input signal. However, before a PLL can track, it must first reach the **phase-locked condition**.
- In general, the VCO center frequency differs from the frequency of the input signal.
- First the VCO frequency has to be tuned to the input frequency by the loop. This process is called *frequency pull-in*.
- Then the VCO phase has to be adjusted according to the input phase. This process is known as *phase lock-in*.

How the PLL works?

$$v_{in}(t) = A \sin(\omega_c t + \theta_i)$$



$$x(t) = AB/2 [\sin(\theta_i - \theta_o) + \sin(2\omega_c t + \theta_i + \theta_o)]$$

$$e_0(t) = 0.5 AB \sin(\theta_i - \theta_o) = 0.5 AB \sin(\theta_e)$$



EE325: Chapter 4 (Lec. #5)

Effect of Noise on Analog Communication Systems

M. A. Smadi

Introduction

- Angle modulation systems and FM can provide a high degree of noise immunity
- This noise immunity is obtained at the price of sacrificing channel bandwidth
- Bandwidth requirements of angle modulation systems are considerably higher than that of amplitude modulation systems

EFFECT OF NOISE ON AM SYSTEMS

- Effect of Noise on a Baseband System
- Effect of Noise on DSB-SC AM
- Effect of Noise on Conventional AM

$$r(t) = u(t) + n(t)$$

- $u(t)$: is the Txd signal
- $n(t)$: is the additive White Gaussian noise process (thermal noise) characterized by its flat PSD of

$$S_n(f) = \frac{N_0}{2} \text{ (Watt / Hz)}$$

Effect of Noise on a Baseband System

- Since baseband systems serve as a basis for comparison of various modulation systems, we begin with a noise analysis of a baseband system.
- In this case, there is no carrier demodulation to be performed.
- The receiver consists only of an ideal low pass filter with the bandwidth W .
- The baseband noise power at the output of the receiver, for a white noise input, is

$$P_n = \int_{-W}^{W} \frac{N_0}{2} df = N_0 W$$

- If we denote the received power by P_R , the baseband SNR is given by

$$\left(\frac{S}{N} \right)_b = \frac{P_R}{N_0 W}$$

White Noise Process

- White process is processes in which all frequency components appear with equal power, i.e., the power spectral density (PSD), $S_x(f)$, is a constant for all frequencies.
- the PSD of thermal noise, $S_n(f)$, is usually given as $S_n(f) = \frac{kT}{2}$ (where k is Boltzmann's constant and T is the temperature)
- The value kT is usually denoted by N_0 , Then $S_n(f) = \frac{N_0}{2}$

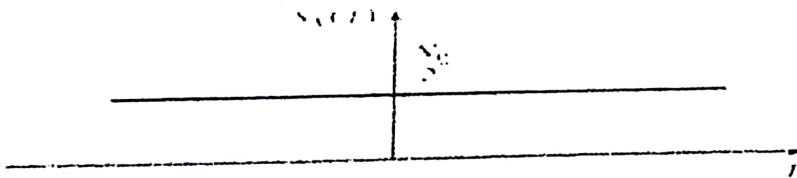


Figure 5.19 Power spectrum of a white process.

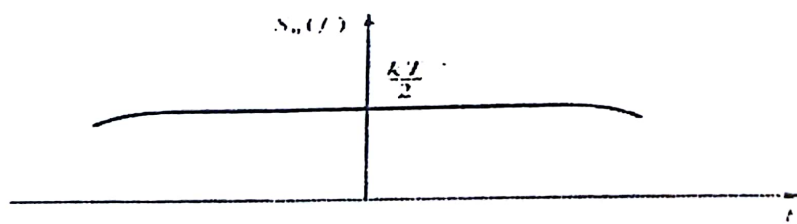
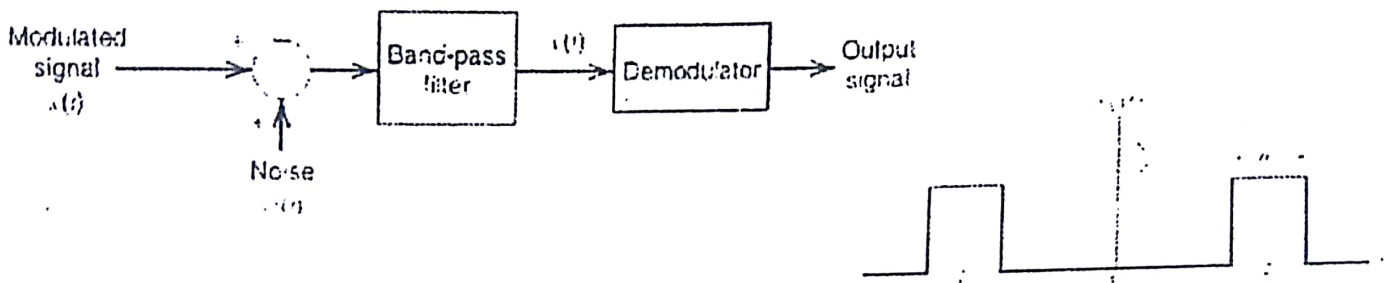


Figure 5.20 Power spectrum of thermal noise.

Effect of Noise on DSB-SC AM



- Transmitted signal : $u(t) = A_c m(t) \cos(2\pi f_c t)$
- A filtered noise process can be expressed in terms of its in-phase and quadrature components as

$$n(t) = A(t) \cos[2\pi f_c t + \theta(t)] = A(t) \cos \theta(t) \cos(2\pi f_c t) - A(t) \sin \theta(t) \sin(2\pi f_c t)$$

$$= n_c(t) \cos(2\pi f_c t) - n_s(t) \sin(2\pi f_c t)$$

- where $n_c(t)$ is in-phase component and $n_s(t)$ is quadrature component

Effect of Noise on DSB-SC AM

- Received signal (Adding the filtered noise to the modulated signal)

$$r(t) = u(t) + n(t)$$

$$= A_c m(t) \cos(2\pi f_c t) + n_c(t) \cos(2\pi f_c t) - n_s(t) \sin(2\pi f_c t)$$

- Demodulate the received signal by first multiplying $r(t)$ by a locally generated sinusoid $\cos(2\pi f_c t)$, where θ is the phase of the sinusoid (coherent detection).
- Then passing the product signal through an ideal lowpass filter having a bandwidth W .

Effect of Noise on DSB-SC AM

- The multiplication of $r(t)$ with $\cos(2\pi f_c t)$ yields

$$\begin{aligned}r(t) \cos(2\pi f_c t) &= u(t) \cos(2\pi f_c t) + n(t) \cos(2\pi f_c t) \\ &= A_c m(t) \cos(2\pi f_c t) \cos(2\pi f_c t) \\ &\quad + n_c(t) \cos(2\pi f_c t) \cos(2\pi f_c t) - n_s(t) \sin(2\pi f_c t) \cos(2\pi f_c t) \\ &= \frac{1}{2} A_c m(t) + \frac{1}{2} A_c m(t) \cos(4\pi f_c t) \\ &\quad + \frac{1}{2} n_c(t) + \frac{1}{2} [n_c(t) \cos(4\pi f_c t) - n_s(t) \sin(4\pi f_c t)]\end{aligned}$$

- The lowpass filter rejects the double frequency components and passes only the lowpass components:

$$y(t) = \frac{1}{2} A_c m(t) + \frac{1}{2} n_c(t)$$

Effect of Noise on DSB-SC AM

- Therefore, at the receiver output, the message signal and the noise components are additive and we are able to define a meaningful SNR. The message signal power is given by

$$P_o = \frac{A_c^2}{4} P_M = \frac{1}{2} P_R$$

– power P_M is the content of the message signal

- The noise power is given by

$$P_{n_b} = \frac{1}{4} P_{n_c} = \frac{1}{4} P_n$$

- The power content of $n(t)$ can be found by noting that it is the result of passing $n_c(t)$ through a filter with bandwidth $2W$.

Effect of Noise on DSB-SC AM

- Therefore, the PSD of $n(t)$ is given by

$$S_n(f) = \begin{cases} \frac{N_0}{2} & |f - f_c| < W \\ 0 & \text{otherwise} \end{cases}$$

- The bandpass noise power is

$$P_n = \int_{-\infty}^{\infty} S_n(f) df = \frac{N_0}{2} \times 4W = 2WN_0$$

- Now we can find the output SNR as

$$\left(\frac{S}{N}\right)_o = \frac{P_o}{P_{n_o}} = \frac{\frac{A_o^2}{4} P_M}{\frac{1}{4} 2WN_0} = \frac{A_o^2 P_M}{2WN_0}$$

- In this case, the received signal power is

$$P_R = A_c^2 P_M / 2$$

10

Effect of Noise on DSB-SC AM

- The output SNR for DSB-SC AM may be expressed as

$$\left(\frac{S}{N}\right)_{\text{DSB}} = \frac{P_R}{N_0 W}$$

- which is identical to baseband SNR.
- In DSB-SC AM, the output SNR is the same as the SNR for a baseband system
⇒ DSB-SC AM does not provide any SNR improvement over a simple baseband communication system

”

Effect of Noise on Conventional AM

- DSB AM signal : $u(t) = A_c[1 + am(t)]\cos(2\pi f_c t)$

- Received signal at the input to the demodulator

$$\begin{aligned}r(t) &= A_c[1 + am(t)]\cos(2\pi f_c t) + n(t) \\ &= A_c[1 + am(t)]\cos(2\pi f_c t) + n_c(t)\cos(2\pi f_c t) - n_s(t)\sin(2\pi f_c t) \\ &= [A_c[1 + am(t)] + n_c(t)]\cos(2\pi f_c t) - n_s(t)\sin(2\pi f_c t)\end{aligned}$$

- a is the modulation index

- $m(t)$ is normalized so that its minimum value is -1

- If a synchronous demodulator is employed, the situation is basically similar to the DSB case, except that we have $1 + am(t)$ instead of $m(t)$.

- After mixing and lowpass filtering

- $y(t) = \frac{1}{2}[A_c am(t) + n_c(t)]$

Effect of Noise on Conventional AM

- Received signal power

$$P_R = \frac{A_c^2}{2} [1 + a^2 P_M]$$

- Assumed that the message process is zero mean.

- Now we can derive the output SNR as

$$\begin{aligned} \left(\frac{S}{N}\right)_{AM} &= \frac{\frac{1}{4} A_c^2 a^2 P_M}{\frac{1}{4} P_n} = \frac{A_c^2 a^2 P_M}{2 N_0 W} = \frac{a^2 P_M}{1 + a^2 P_M} \frac{\frac{A_c^2}{2} [1 + a^2 P_M]}{N_0 W} \\ &= \frac{a^2 P_M}{1 + a^2 P_M} \frac{P_R}{N_0 W} = \frac{a^2 P_M}{1 + a^2 P_M} \left(\frac{S}{N}\right)_b = \eta \left(\frac{S}{N}\right)_b \end{aligned}$$

- η denotes the modulation efficiency
- Since $a^2 P_M < 1 + a^2 P_M$, the SNR in conventional AM is always smaller than the SNR in a baseband or DSB systems.

Effect of Noise on Conventional AM

- In practical applications, the modulation index a is in the range of 0.8-0.9.
- Power content of the normalized message process depends on the message source.
- Speech signals : Large dynamic range, P_M is about 0.1.
 - The overall loss in SNR, when compared to a baseband system, is a factor of 0.075 or equivalent to a loss of 11 dB.
- **The reason for this loss** is that a large part of the transmitter power is used to send the carrier component of the modulated signal and not the desired signal.
- To analyze the envelope-detector performance in the presence of noise, we must use certain approximations.
 - This is a result of the nonlinear structure of an envelope detector, which makes an exact analysis difficult.

Effect of Noise on Conventional AM

- In this case, the demodulator detects the envelope of the received signal and the noise process.
- The input to the envelope detector is

$$r(t) = [A_c[1 + am(t)] + n_c(t)]\cos(2\pi f_c t) - n_s(t)\sin(2\pi f_c t)$$

- Therefore, the envelope of $r(t)$ is given by

$$V_r(t) = \sqrt{[A_c[1 + am(t)] + n_c(t)]^2 + n_s^2(t)}$$

- Now we assume that the signal component in $r(t)$ is much stronger than the noise component. Then

$$P(n_c(t) \ll A_c[1 + am(t)]) \approx 1$$

- Therefore, we have a high probability that

$$V_r(t) \approx A_c[1 + am(t)] + n_c(t)$$

Effect of Noise on Conventional AM

- After removing the DC component, we obtain

$$y(t) = A_c am(t) + n_c(t)$$

- which is basically the same as $y(t)$ for the synchronous demodulation without the $\frac{1}{2}$ coefficient.
- This coefficient, of course, has no effect on the final SNR.
- So we conclude that, under the assumption of high SNR at the receiver input, the performance of synchronous and envelope demodulators is the same in terms of power efficiency.



EE325: Chapter 5 (Lec. #1)

ANGLE MODULATION & DEMODULATIONS

M. A. Smadi

Outlines

Introduction

Concepts of instantaneous frequency

Bandwidth of angle modulated signals

Narrow-band and wide-band frequency modulations

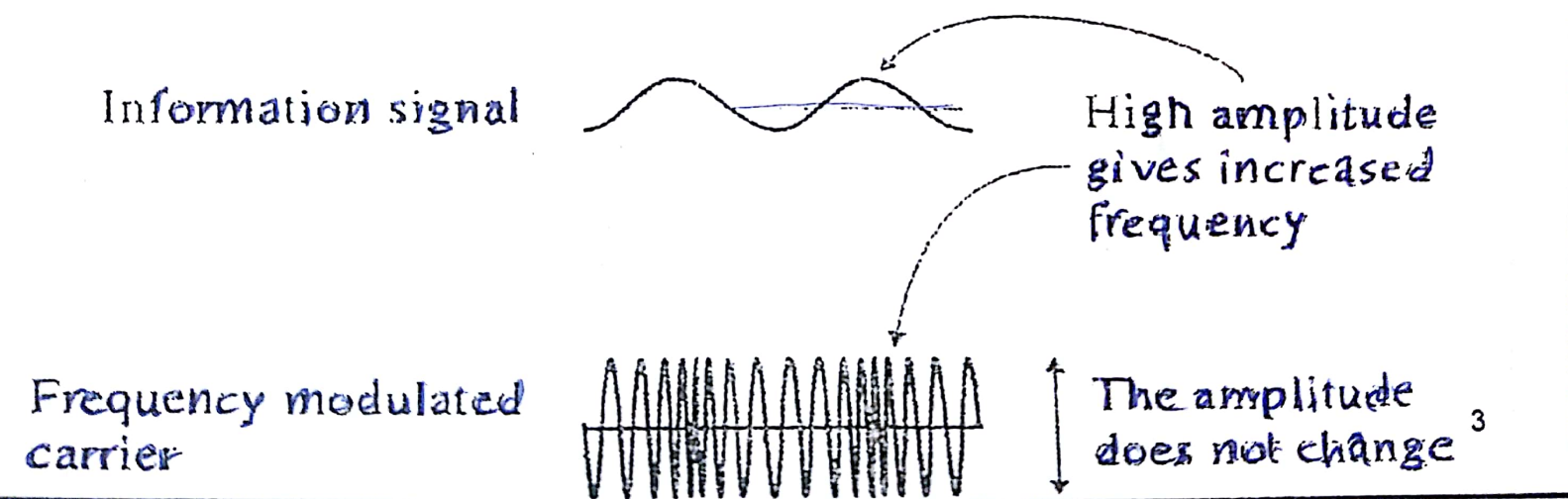
Generation of FM signals

Demodulation of FM signals

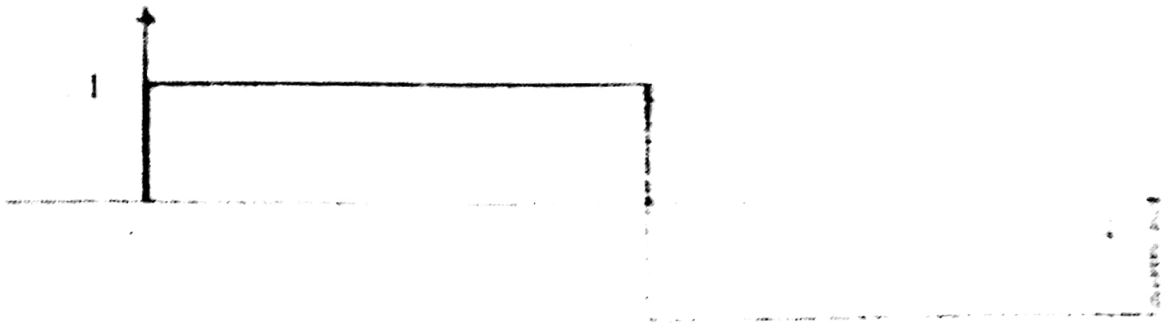
superhetrodyne FM radio

Introduction

- Angle modulation: either frequency modulation (FM) or phase modulation (PM).
- Basic idea: vary the carrier frequency (FM) or phase (PM) according to the message signal.



rectangular pulse



FM
signal



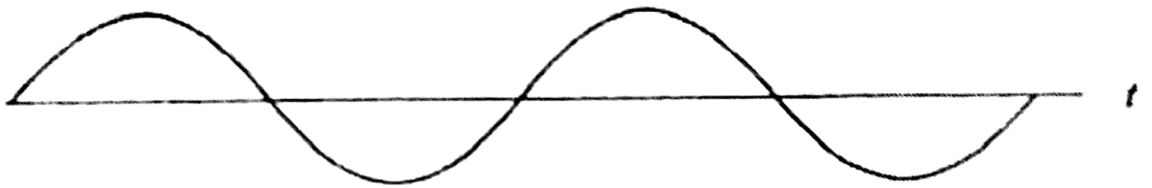
PM
signal



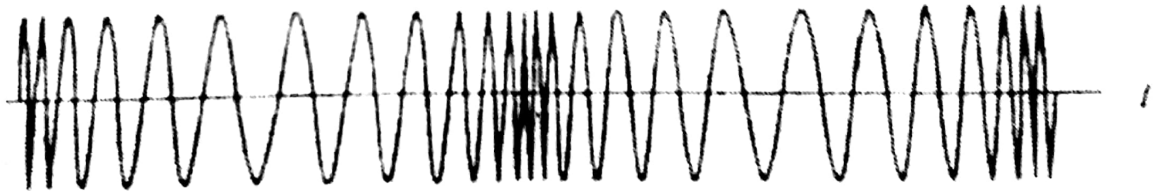
Carrier



Message



PM signal



FM signal



- While AM is linear process, FM and PM are highly nonlinear.
- FM/PM provide many advantages (main – noise immunity, interference^{تداخل}, exchange of power with bandwidth) over AM, at a cost of larger transmission bandwidth.
- Demodulation may be complex, but modern ICs allow cost-effective implementation.
Example: FM radio (high quality, not expensive receivers).

Concepts of Instantaneous Frequency

- A general form of an angle modulated signal is given by

$$S(t) = A \cos \theta_i(t) = A \cos(2\pi f_c t + \phi_i(t))$$

$\theta_i(t)$ is the instantaneous angle

$\phi_i(t)$ is the instantaneous phase deviation.

- The instantaneous angular frequency of $S(t)$

$$\omega_i(t) = \frac{d\theta_i(t)}{dt} = \omega_c + \frac{d\phi_i(t)}{dt}$$

- The instantaneous frequency of $S(t)$

$$f_i(t) = \frac{1}{2\pi} \frac{d\theta_i(t)}{dt} = f_c + \frac{1}{2\pi} \frac{d\phi_i(t)}{dt}$$

- The ^{maximum} instantaneous frequency deviation

$$\Delta f_i(t) = \frac{1}{2\pi} \frac{d\phi_i(t)}{dt}$$

Example

- for the signal below find the instantaneous frequency and maximum frequency deviation.

$$x(t) = A \cos(10 \pi t + \pi t^2)$$

$$f_i(t) = 5 + t$$

$$\Delta f_i(t) = t$$

Phase modulation (PM)

- For phase modulation (PM), the instantaneous phase deviation is $\phi_i(t) = k_p m(t)$

$$S_{PM}(t) = A \cos [2\pi f_c t + k_p m(t)]$$

- k_p is the *phase sensitivity* of the PM modulator expressed in (rad/ V) if $m(t)$ is in Volts
- The instantaneous frequency of $S_{PM}(t)$

$$f_i(t) = f_c + \frac{k_p}{2\pi} \frac{dm(t)}{dt}$$

Frequency Modulation (FM)

- For Frequency Modulation (FM), the instantaneous phase deviation is

$$\phi_i(t) = k_f \int m(\alpha) d\alpha$$

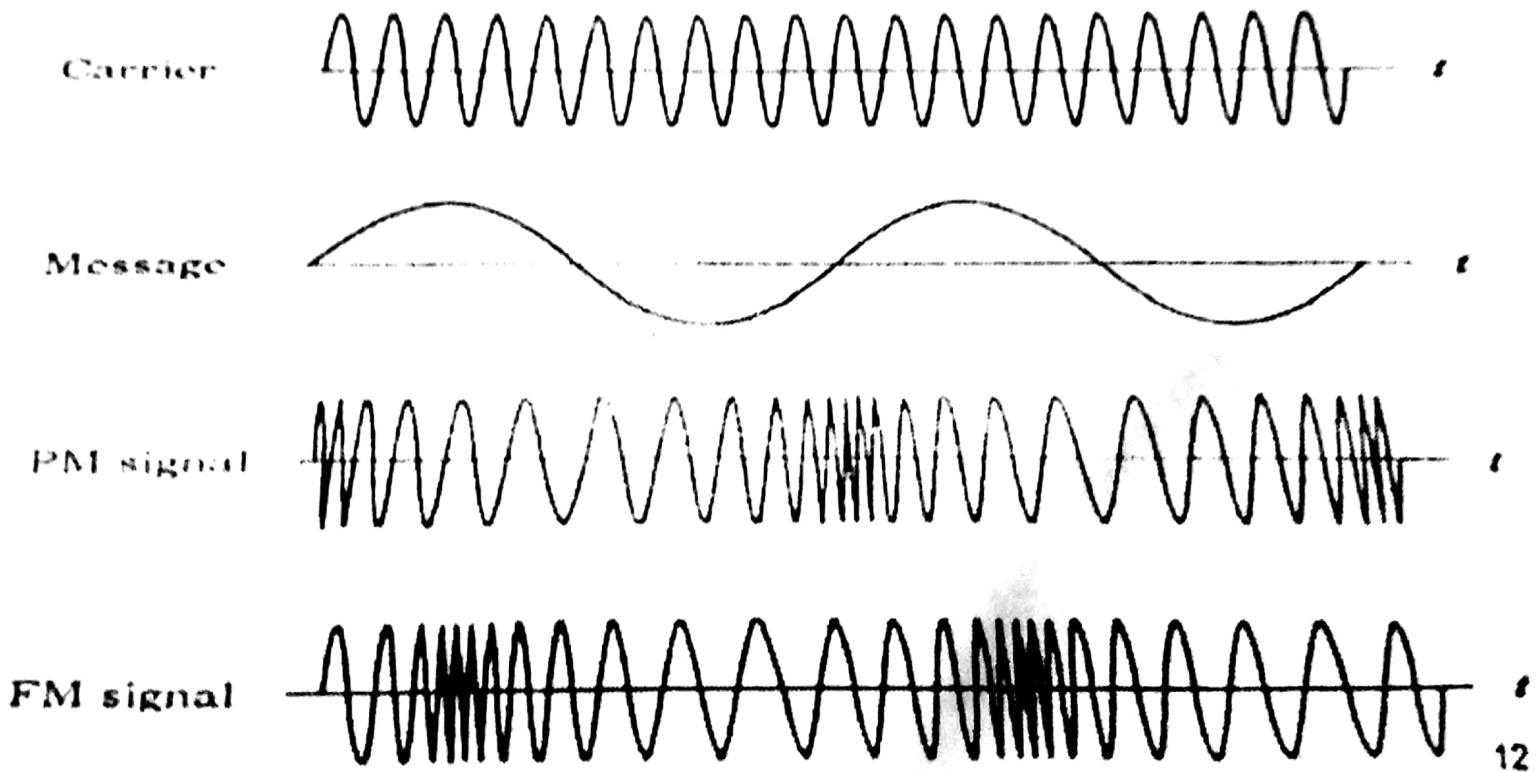
$$S_{FM}(t) = A \cos \left[2\pi f_c t + k_f \int m(\alpha) d\alpha \right]$$

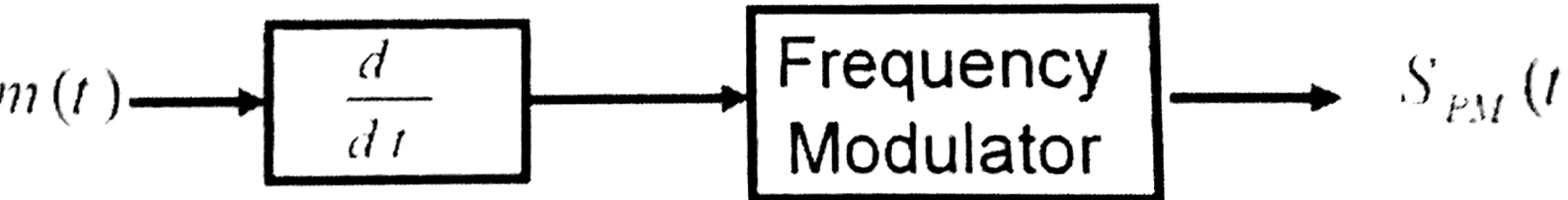
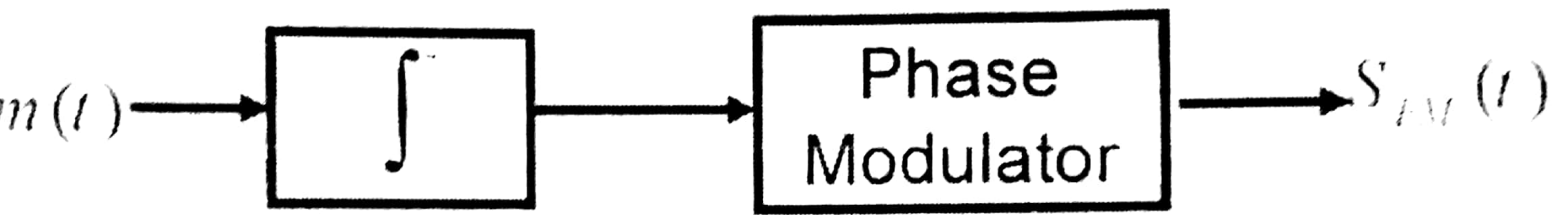
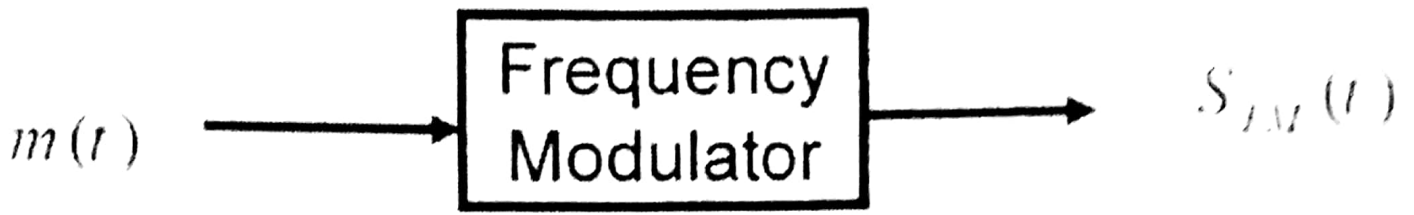
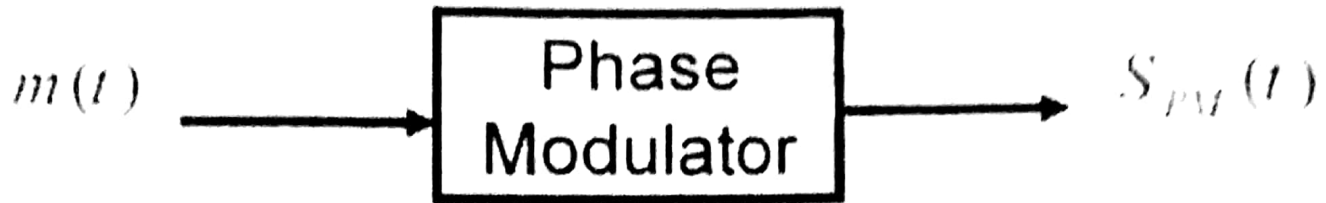
- k_f is the frequency sensitivity of the FM modulator expressed in rad/V s if $m(t)$ in Volts.

- The instantaneous frequency of $S_{FM}(t)$

$$f_i(t) = f_c + \frac{k_f}{2\pi} m(t)$$

Angle modulation viewed as FM or PM





Facts

- A PM/FM modulator may be used to generate an FM/PM waveform
- FM is much more frequently used than PM
- All the properties of a PM signal may be deduced from that of an FM signal
- In the remaining part of the chapter we deal mainly with FM signals.

Example 5.1

- Sketch FM and PM waves for the modulating signal $m(t)$ shown in Fig. 5.4a. The constants k_f and k_p are $2\pi \times 10^5$ and 10π , respectively, and the carrier frequency f_c is 100 MHz..

FM

$$f_i(t) = f_c + \frac{k_f}{2\pi} m(t) = 10^8 + 10^5 m(t)$$

$$(f_i)_{\min} = 99.9 \text{ MHz}$$

$$(f_i)_{\max} = 100.1 \text{ MHz}$$

Example Cont.

PM

$$f_i(t) = f_c + \frac{k_p}{2\pi} m'(t) = 10^8 + 5m'(t)$$

$$(f_i)_{\min} = 99.9\text{MHz}$$

$$(f_i)_{\max} = 100.1\text{MHz}$$



EE325: Chapter 5 (Lec. #2)

ANGLE MODULATION & DEMODULATIONS

M. A. Smadi

Bandwidth of Angle Modulated Signals

1) FM signals (using Taylor series)

(Function) تقریب یعنی $j \sin(\omega_c t) * \int k_f a(t)$

$$\begin{aligned}
 S_{FM}(t) &= A \operatorname{Re}\{e^{j[2\pi f_c t + k_f a(t)]}\} \\
 &= A \left[\cos(2\pi f_c t) - k_f a(t) \sin(2\pi f_c t) \right] \\
 &\quad + A \left[-\frac{k_f^2}{2!} a^2(t) \cos(2\pi f_c t) + \frac{k_f^3}{3!} a^3(t) \sin(2\pi f_c t) + \dots \right]
 \end{aligned}$$

where $a(t) = \int_{-\infty}^t m(\alpha) d\alpha$

$B_{FM} = \infty$ (theoretically)

- Narrow-Band Frequency Modulation (NBFM):

$$|k_f a(t)| \ll 1$$

$$S_{NBFM}(t) \approx A \left[\cos(2\pi f_c t) - k_f a(t) \sin(2\pi f_c t) \right]$$

$$B_{NBFM} = 2B$$

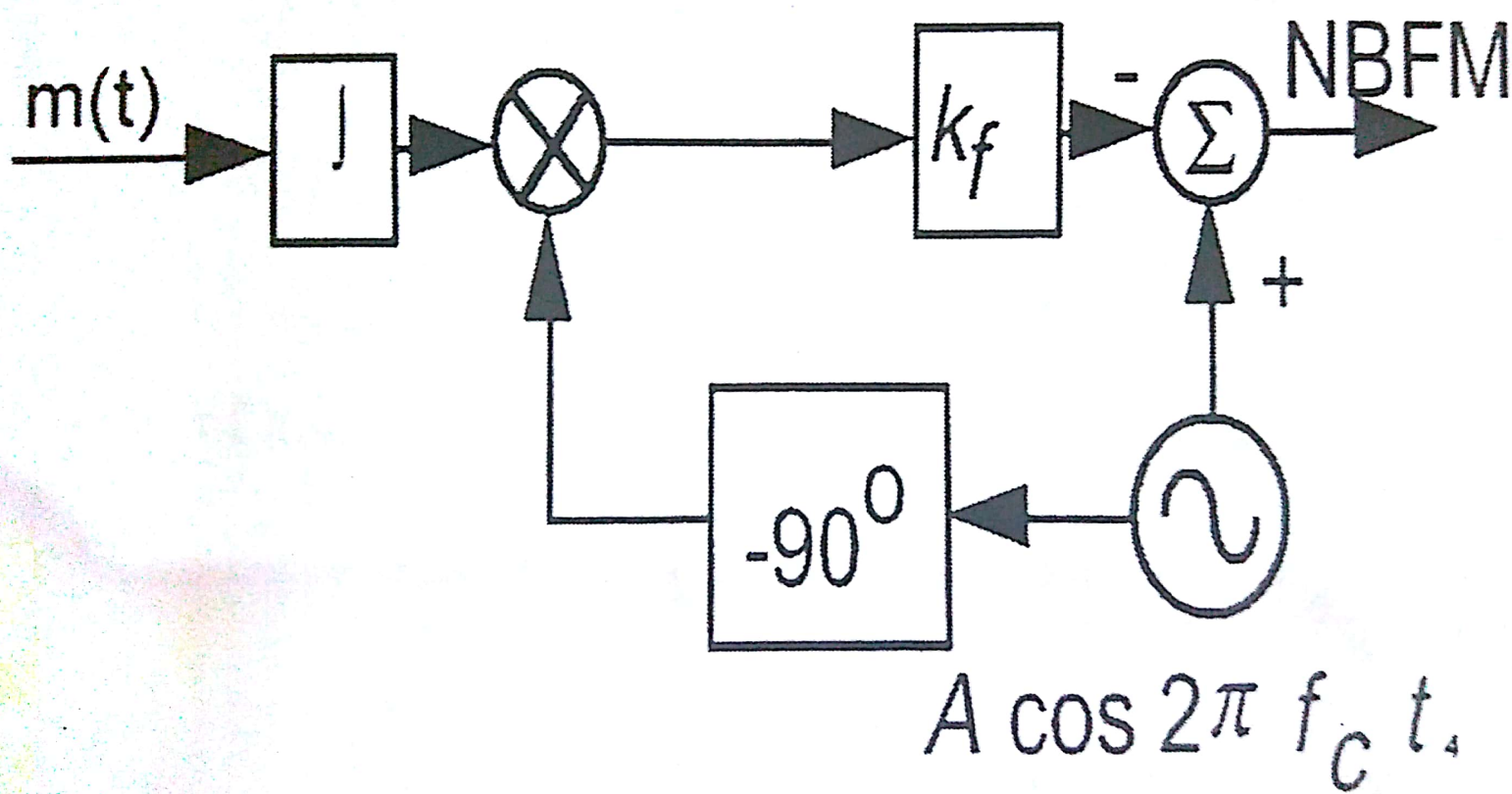


- Narrow-Band Phase Modulation (NBPM):

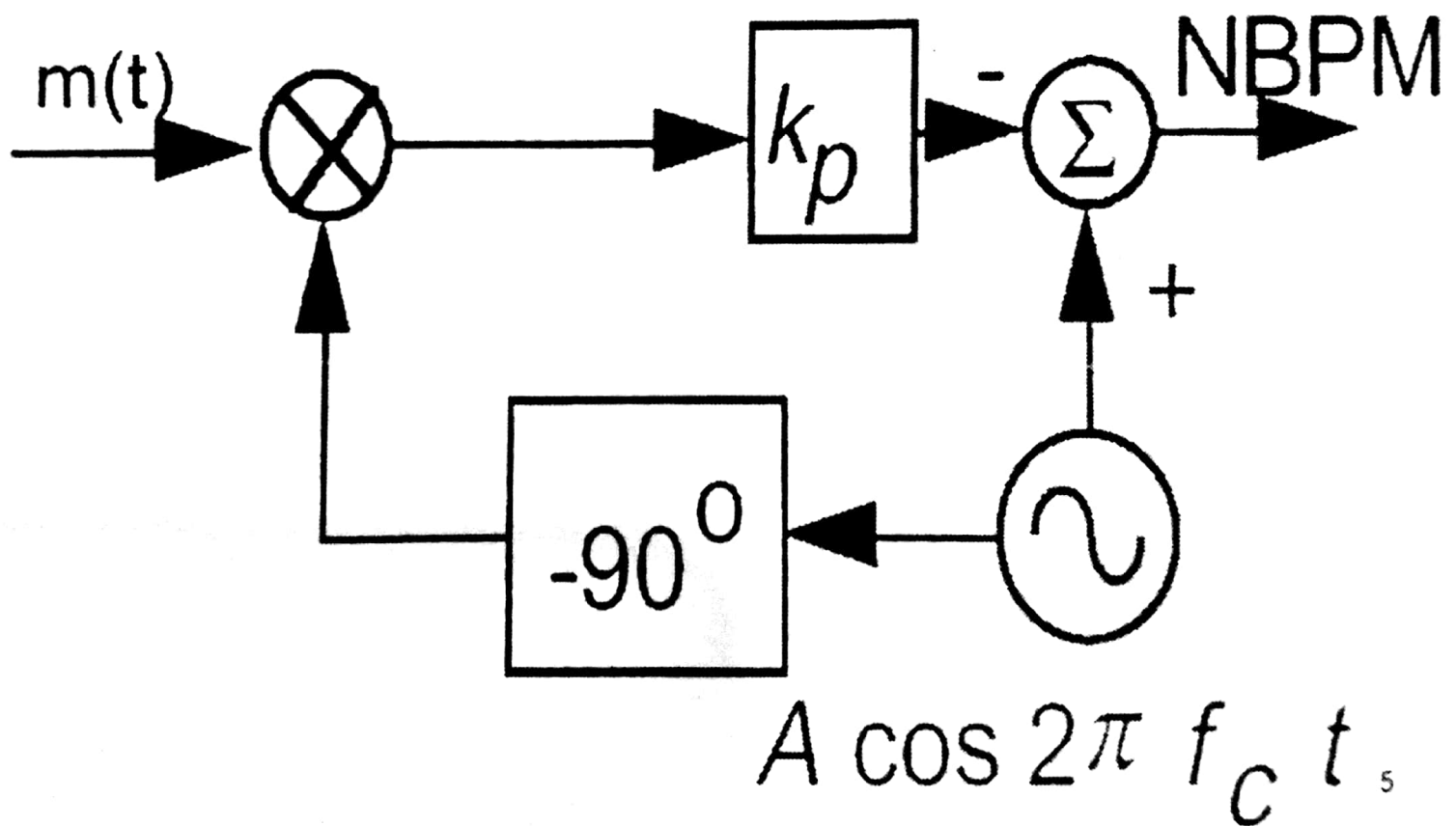
$$|k_p m(t)| \ll 1$$

$$S_{NBPM}(t) \approx A \left[\cos(2\pi f_c t) - k_p m(t) \sin(2\pi f_c t) \right]$$

$$B_{NBPM} = 2B$$



Generation of NBPM



- If $|k_f a(t)| \ll 1$ (See derivation)

$$B_{FM} = 2(\Delta f + B) = 2B(\beta + 1)$$

$$\Delta f = \frac{k_f m_p}{2\pi} \quad \beta = \frac{\Delta f}{B}$$

Δf : maximum carrier frequency deviation

β : deviation ratio or modulation index

$$m_p = \max |m(t)|$$

- Wide- Band Frequency Modulation (WBFM)

$$|k_f a(t)| \gg 1 \text{ or } \beta \gg 100$$

$$B_{WBFM} \approx 2\Delta f$$

Single tone modulation

- Let $m(t) = \cos 2\pi f_m t$; $B = f_m$

$$x_{FM}(t) = A \cos[2\pi f_c t + \beta \sin(2\pi f_m t)] = A \operatorname{Re}[e^{j(\omega_c t + \beta \sin \omega_m t)}]$$

- By FS $e^{j\beta \sin \omega_m t} = \sum_{n=-\infty}^{\infty} J_n(\beta) e^{jn\omega_m t}$

- Hence,

$$x_{FM}(t) = A \sum_{n=-\infty}^{\infty} J_n(\beta) \cos[2\pi(f_c + nf_m)t]$$

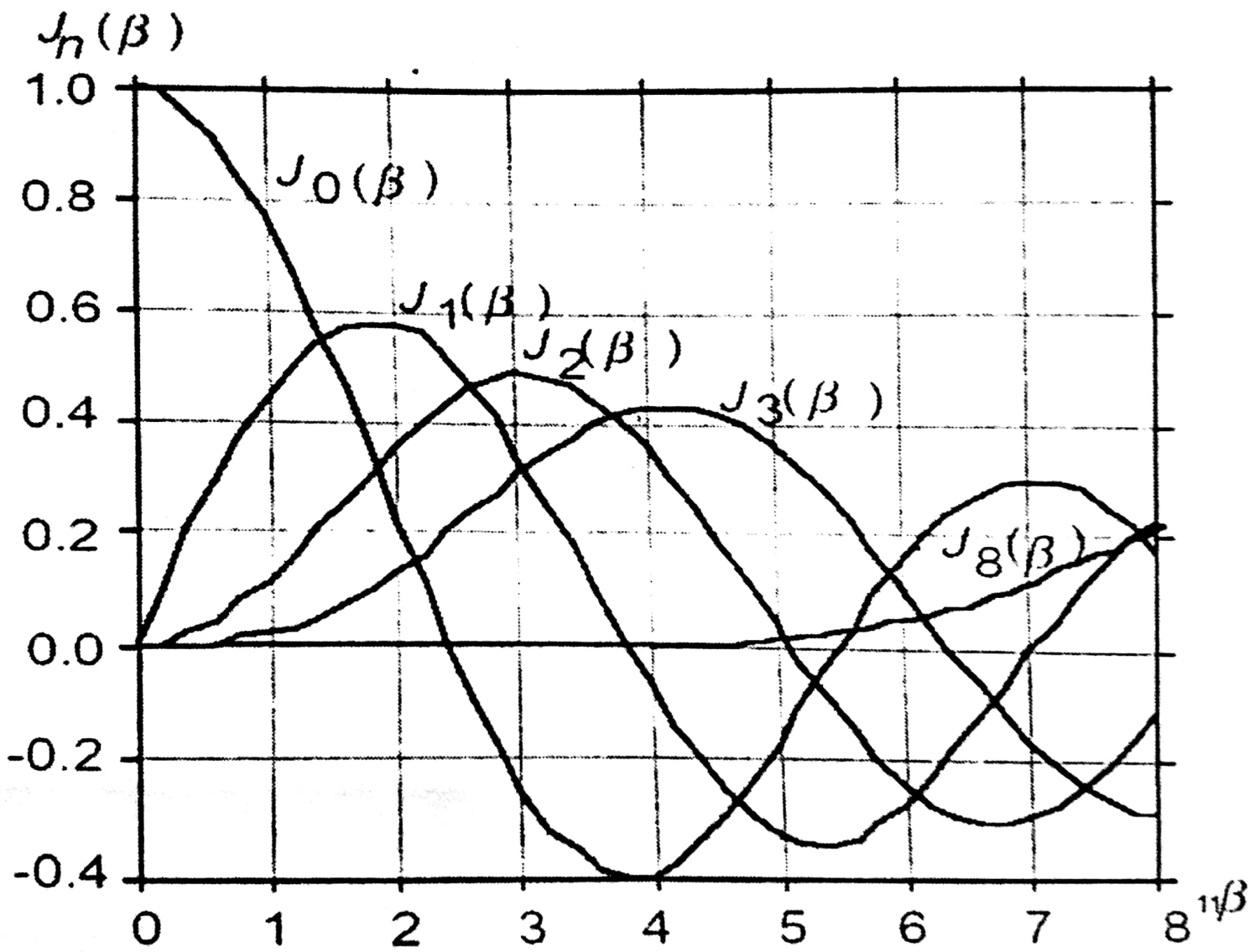
- $J_n(\beta) = \frac{1}{2\pi} \int_{-\pi}^{\pi} e^{j(\beta \sin x - nx)} dx$ is the nth order

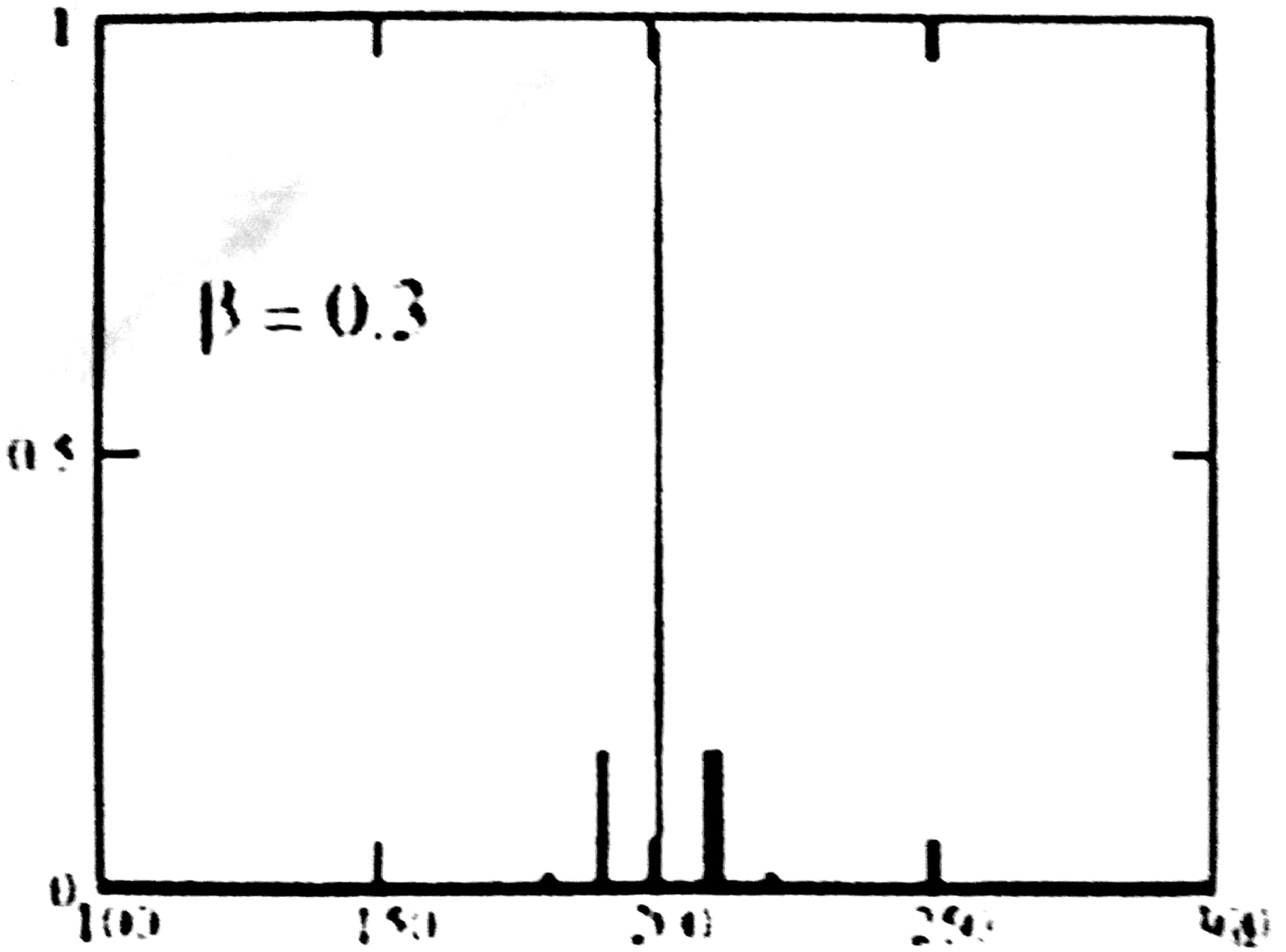
Bessel function with first kind

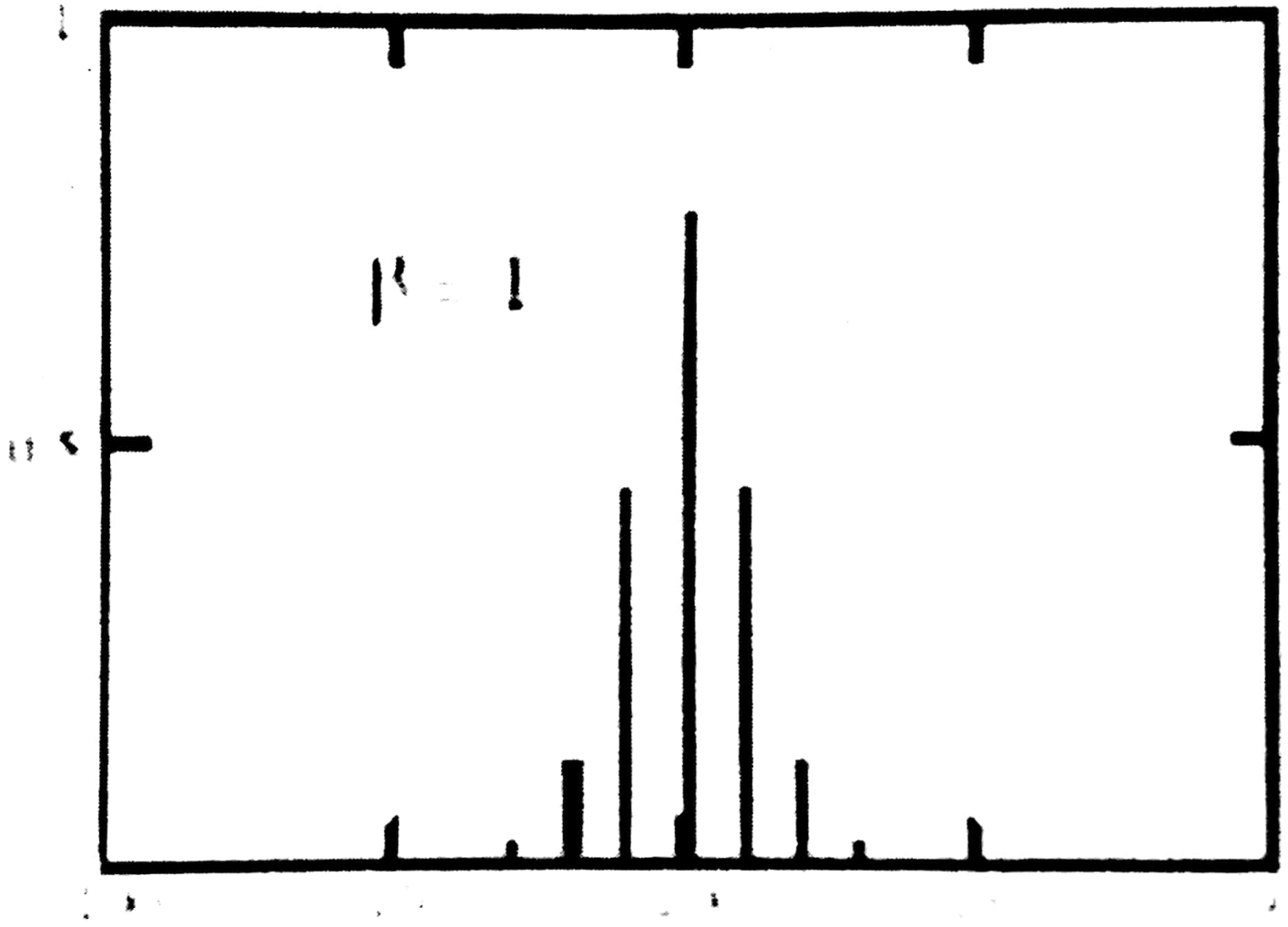
Spectrum of Angle-Modulated Signal

n	$\beta = 0.1$	$\beta = 0.2$	$\beta = 0.5$	$\beta = 1$	$\beta = 2$	$\beta = 5$	$\beta = 8$	$\beta = 10$	n	
0	<u>0.988</u>	<u>0.99</u>	0.988	0.765	0.224	-0.178	-0.72	-0.289	0	
1	0.50	0.19	<u>0.22</u>	0.440	0.577	-0.528	0.345	0.63	1	
2	0.17	0.005	0.31	<u>0.115</u>	0.353	0.27	-0.173	0.58	2	
3				0.20	<u>0.2</u>	0.68	-0.297	0.58	3	
4				0.02	0.31	0.91	-0.195	-0.22	4	
5					0.07	0.91	-0.89	-0.231	5	
6					0.01	<u>0.31</u>	0.338	-0.074	6	
7	the last significant spectral component:						0.53	-0.321	-0.217	7
8							0.18	-0.223	0.318	8
9	$n = \lceil \beta - 1 \rceil$						0.006	<u>0.126</u>	-0.292	9
10							0.01	0.061	0.297	10
11							0.29	<u>0.23</u>	11	
12							0.07	0.53	12	
13							0.03	0.29	13	
14							0.01	0.12	14	
15								0.004	15	
16								0.001	16	

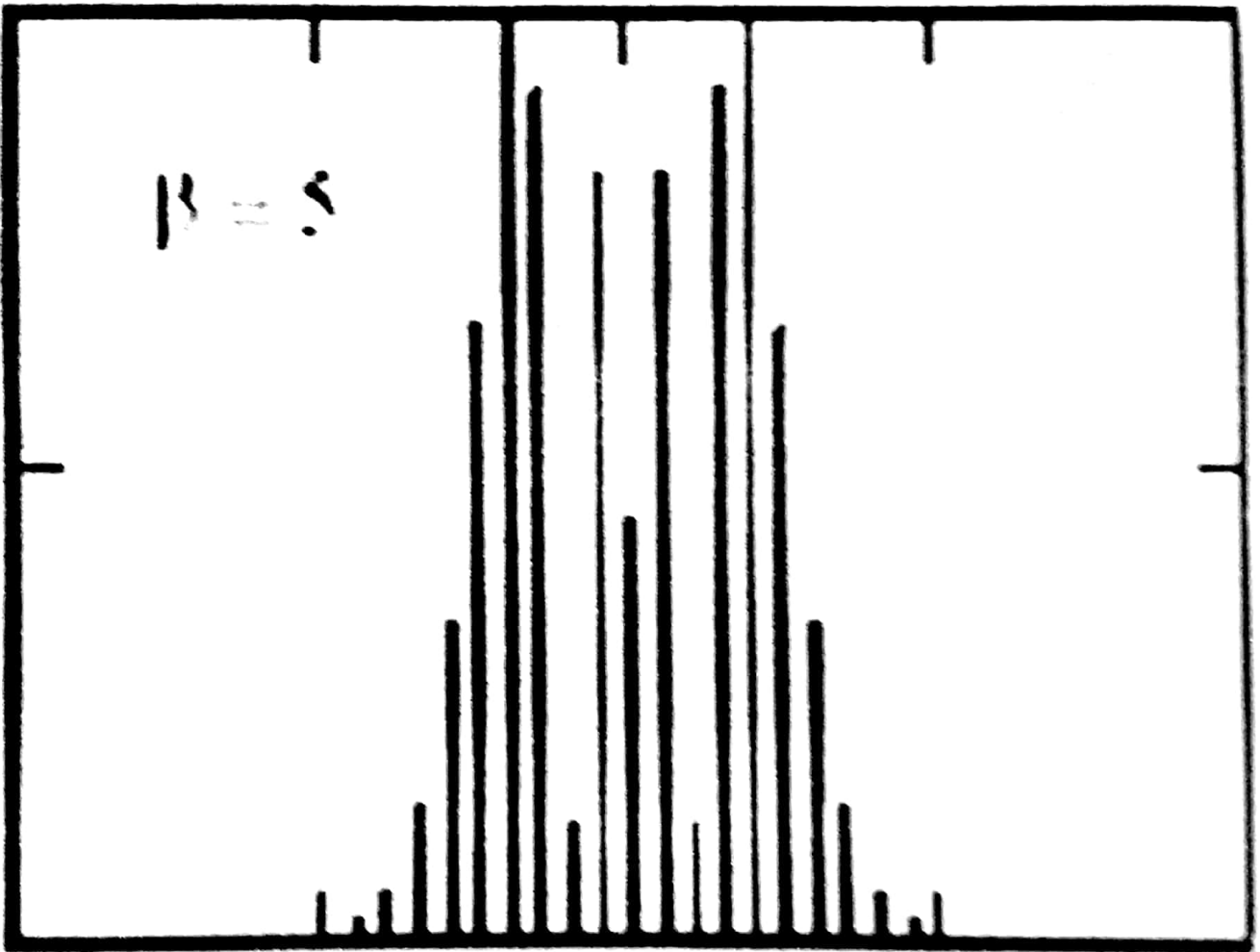
n	$\beta = 0.1$	$\beta = 0.2$	$\beta = 0.5$	$\beta = 1$	$\beta = 2$
0	<u>0.998</u>	<u>0.990</u>	0.938	0.765	0.224
1	0.050	0.100	<u>0.242</u>	0.440	0.577
2	0.001	0.005	0.031	<u>0.115</u>	0.353
3				0.020	<u>0.129</u>
4				0.002	0.034
5					0.007
6					0.001







B = S

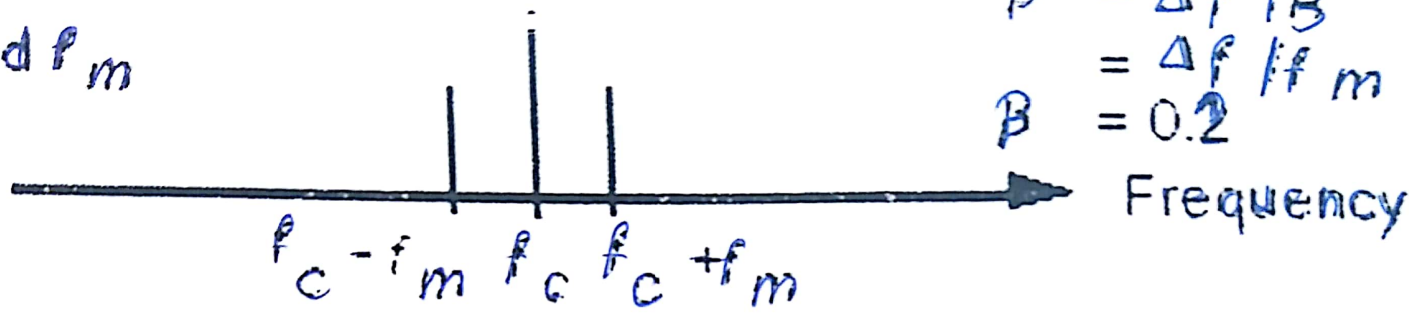


fixed f_m

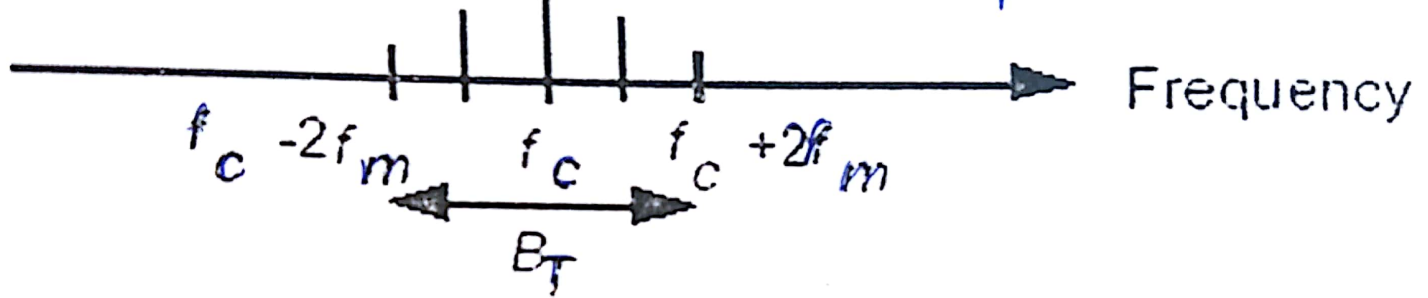
$$\beta = \Delta f / B$$

$$= \Delta f / f_m$$

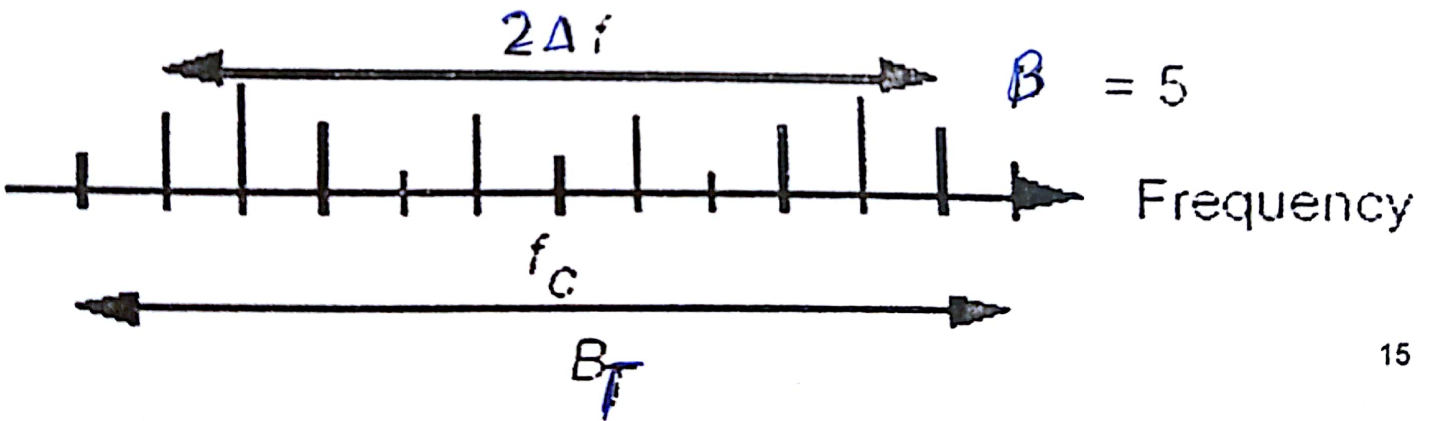
$$\beta = 0.2$$



$$\beta = 1$$



$$\beta = 5$$



- Note, $J_n(\beta)$ is negligible for $n > \beta + 1$

$$B_{FM} = 2(\beta + 1)f_m$$

$$\Delta f = \frac{\alpha k_f}{2\pi}$$

$$\beta = \frac{\Delta f}{f_m}$$

- The results is valid only for sinusoidal signal
- The single tone method can be used for finding the spectrum of an FM wave when m(t) is any periodic signal.



EE325: Chapter 5 (Lec. #3)

ANGLE MODULATION & DEMODULATIONS

M. A. Smadi

Features of Angle Modulation

- Channel bandwidth may be exchanged for improved noise performance. Such trade-off is *not* possible with AM
- Angle modulation is less vulnerable than AM to small signal interference from adjacent channels and more resistant to noise.
- Immunity of angle modulation to nonlinearities thus used for high power systems as microwave radio.

Am
noise

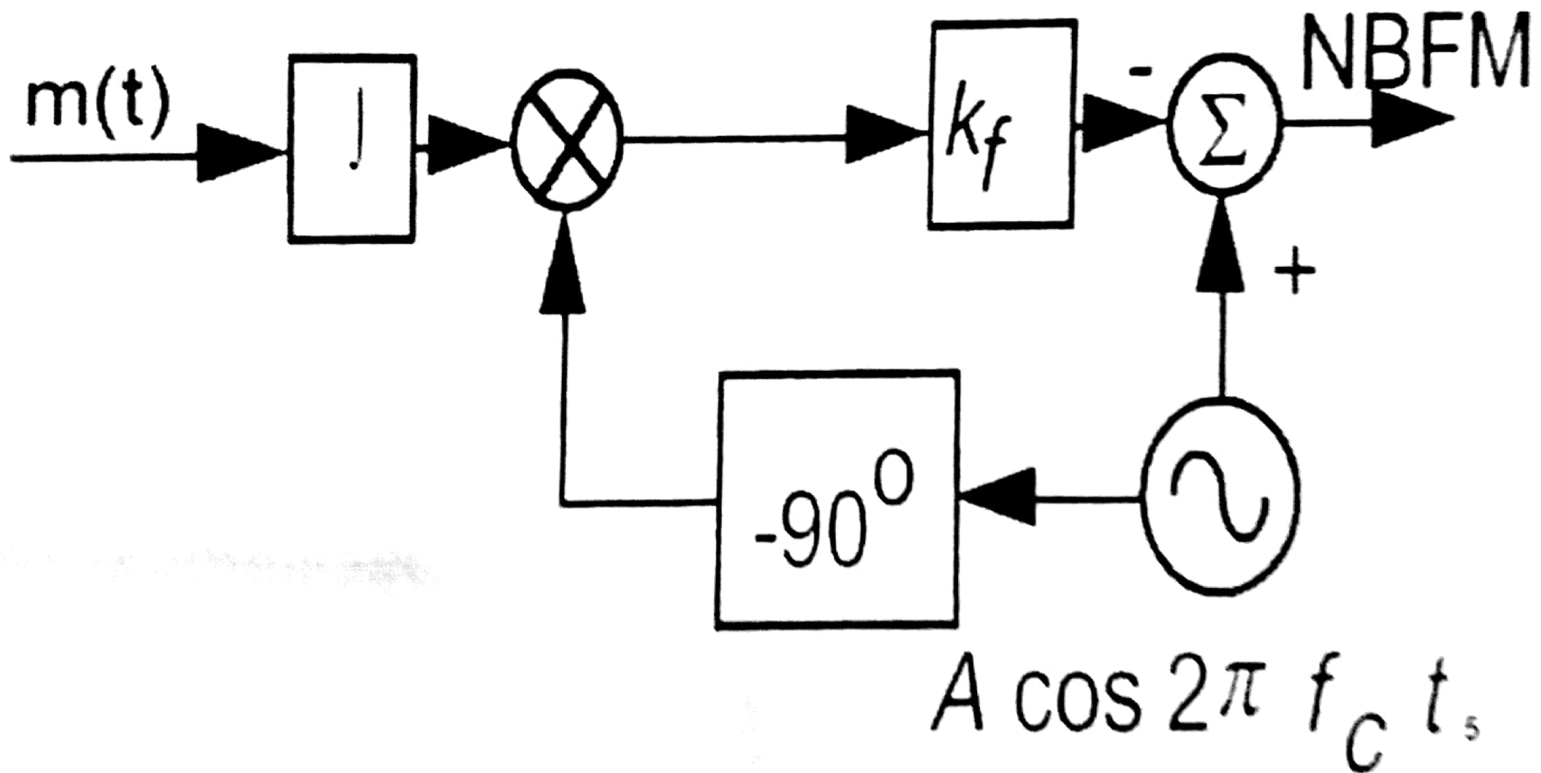
- FM is used for: radio broadcasting, sound signal in TV, two-way fixed and mobile radio systems, cellular telephone systems, and satellite communications.
- PM is used extensively in data communications and for indirect FM.
- WBFM is used widely in space and satellite communication systems.
- WBFM is also used for high fidelity radio transmission over rather limited areas.

Generation of FM Signals

- There are two ways of generating FM waves:
 - Indirect generation
 - Direct generation

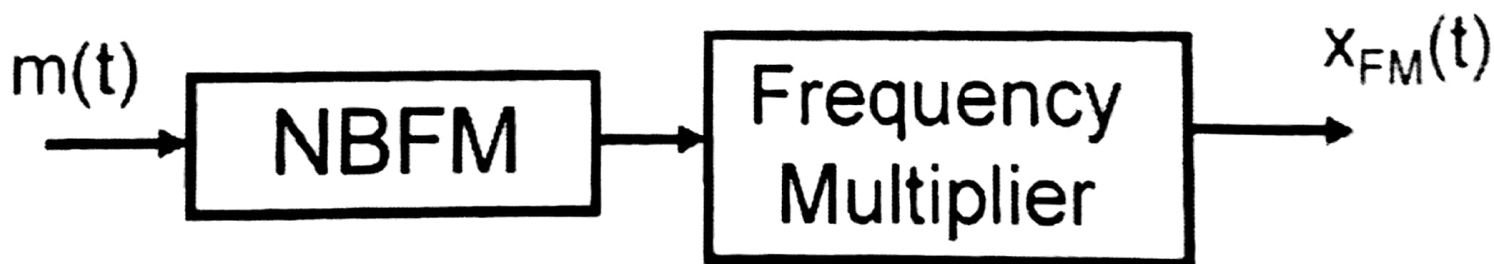
Indirect Generation of NBFM

$$S_{\text{NBFM}}(t) \approx A [\cos(2\pi f_c t) - k_f m(t) \sin(2\pi f_c t)]; |k_f m(t)| \ll 1$$

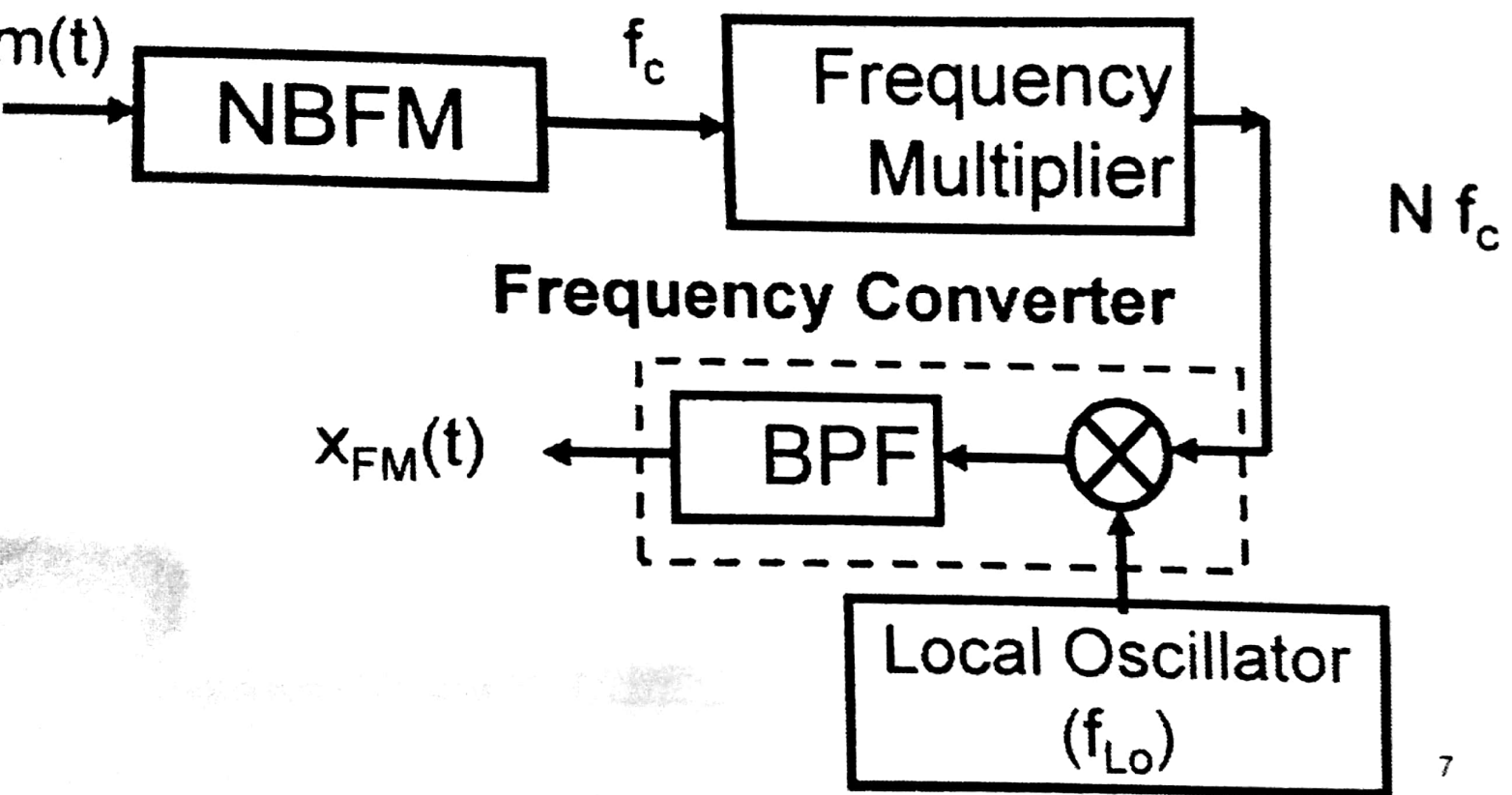


Indirect Generation of Wideband FM

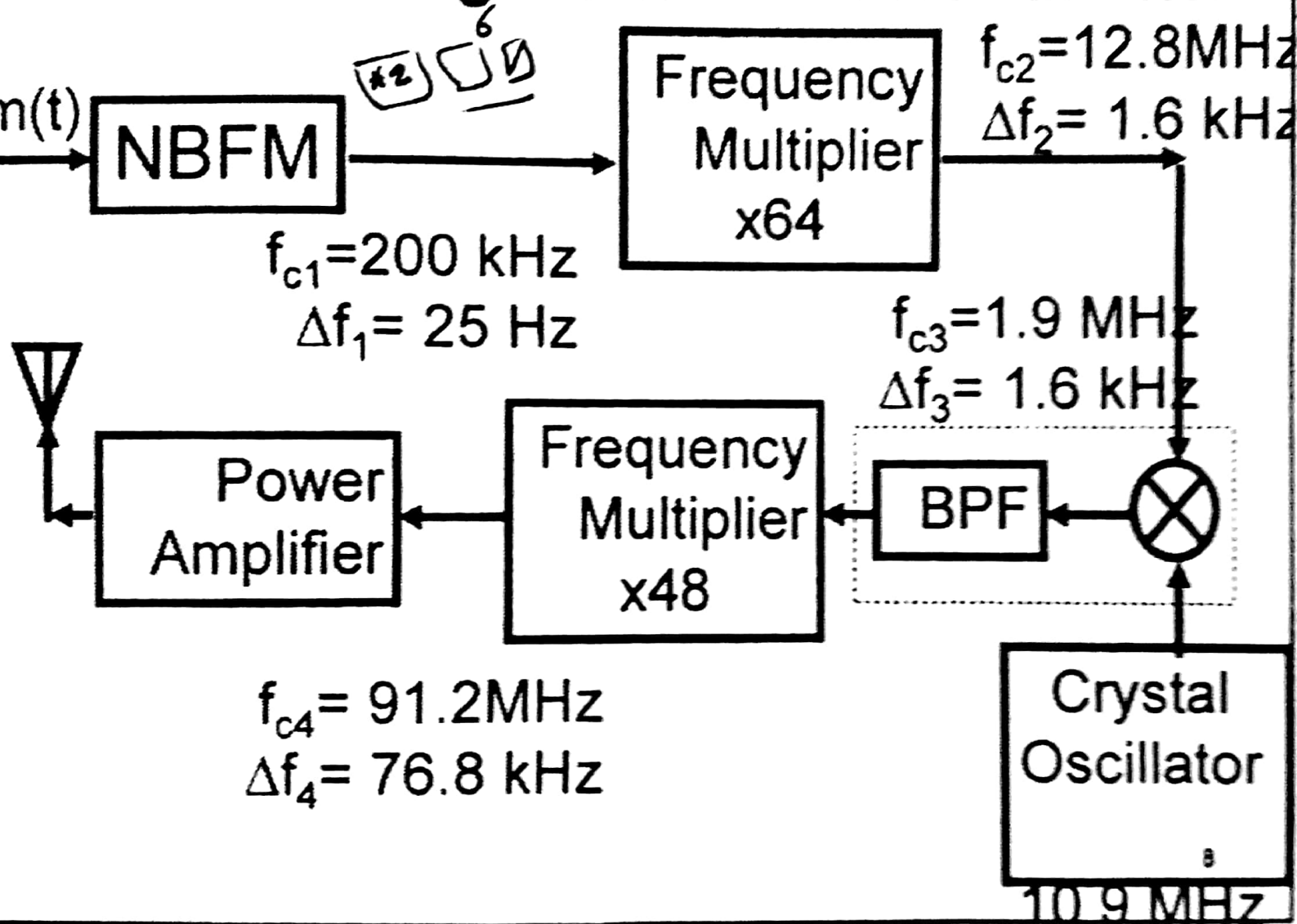
- In this method, a narrowband frequency-modulated signal is first generated and then a frequency multiplier (nonlinear device) is used to increase the modulation index.



Indirect Wideband FM

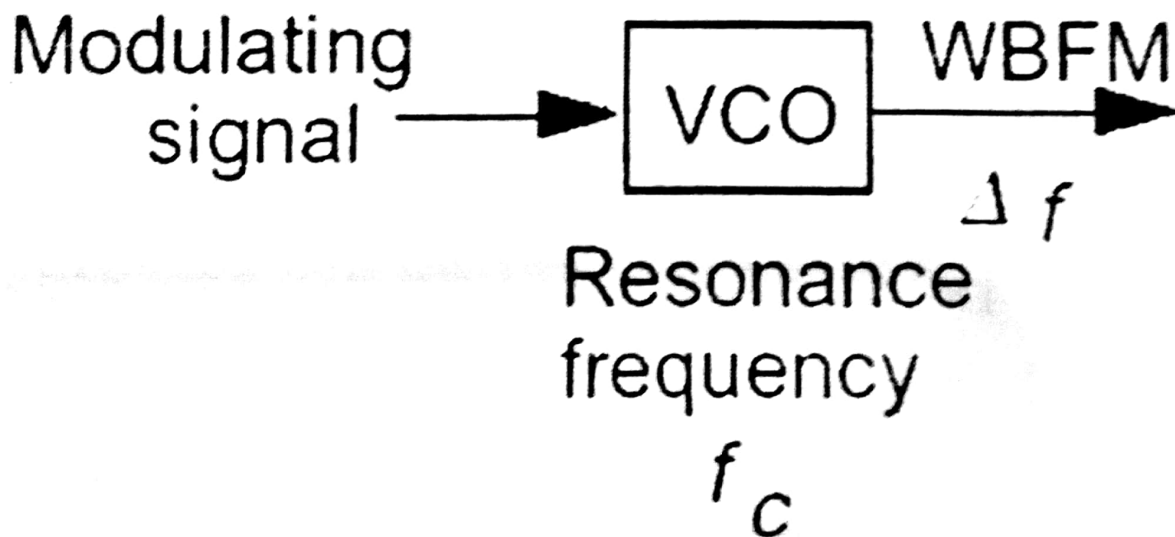


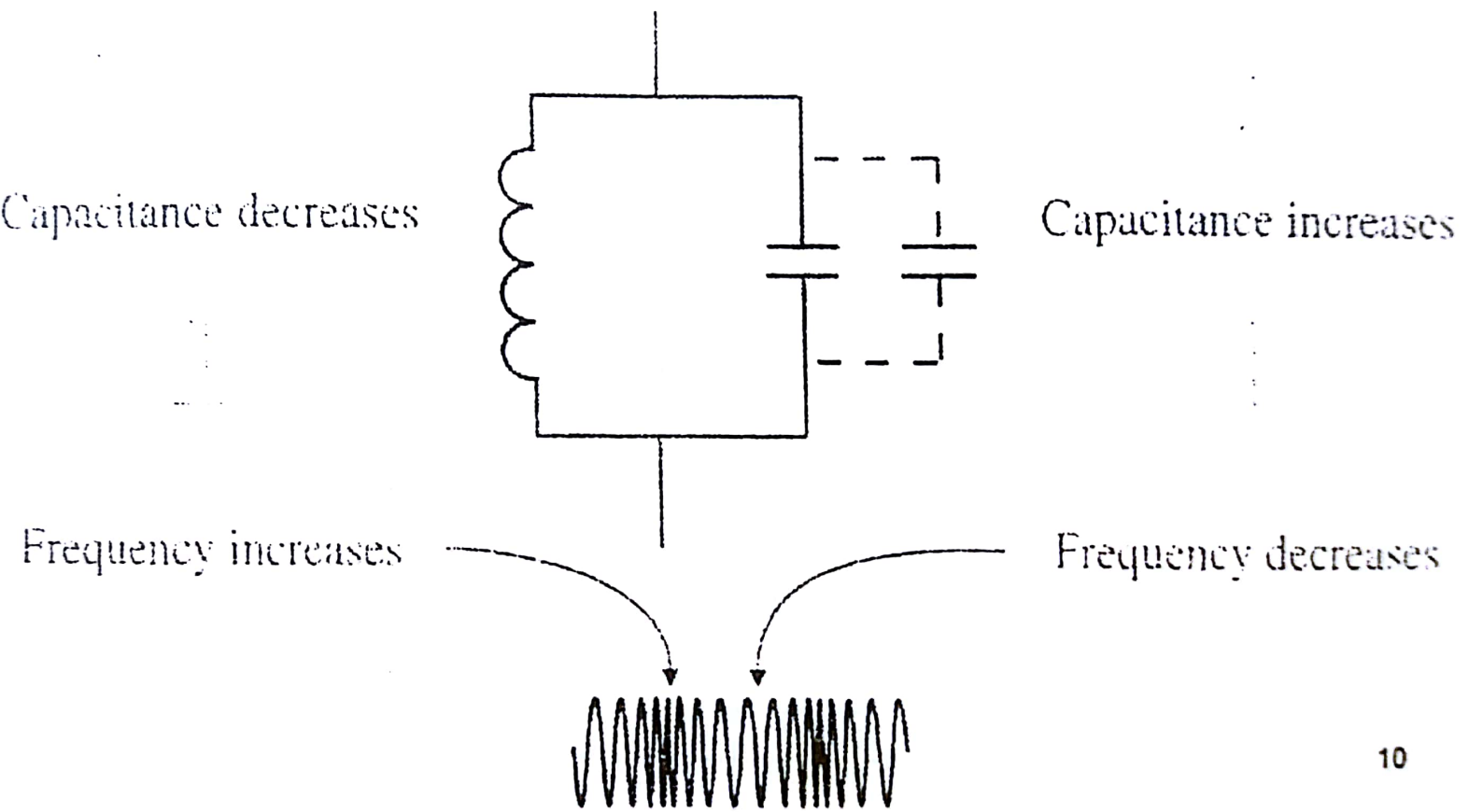
Armstrong Indirect FM Transmitter



Direct Generation

- The modulating signal $m(t)$ directly controls the carrier frequency. [$f_c(t) = f_c + k_f m(t)$]
- A common method is to vary the inductance or capacitance of a voltage controlled oscillator.





- In Hartley or Colpitt oscillator, the frequency is given by

$$\omega_0 = \frac{1}{\sqrt{LC}}$$

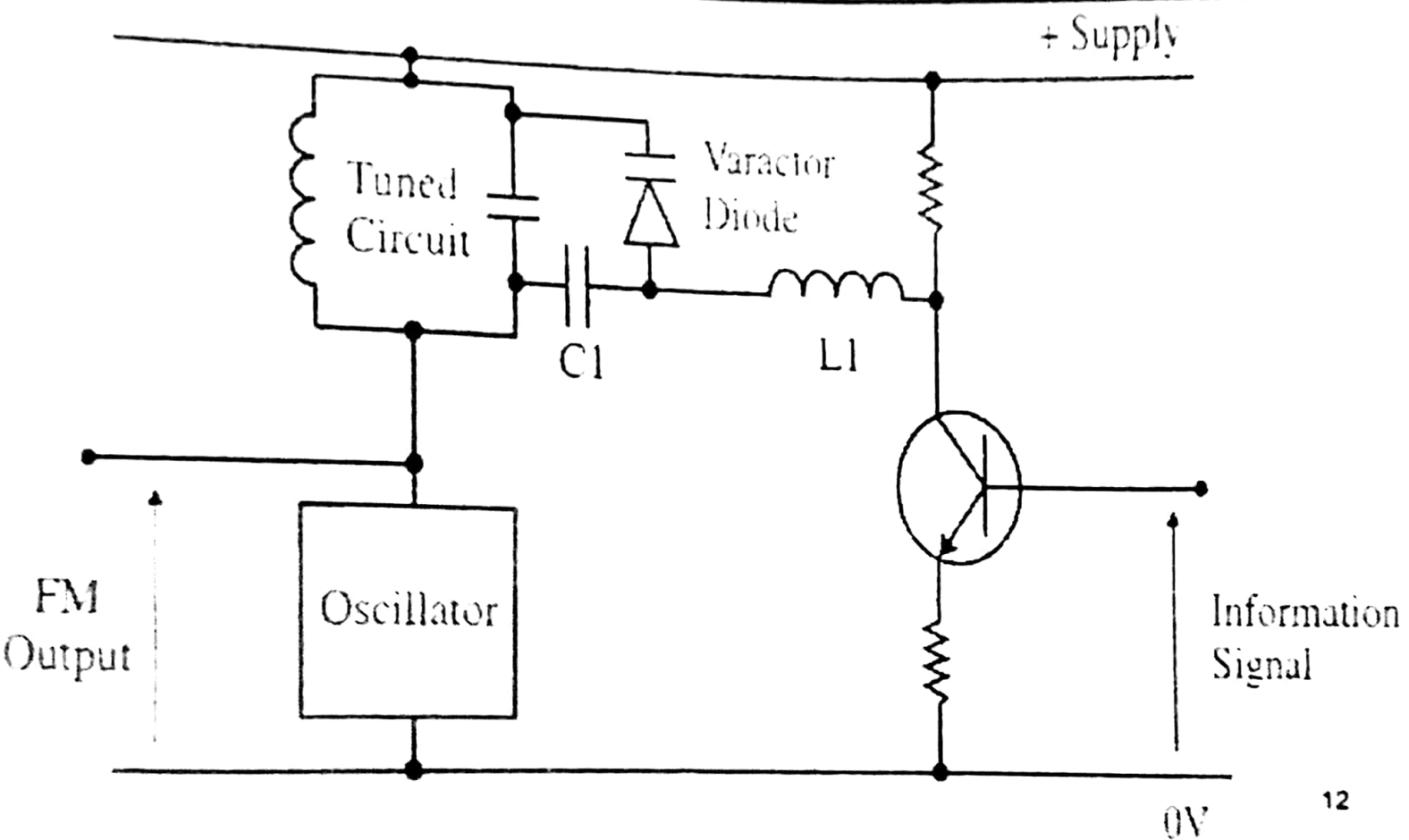
- If $C = C_0 - km(t)$ and $km(t) \ll C_0$

$$\omega_0 = \frac{1}{\sqrt{LC_0 \left| 1 - \frac{km(t)}{C_0} \right|}} \approx \frac{1}{\sqrt{LC_0}} \left| 1 + \frac{km(t)}{2C_0} \right|,$$

$$(1+x)^n \approx 1+nx, \quad x \ll 1$$

- Hence $\omega_0 = \omega_c + k_f m(t), \quad k_f = \frac{k \omega_c}{2C_0}$ 11

Varactor Modulator Circuit



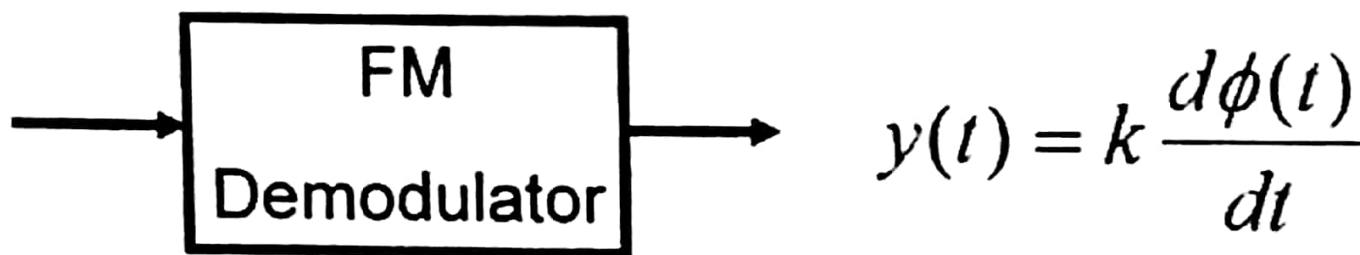
- Advantage - Large frequency deviations are possible and thus less frequency multiplication is needed.
- Disadvantage - The carrier frequency ^{is} tends to drift and additional circuitry is required for frequency stabilization.
- To stabilize the carrier frequency, a phase-locked loop can be used.

Examples 5.6 & 5.7

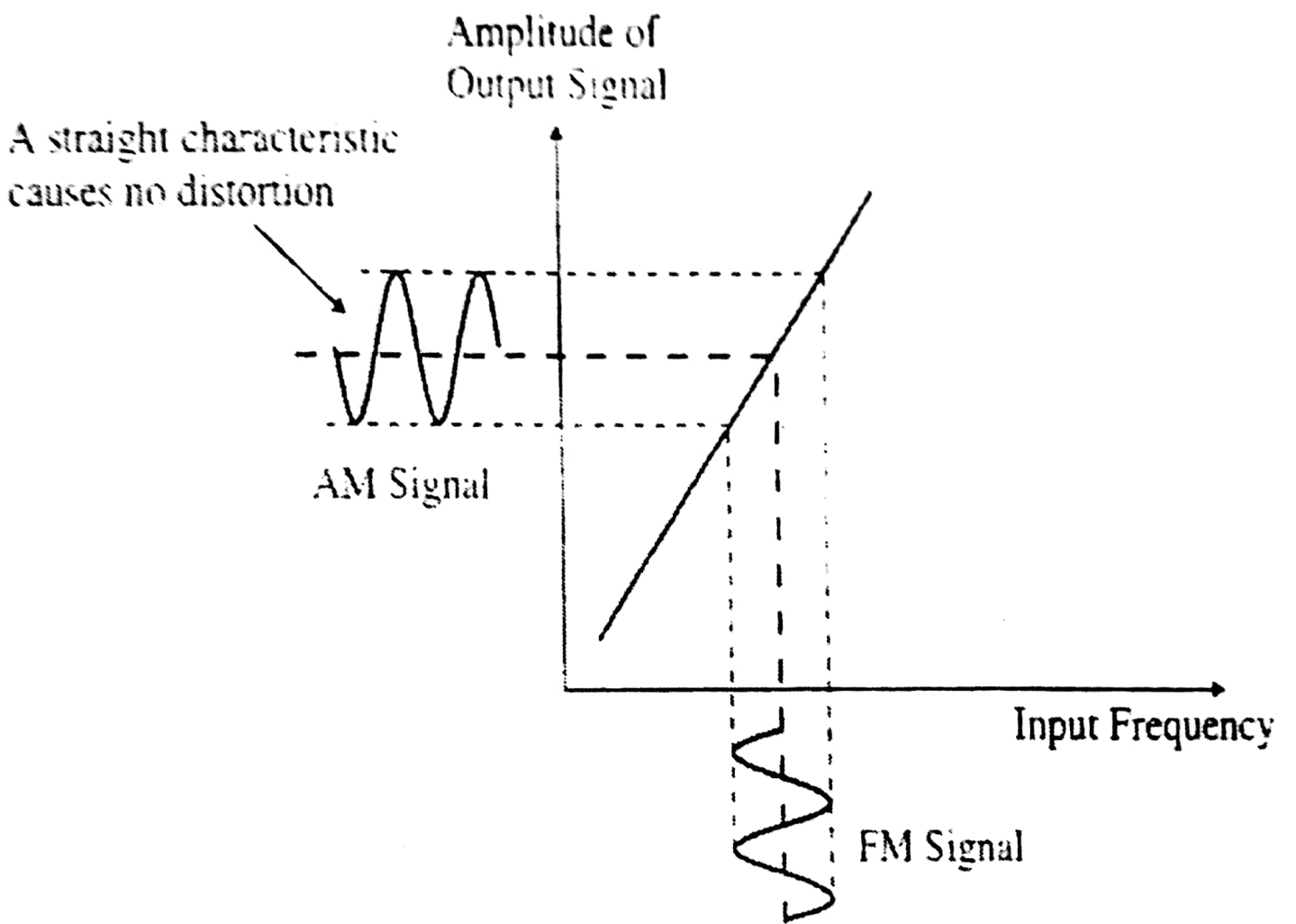
- Later!!!

Demodulation of FM Signals

$$x(t) = A \cos [\omega_c t + \phi(t)]$$

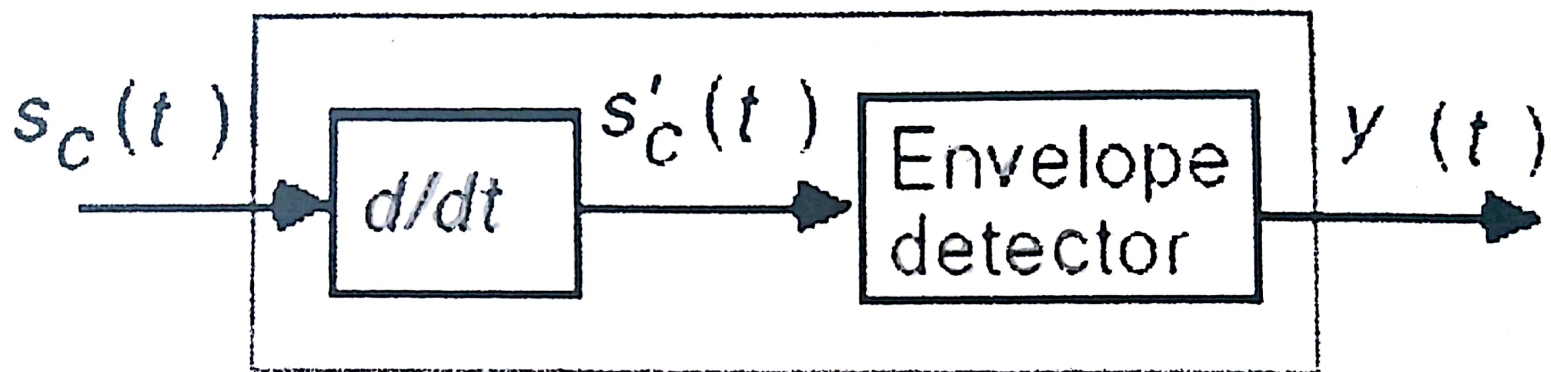


- Demodulation of an FM signal requires a system that produces an output proportional to the instantaneous frequency deviation of the input signal.
- Such system is called a frequency discriminator.



- A frequency-selective network with a transfer function of the form $|H(\omega)| = a \omega$ over the FM band would yield an output proportional to the instantaneous frequency.
- There are several possible examples for frequency discriminator, the simplest is the FM demodulator by direct differentiation

FM demodulator via direct differentiation

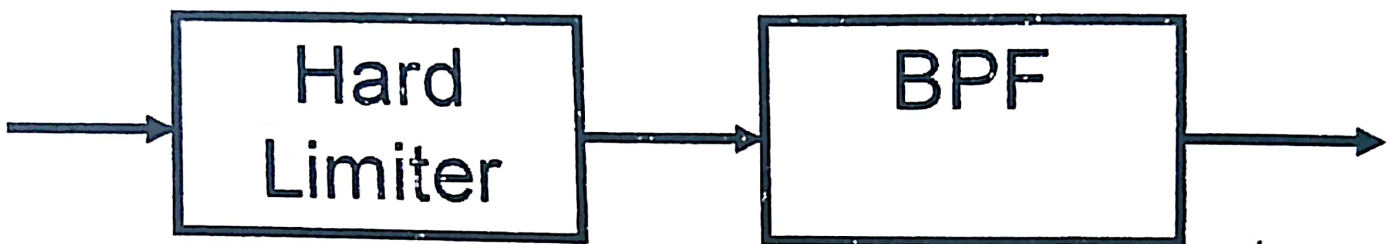


$$s_c(t) = A \cos \left[2\pi f_c t + k_f \int_{-\infty}^t m(\alpha) d\alpha \right]$$

$$s'_c(t) = -A \left[\omega_c + k_f m(t) \right] \sin \left[\omega_c t + k_f \int_{-\infty}^t m(\alpha) d\alpha \right]$$

- The basic idea is to convert FM into AM and then use AM demodulator.

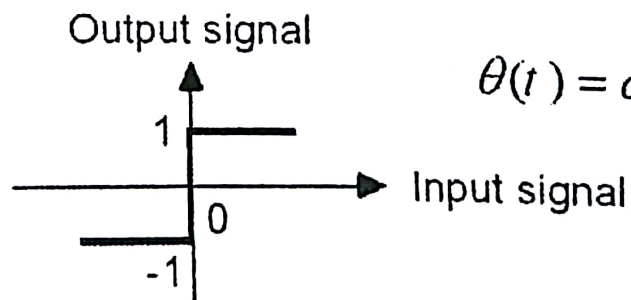
Bandpass Limiter



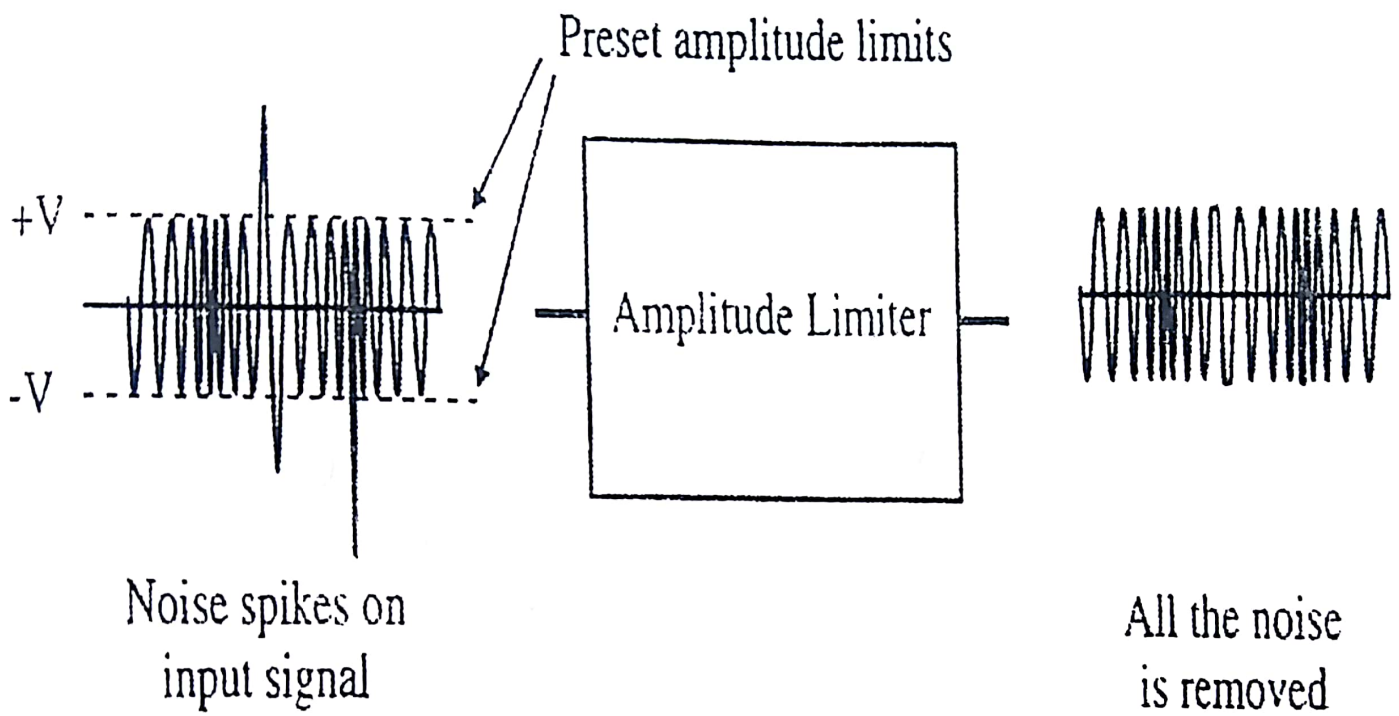
$$v_i(t) = A(t) \cos[\theta(t)] \quad v_o(\theta) = \begin{cases} 1, & \cos(\theta) > 0 \\ -1, & \cos(\theta) < 0 \end{cases} \quad v_o(t) = \frac{4}{\pi} \cos[\theta(t)]$$

By FS

$$v_o(\theta) = \frac{4}{\pi} [\cos(\theta) - 1/3 \cos(3\theta) + 1/5 \cos(5\theta) - \dots]$$



$$\theta(t) = \omega_c t + k_f \int m(\alpha) d\alpha$$



- Any signal which exceeds the preset limits are simply chopped off



EE325: Chapter 5 (Lec. #4)

ANGLE MODULATION & DEMODULATIONS

M. A. Smadi

Practical Frequency Demodulators

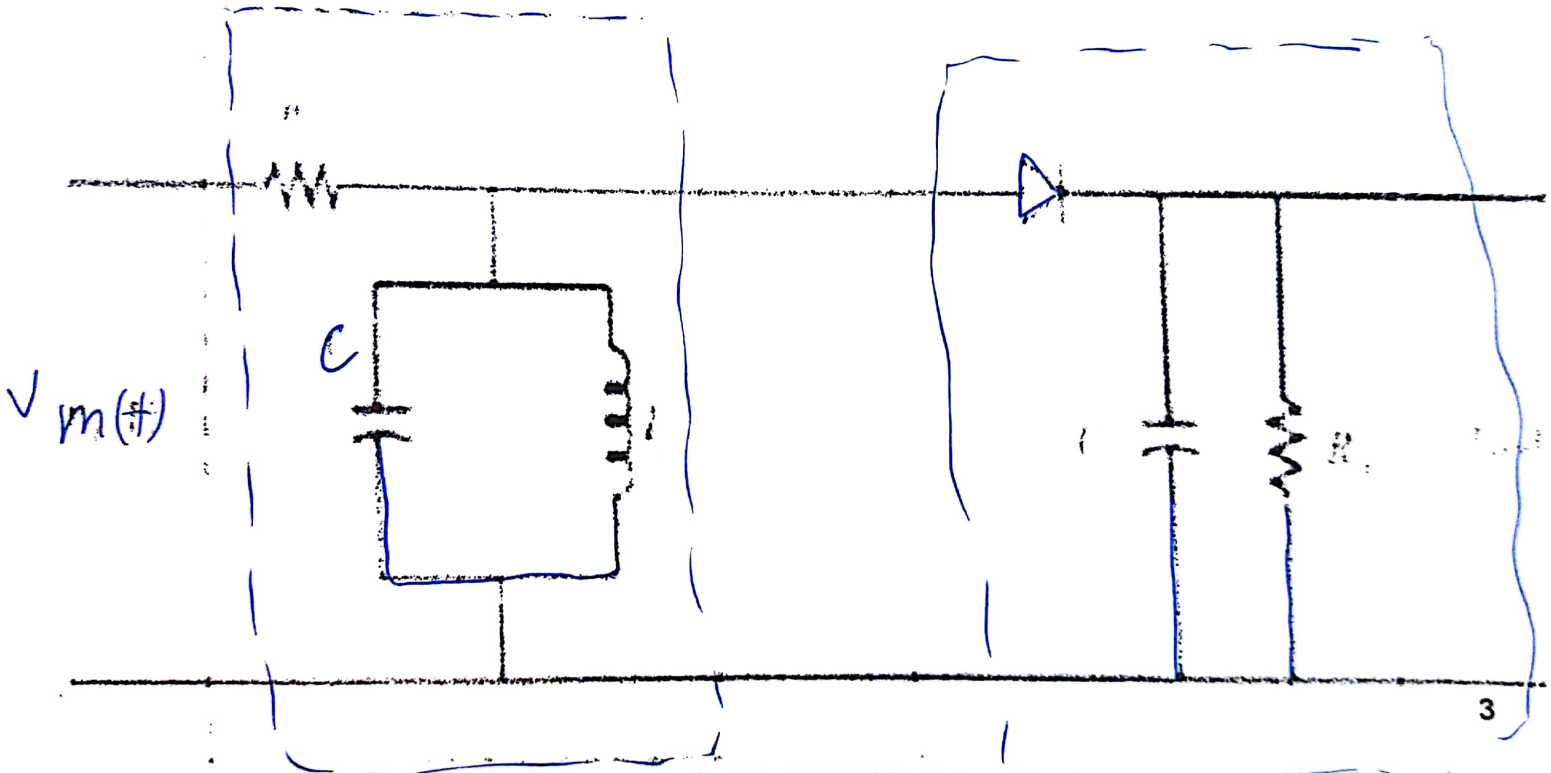
- There are several possible networks for frequency discriminator
 - **FM slope detector**
 - **Balanced discriminator**
 - **Quadrature Demodulator**
- Another superior technique for the demodulation of the FM signal is to use the **Phased locked loop (PLL)**

FM Slope Detector

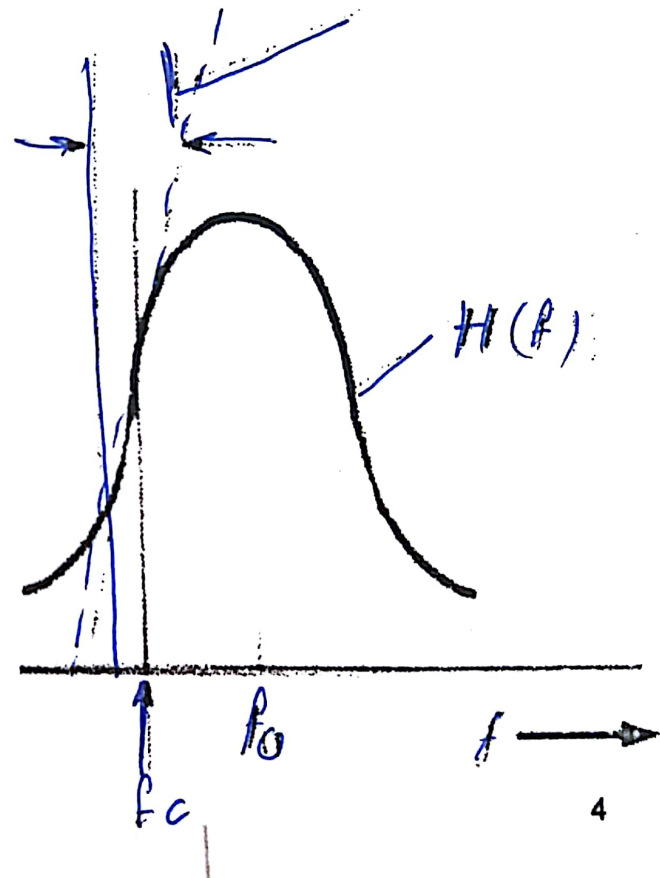
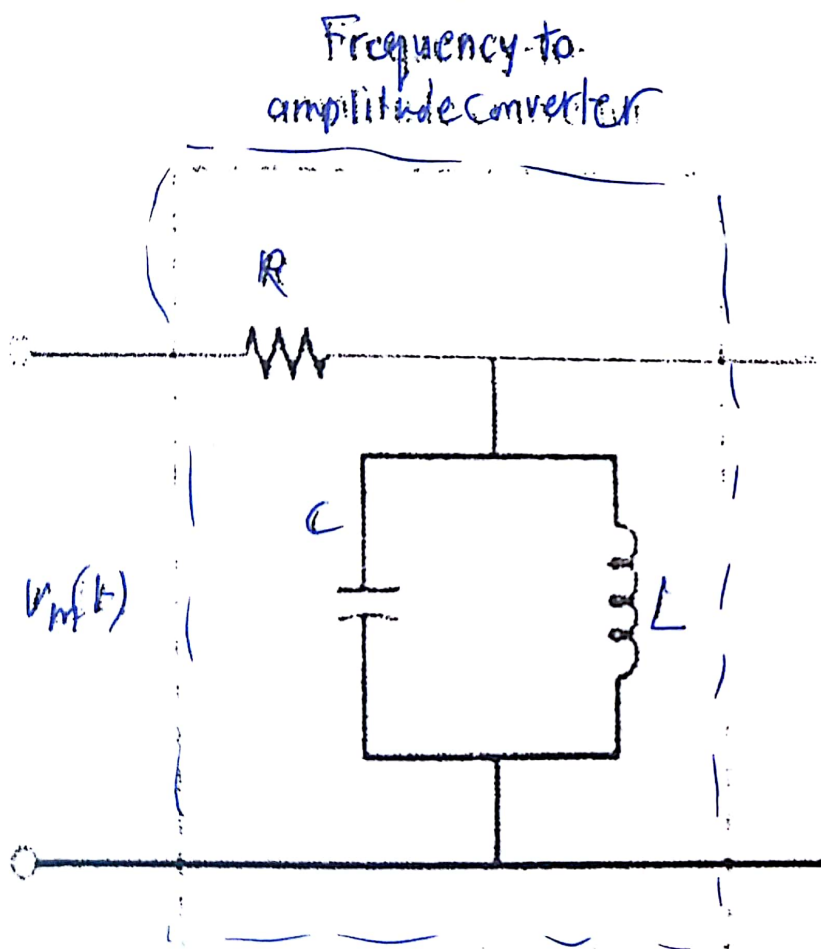
$$s_c(t) = A \cos \left[2\pi f_c t + k_f \int_{-\infty}^t m(\alpha) d\alpha \right]$$

Frequency to
Amplitude Converter

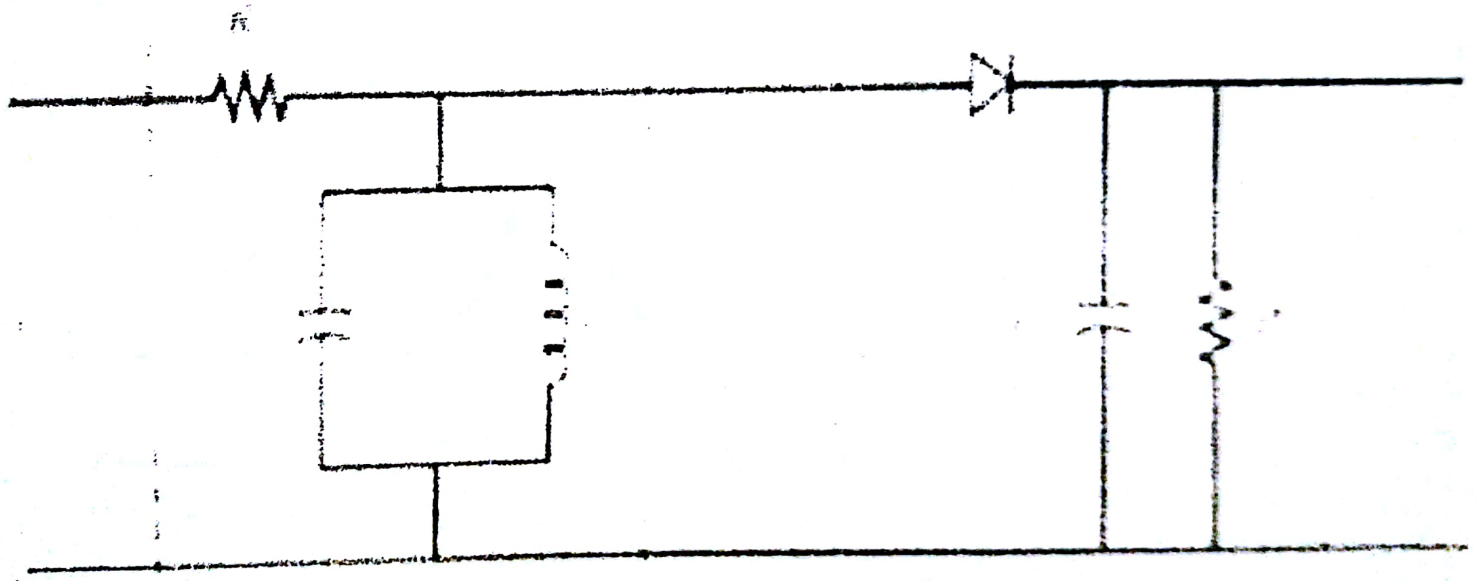
Envelope Detector



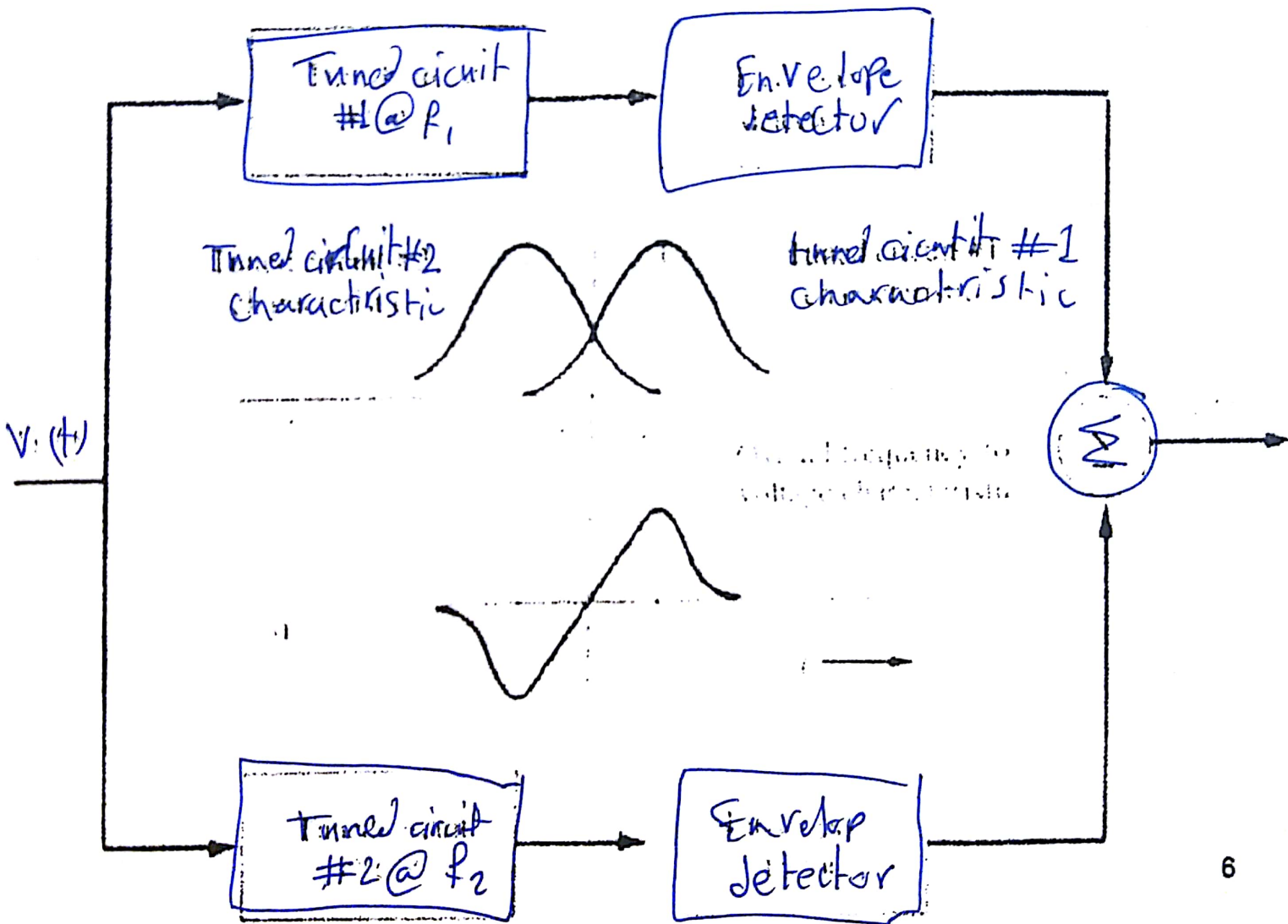
FM Slope Detector



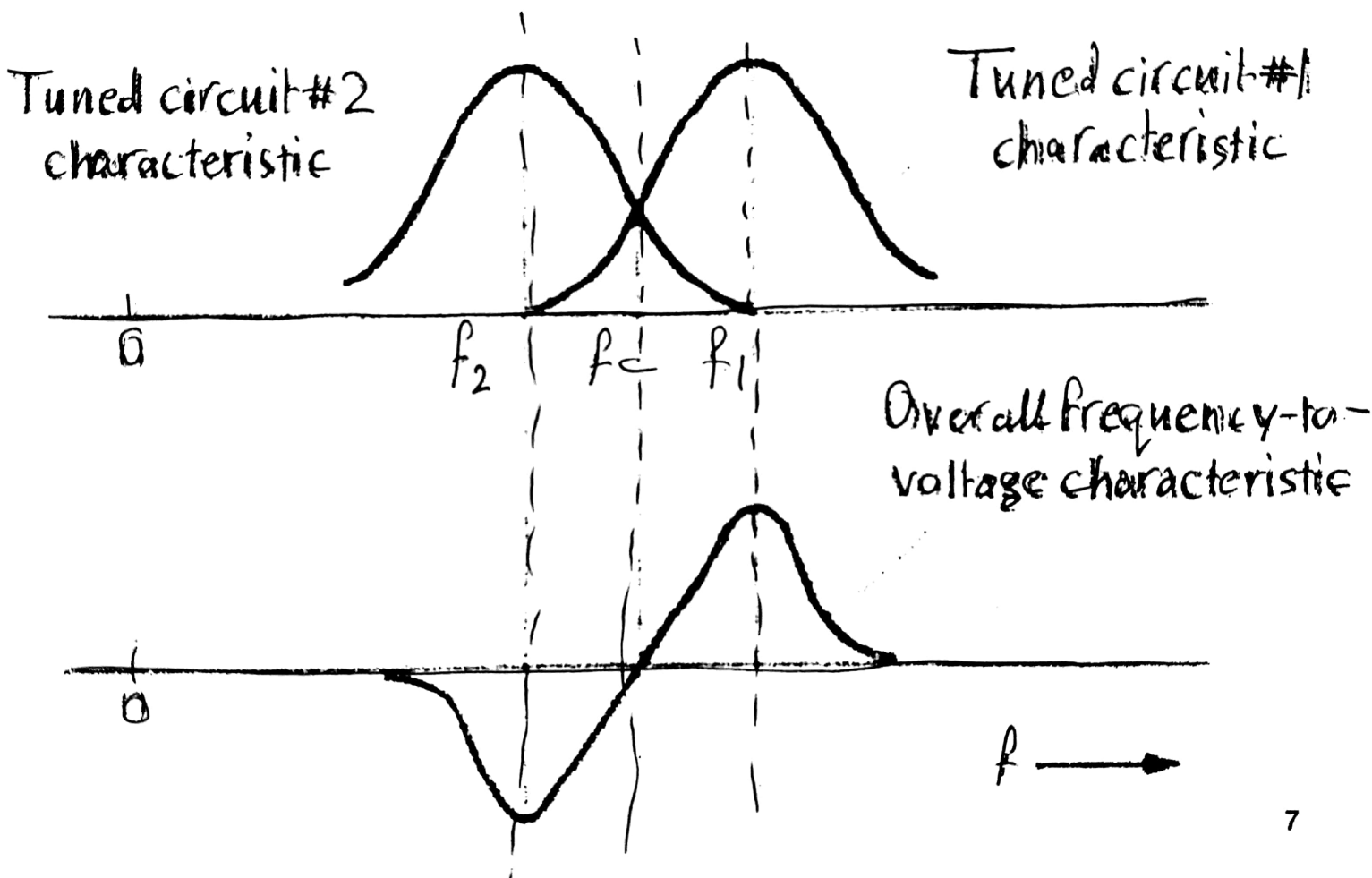
FM Slope Detector



Balanced Discriminator



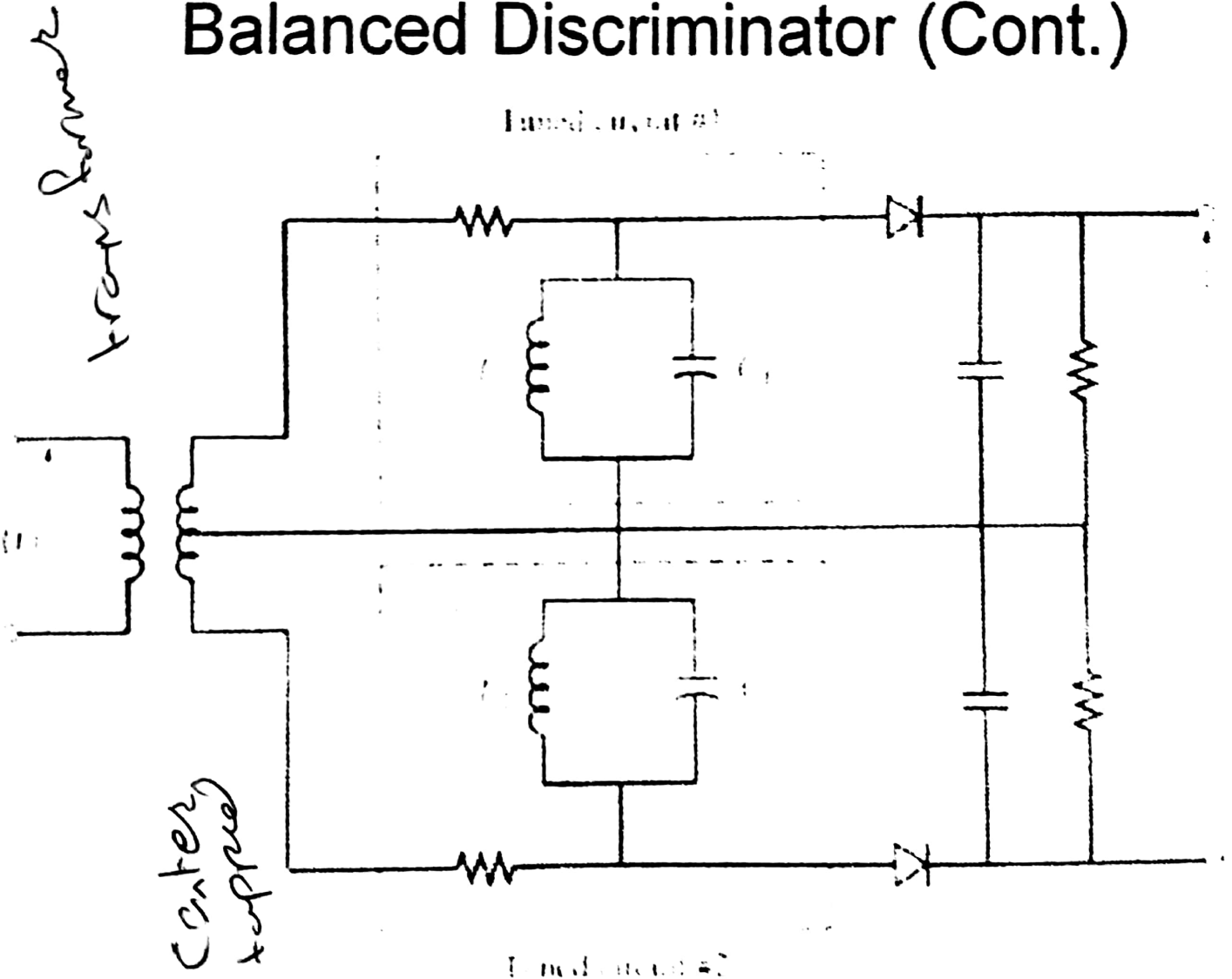
Balanced Discriminator (Cont.)



Balanced Discriminator (Cont.)

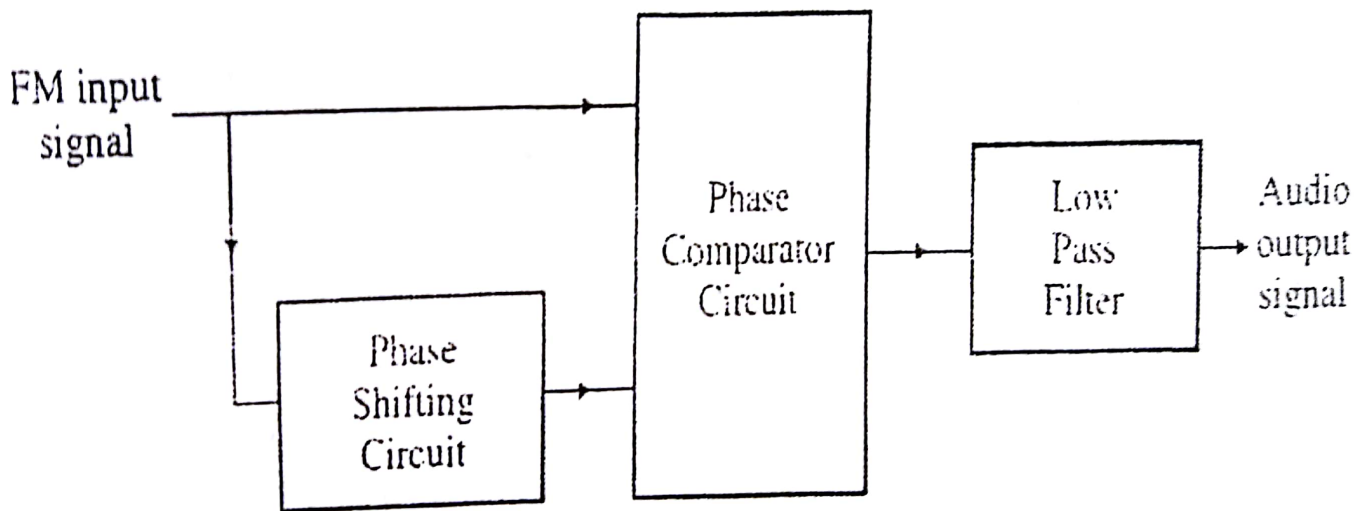
Tuned circuit #1

Tuned circuit #2

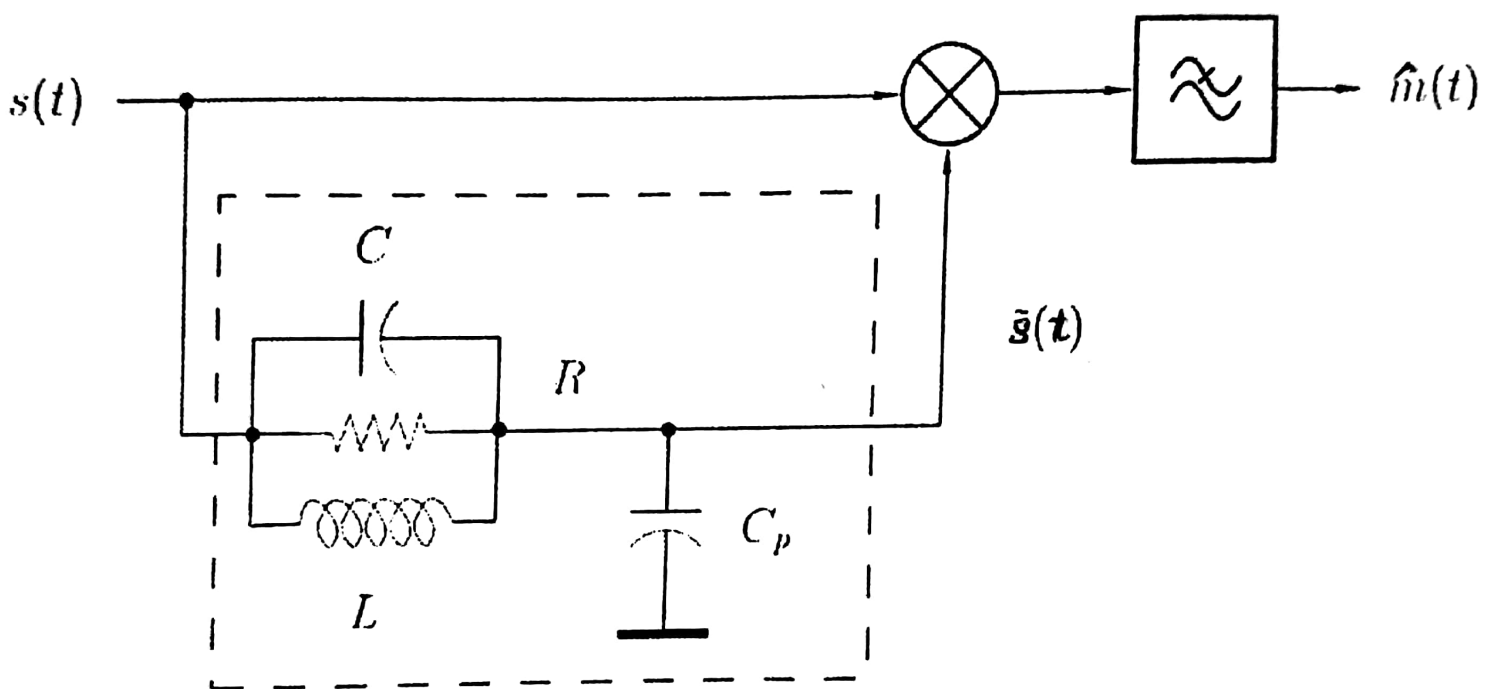


Quadrature Demodulator

- FM is converted into PM
- PM detector is used to recover message signal



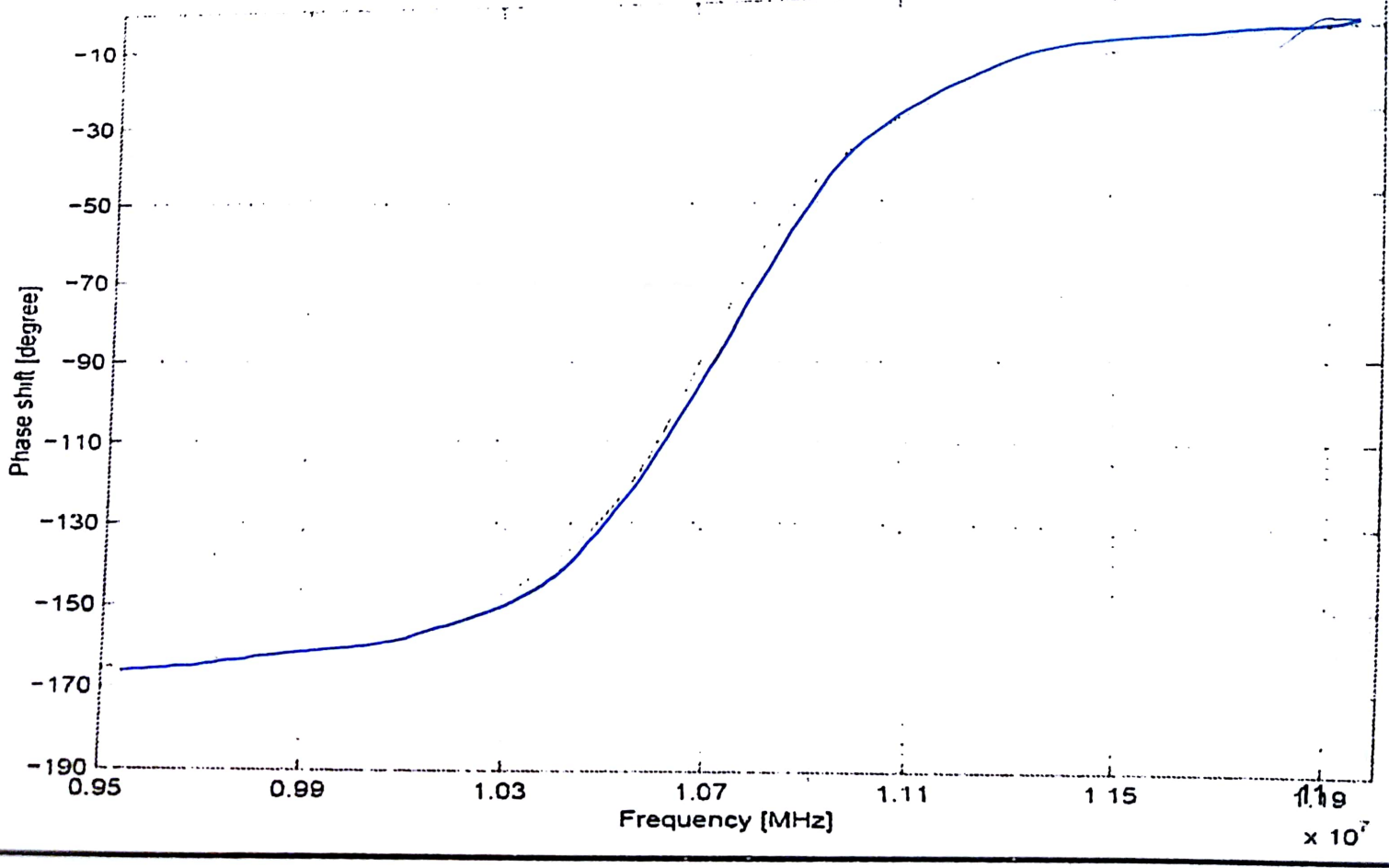
Quadrature Demodulator



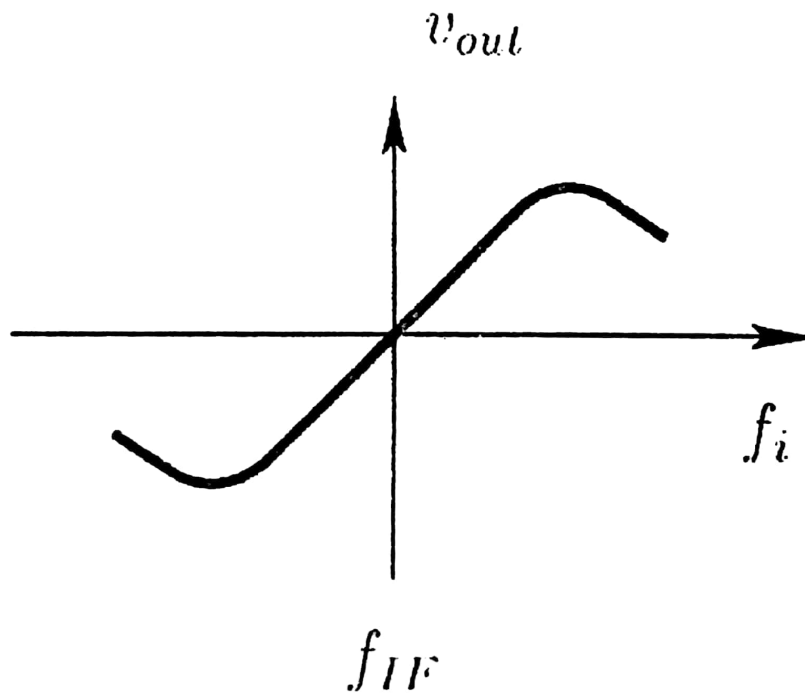
Phase shifter

$$\tilde{s}(t) = A \sin \left[2\pi f_c t + k_f \int_{-\infty}^t m(\alpha) d\alpha + k_f m(t) \right]$$

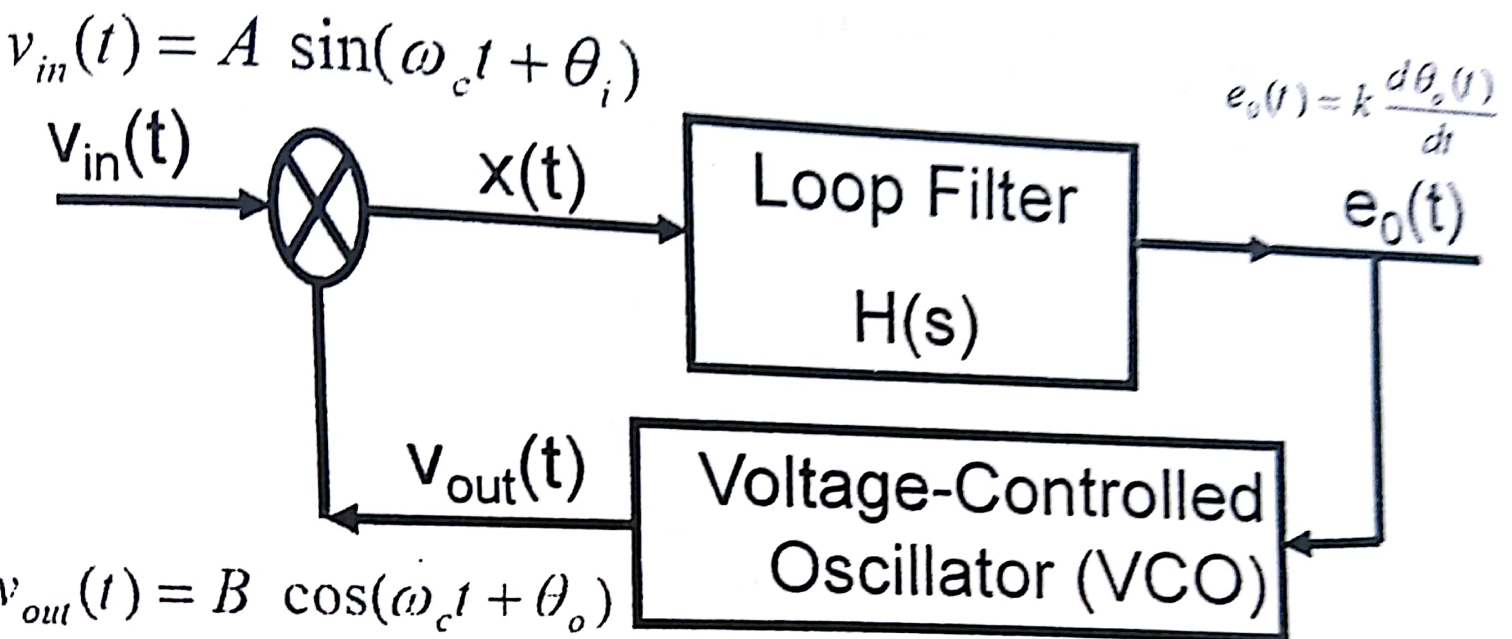
Phase response of phase shifter



Transfer function of Quadrature demodulator



Phase-Locked Loop (PLL)



$$\theta_{in}(t) = k_f \int m(\alpha) d\alpha + \pi/2 \quad \theta_o(t) = k_f \int m(\alpha) d\alpha + \pi/2 - \theta_e(t)$$

$$e(t) = cd\theta_o(t)/dt \approx k_f m(t)$$

Zero-Crossing Detectors

- Zero-Crossing Detectors are also used because of advances in digital integrated circuits.
- These are the frequency counters designed to measure the instantaneous frequency by the number of zero crossings.
- The rate of zero crossings is equal to the instantaneous frequency of the input signal

Summary

- Concepts of instantaneous frequency
- FM and PM signals
- Bandwidth of angle modulated signals
NBFM and WBFM
- Generation of FM signals
 - Direct and indirect generation
- Demodulation of FM signals
 - frequency discriminator
 - PLL



EE325: Chapter 6 (Lec. #1)

Sampling and Pulse Code Modulation

M. A. Smadi

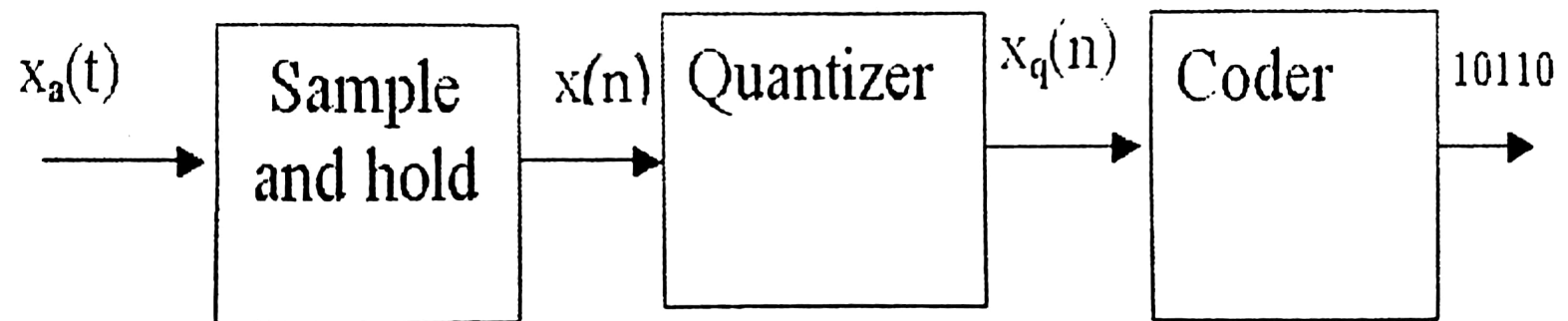
Outline

- introduction
- Sampling and sampling theorem
- Practical sampling and pulse amplitude modulation (PAM)
- Pulse code modulation (PCM)
- Differential pulse code modulation (DPCM)
- Delta modulation.

Introduction

- There is an increase use of digital communication systems
- Digital communications offer several important advantages compared to analog communications such as higher performance, higher security and greater flexibility.
- Digital transmission of analog signals require Analog to Digital conversion (AD).
- Digital pulse modulation

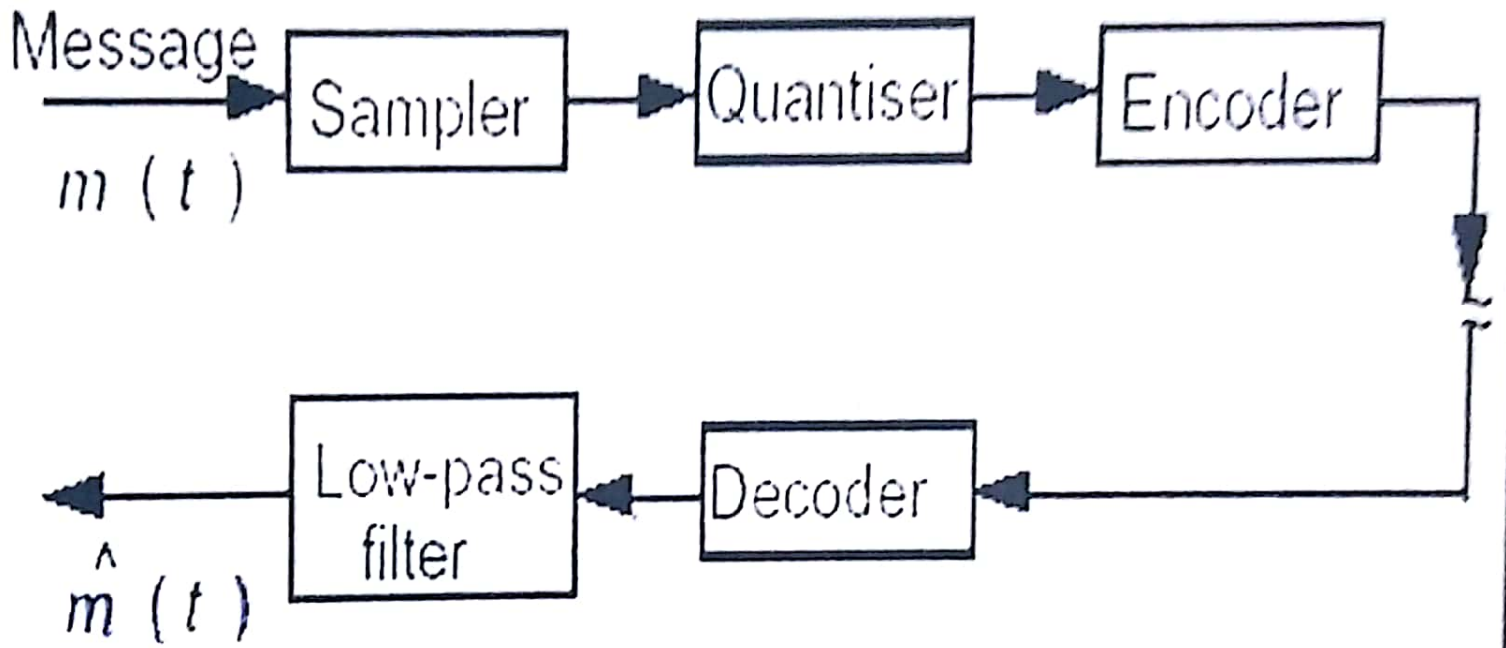
Analog to Digital converter



Quantization error or noise

number of levels 2^n

PCM system



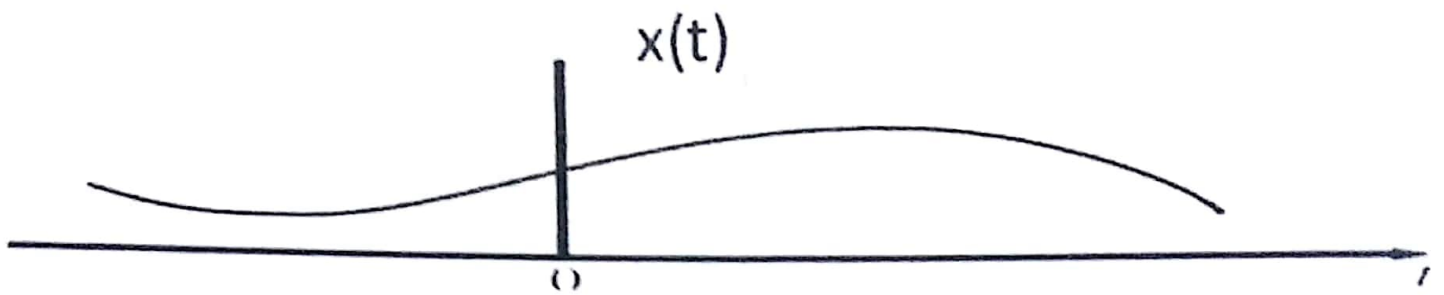
Sampling

- A typical method for obtaining a discrete-time sequence $x(n)$ from a continuous-time signal $x(t)$ is through *periodic sampling*.

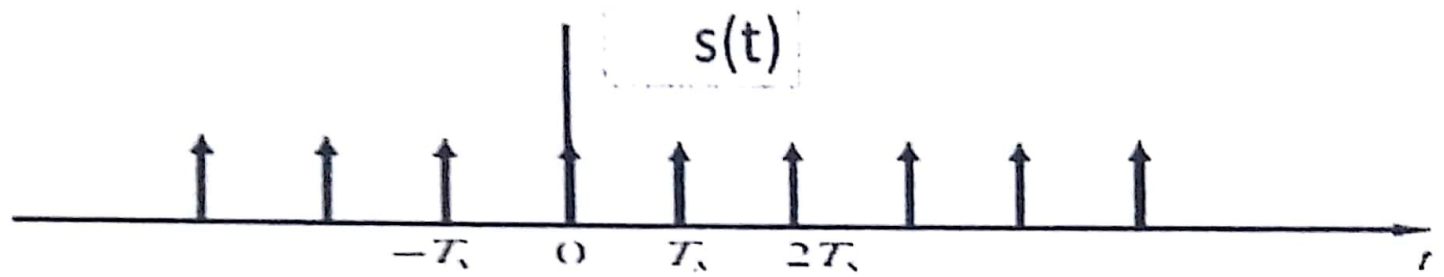
$$x(n) = x(nT_s), \text{ for } -\infty < n < \infty$$

- T_s : sampling period.
- f_s : sampling frequency
or sampling rate

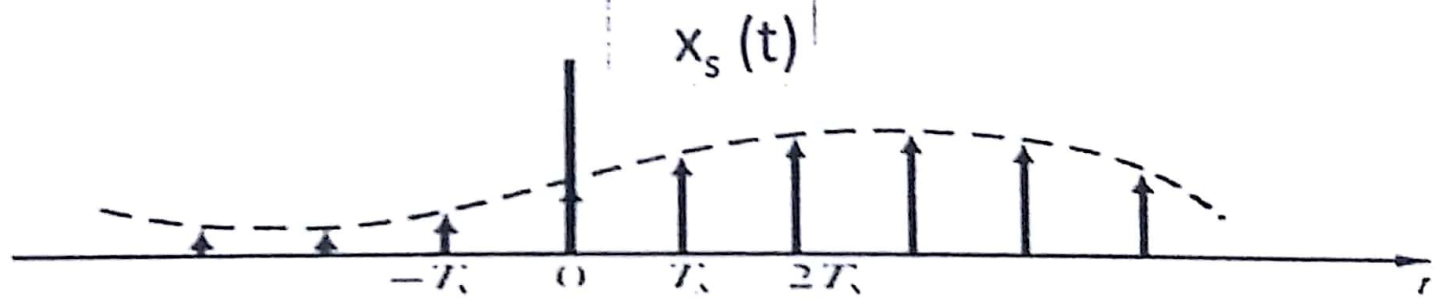
$$\frac{1}{T_s} = f_s$$



(a)

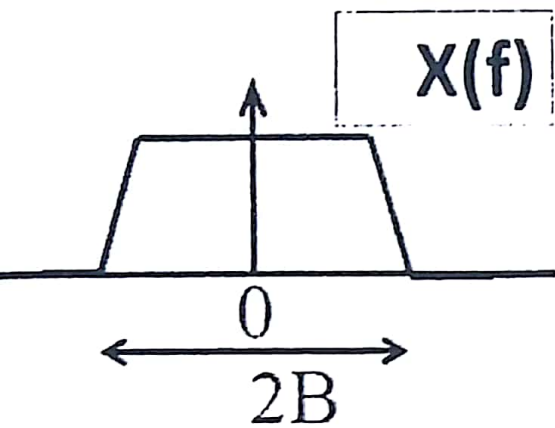


(b)

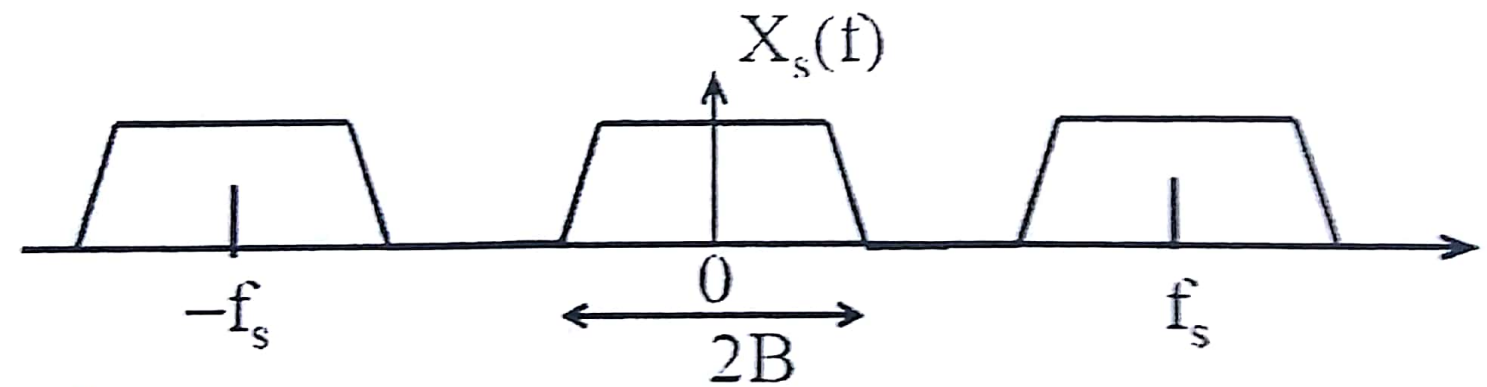


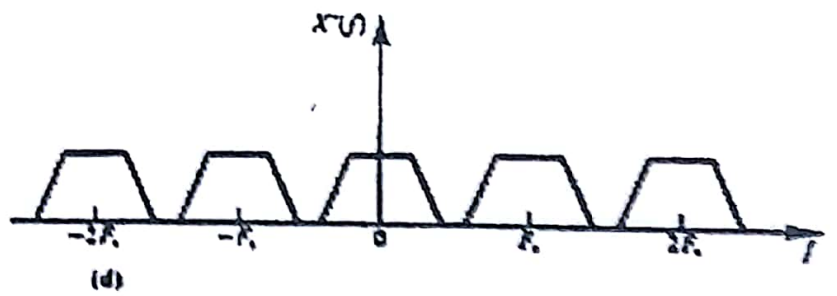
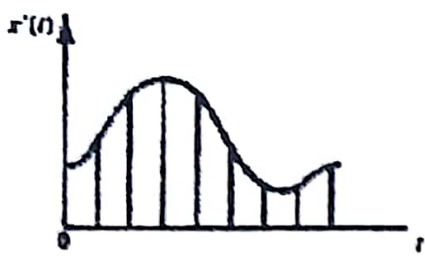
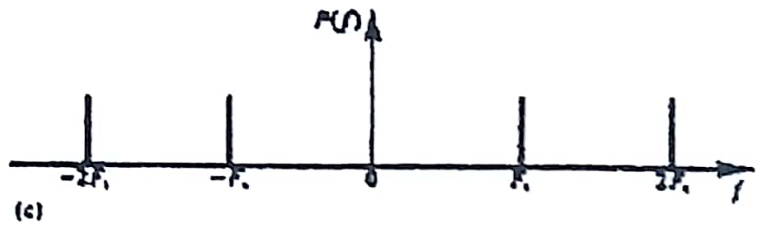
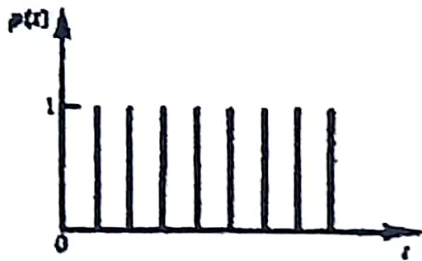
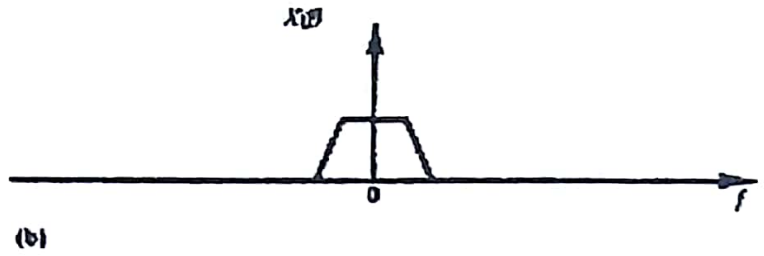
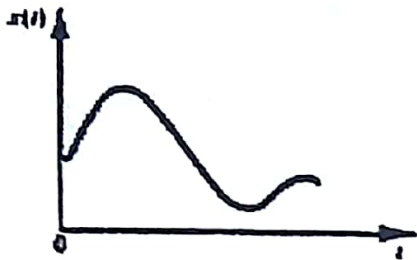
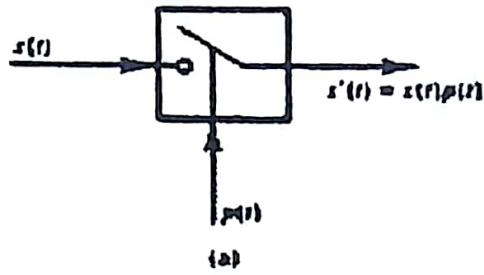
(c)

Spectrum of $X_s(f)$

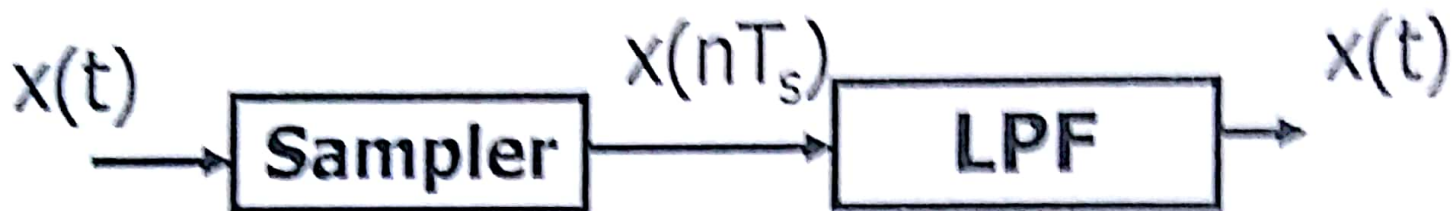


$$X_s(f) = \frac{1}{T_s} \sum_{k=-\infty}^{\infty} X(f - kf_s)$$





Sampling

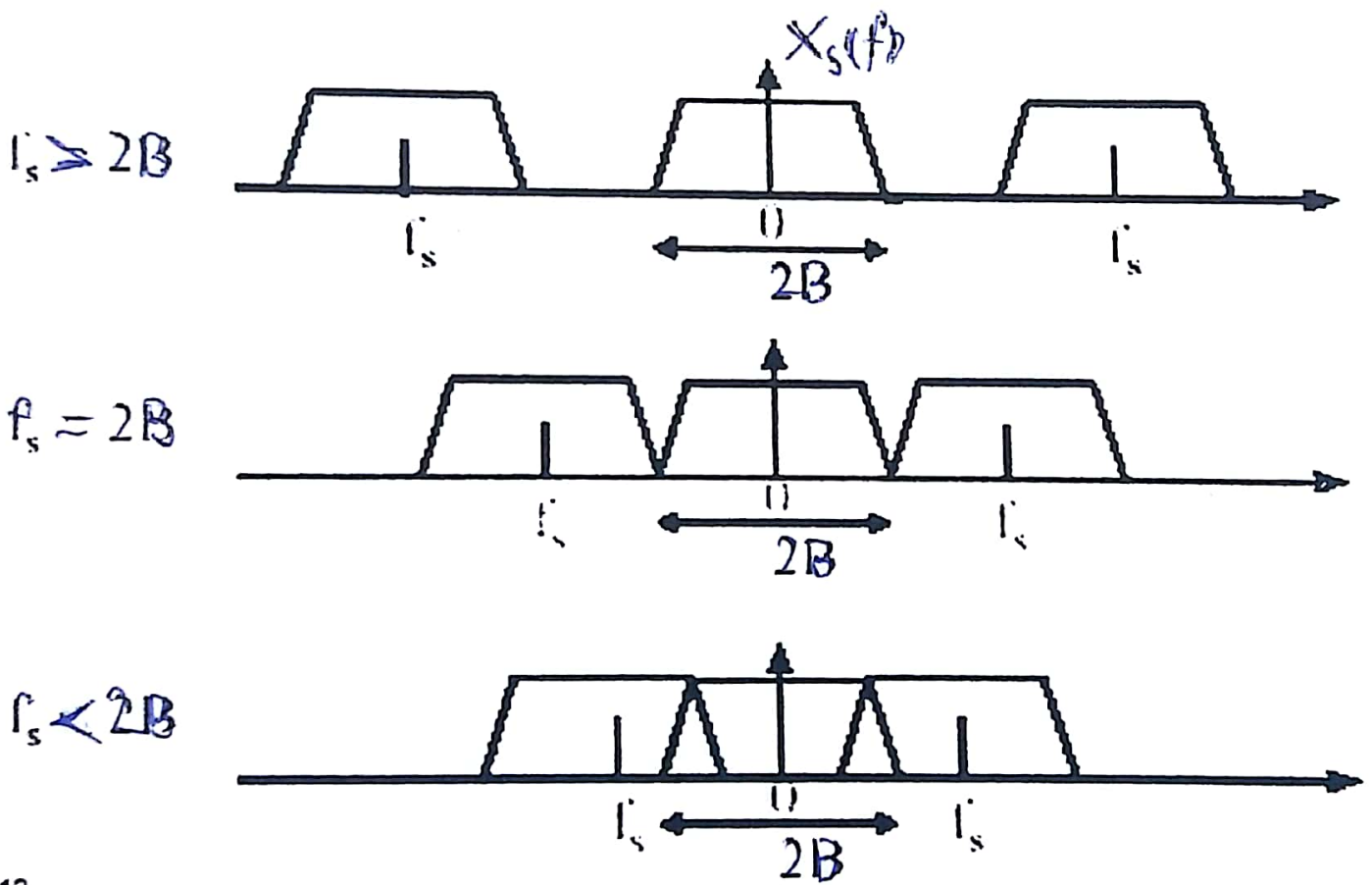


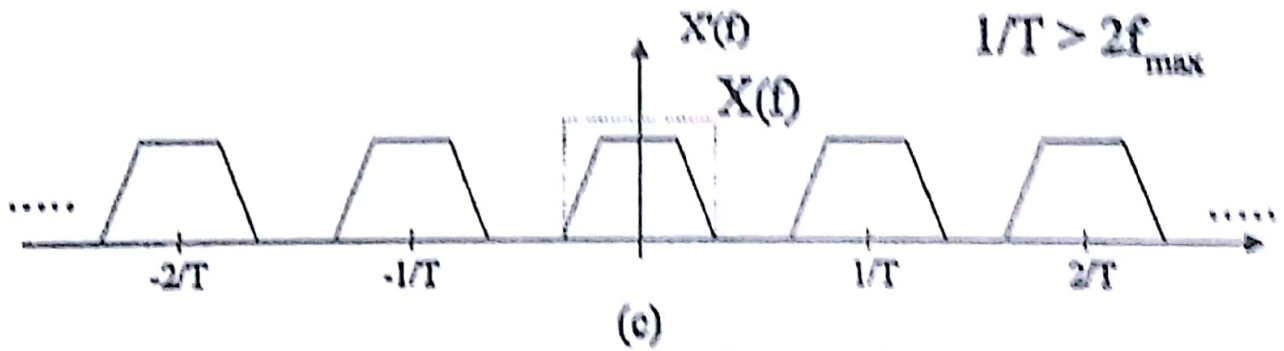
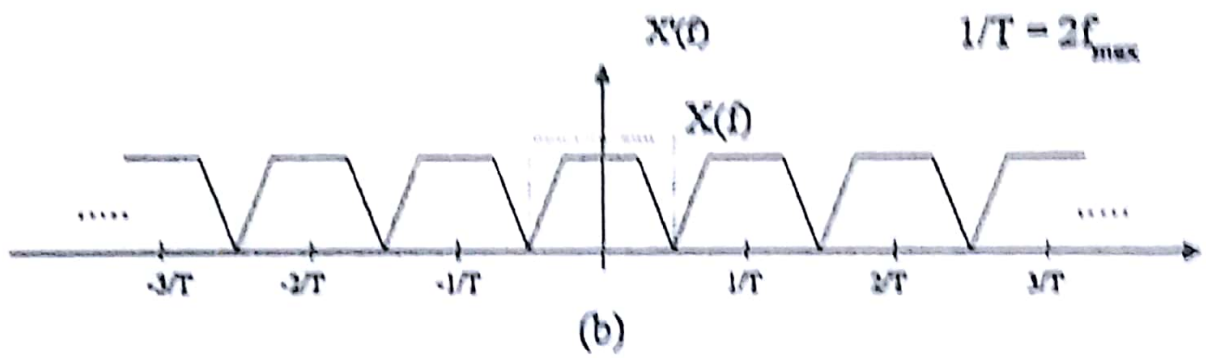
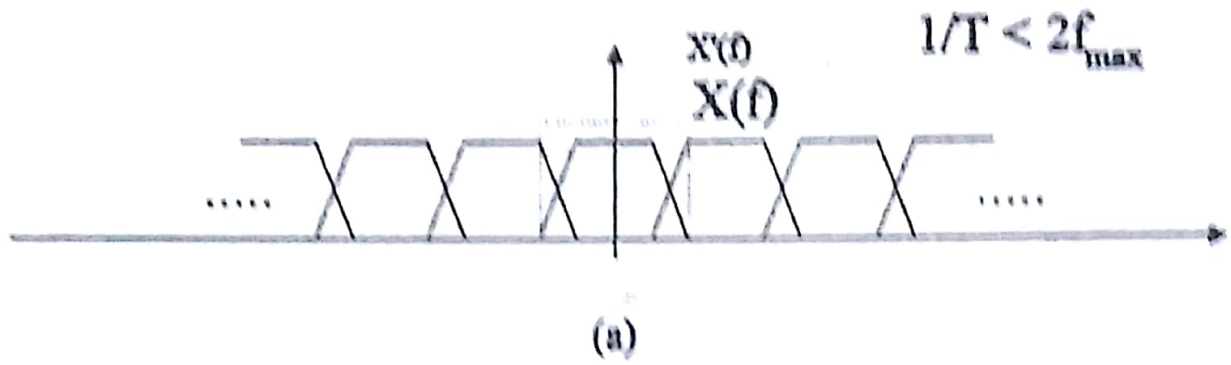
- Is it possible to reconstruct the analog signal from the sampled values?

Sampling

- Given any analog signal, how should we select the sampling period T_s (or the sampling frequency f_s) without losing the important information contained in the signal.

Spectrum of $X_s(f)$





Sampling Theorem

- Let $m(t)$ be a real valued band-limited signal having a bandwidth B , and $m(nT_s)$ be the sample values of $m(t)$ where n is an integer.
- The *sampling theorem* states that the signal $m(t)$ can be reconstructed from $m(nT_s)$ with no distortion if the *sampling frequency*

$$f_s \geq 2B$$

- The minimum sampling rate $2B$ is called the *Nyquist sampling rate*.

Typical sampling rates for some common applications

Application	B	f_s
Speech	4 kHz	8 kHz
Audio	20 kHz	40 kHz
Video	4 MHz	8 MHz

Example

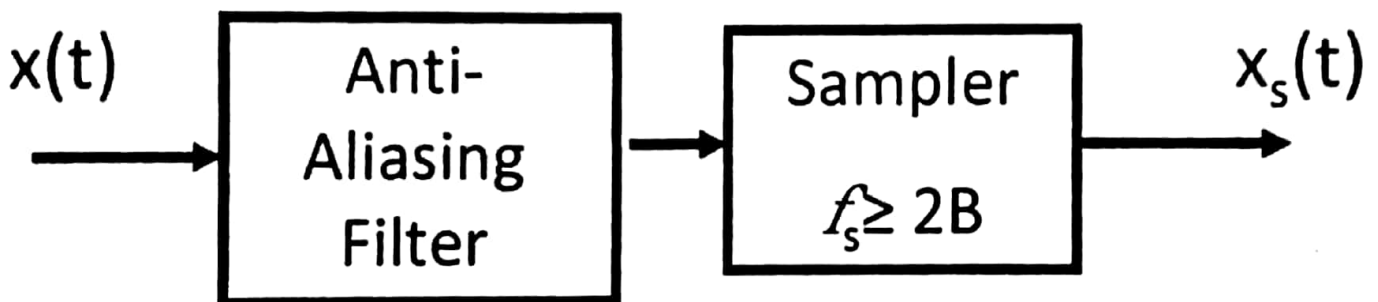
- Determine the Nyquist rate of the following analog signal and plot the spectrum of the sampled signal for :

1. $f_s=150\text{Hz}$ 2. $f_s=300\text{Hz}$ 3. $f_s=500\text{ Hz}$

$$\mathbf{x(t) = 3\cos(50\pi t) + 10\sin(300\pi t) - \cos(100\pi t)}$$

To Avoiding aliasing

- Band-limiting signals (by filtering) before sampling.
- Sampling at a rate that is greater than the Nyquist rate.





EE325: Chapter 6 (Lec. #2)

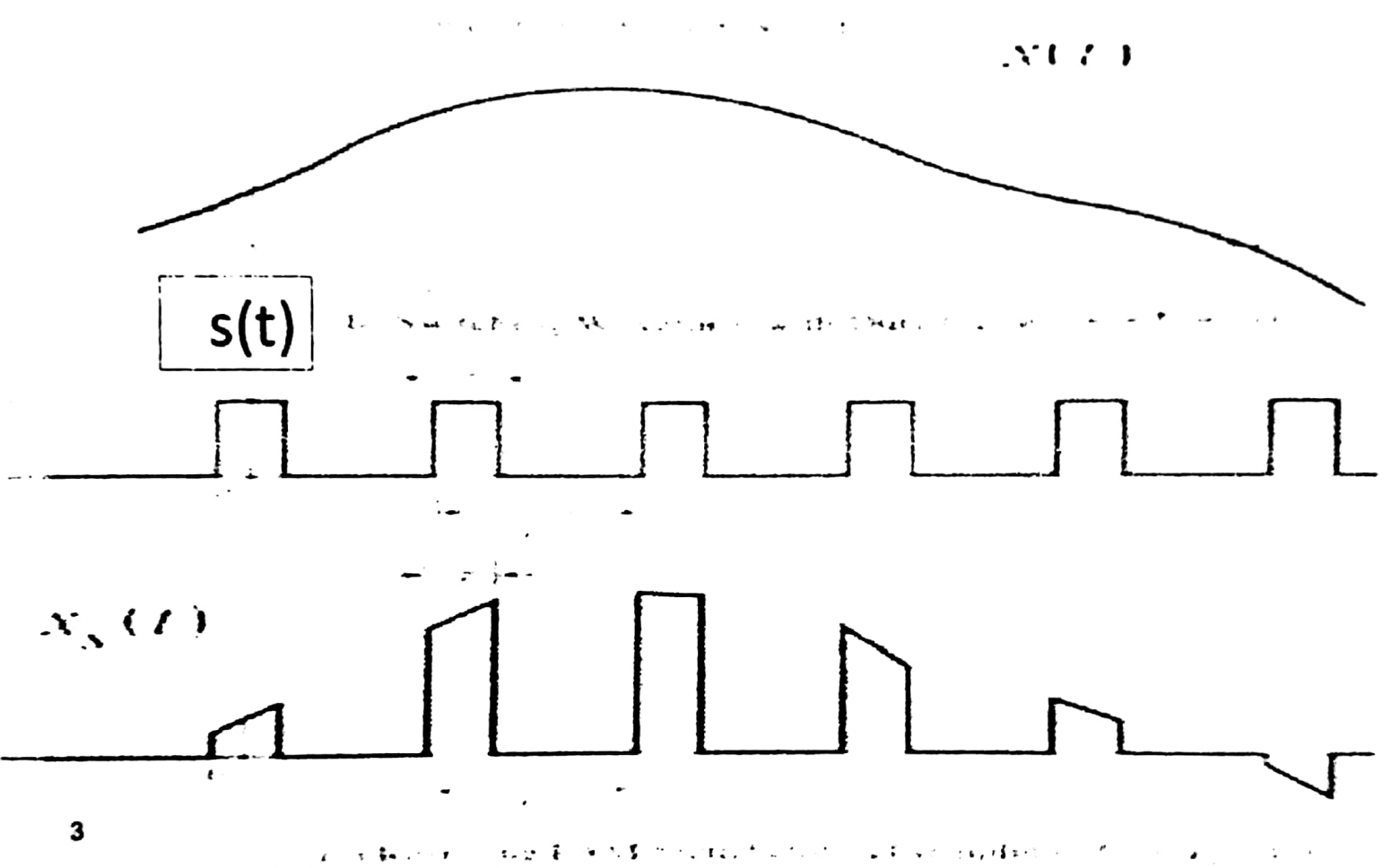
Sampling and Pulse Code Modulation

M. A. Smadi

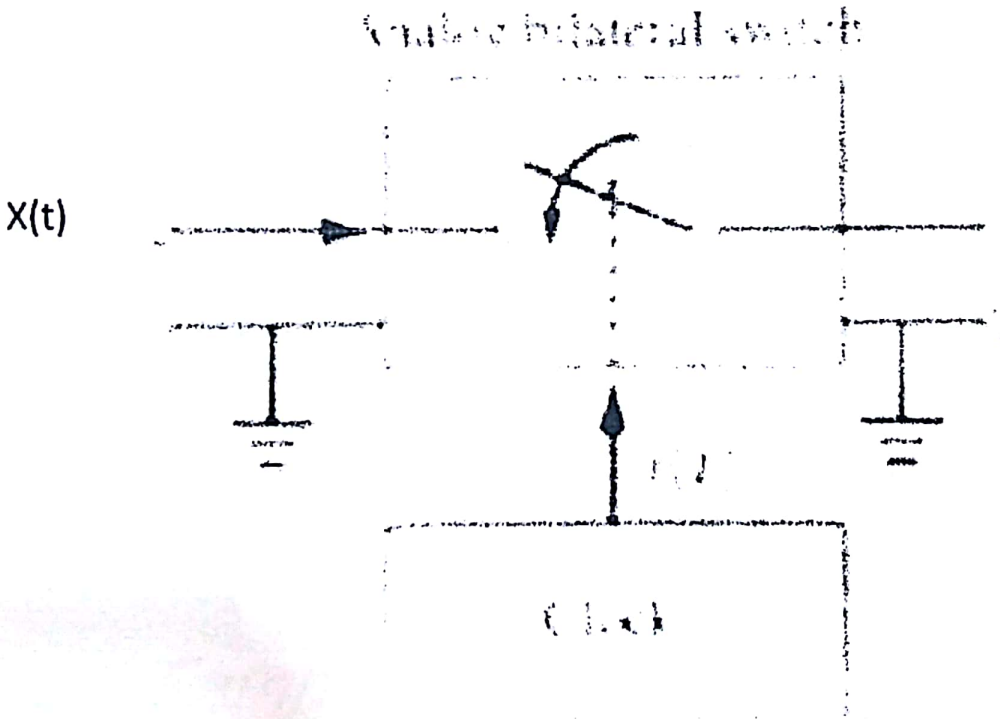
Practical Sampling

- In practice, we multiply a signal $x(t)$ by a train of pulses of finite width.
- There are two types of practical sampling
 - Natural Sampling (Gating)
 - Instantaneous Sampling. Also known as flat-top PAM or sample-and-hold.

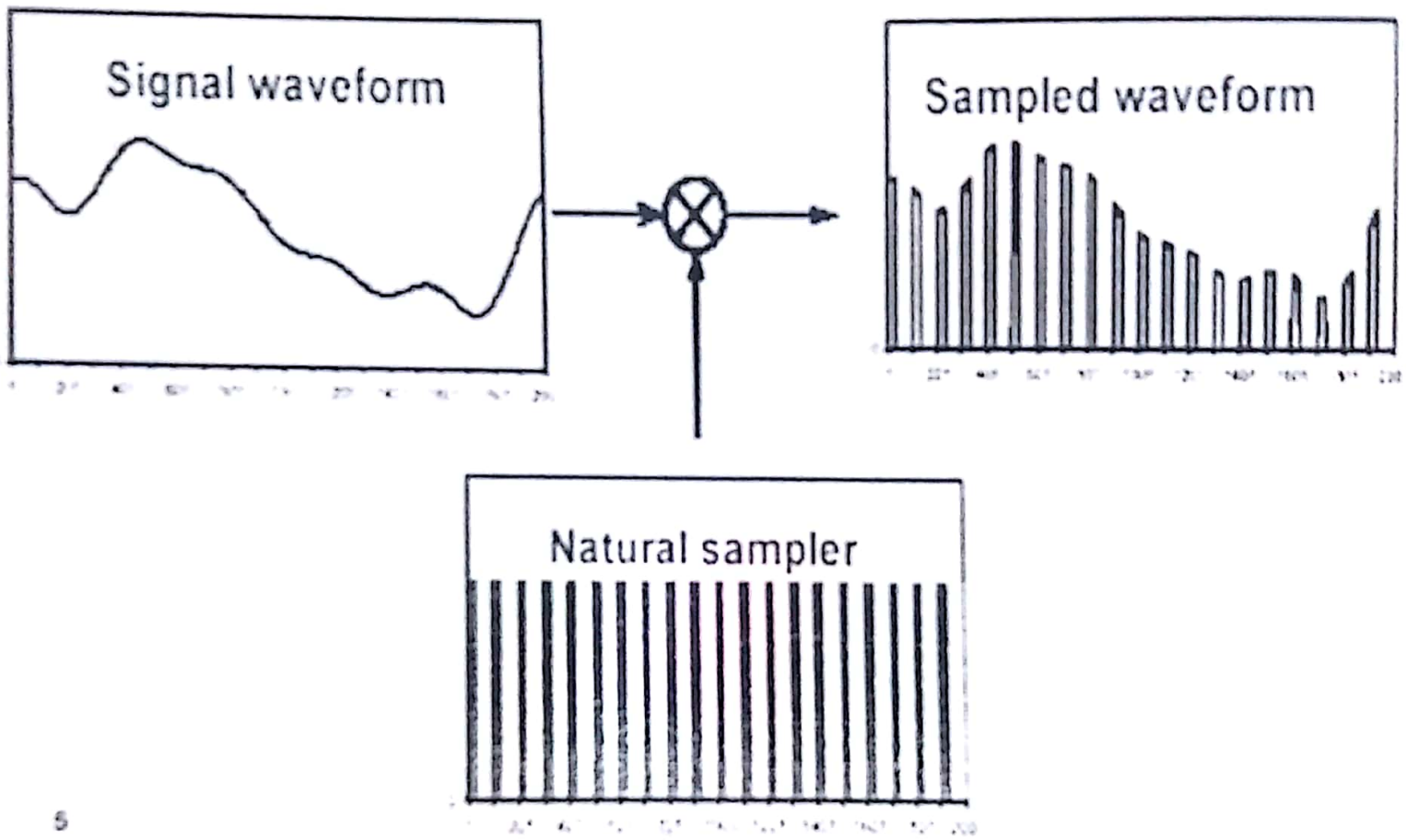
Natural Sampling



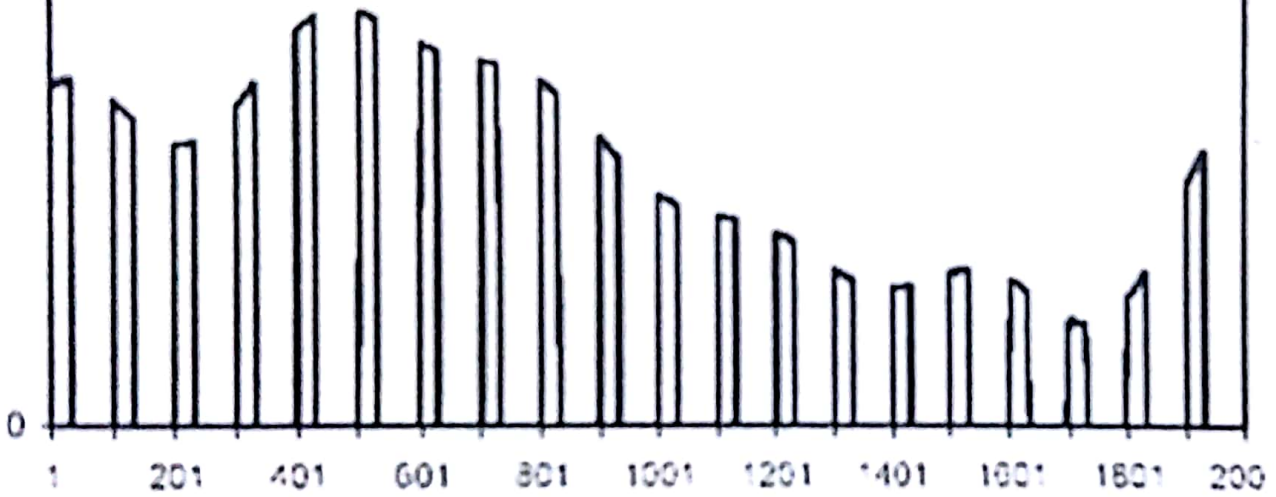
Generation of PAM with natural sampling



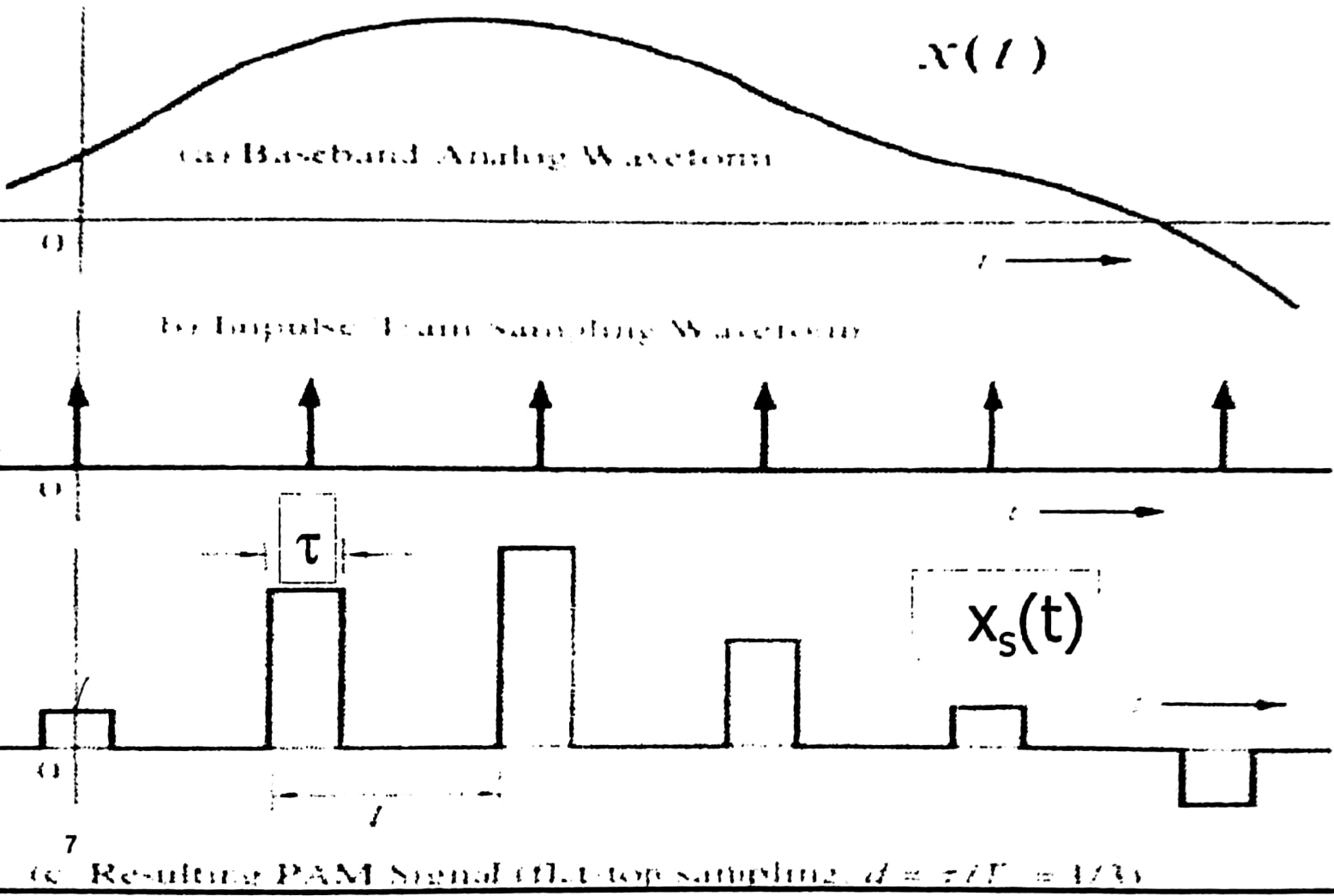
Another example of natural sampling

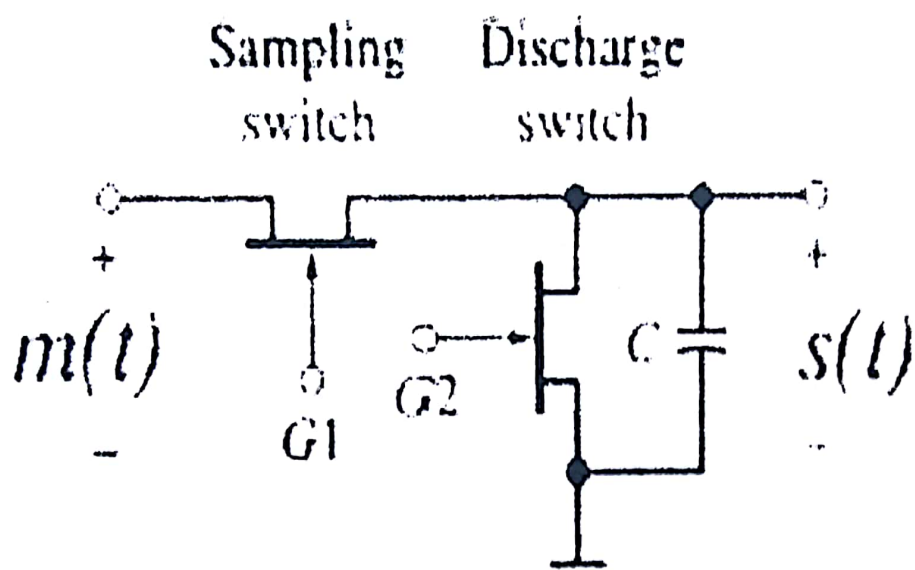


Sampled waveform



Sample-and-Hold



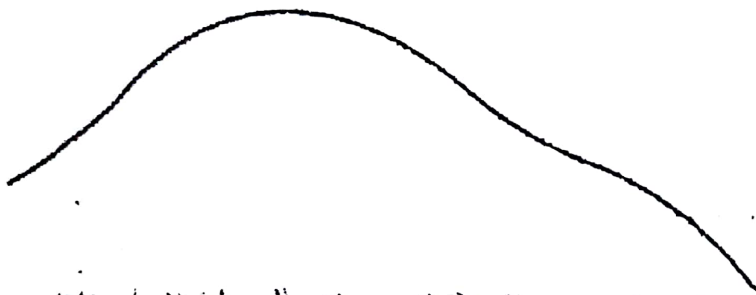


Sample-and-hold (S/H) circuit.

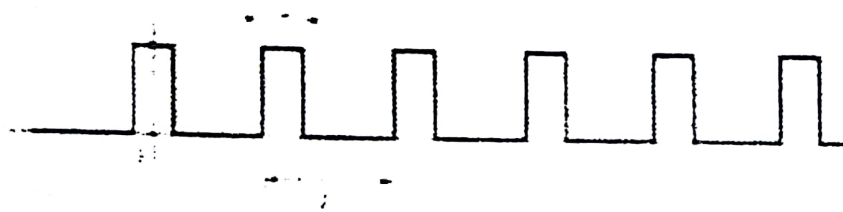
Natural Sampling (Gating)

(a) Plot of a signal waveform

$x(t)$

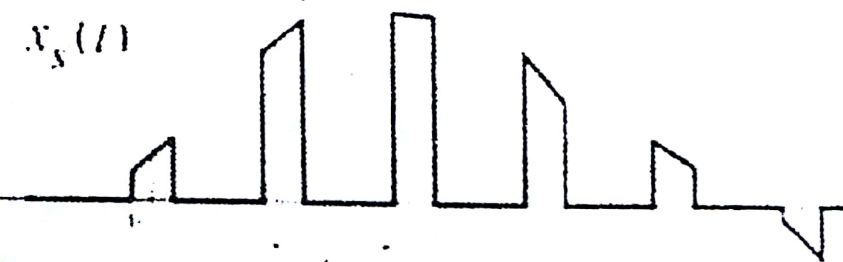


(b) Plot of a sampling waveform with period T_s



$$s(t) = \sum_{k=-\infty}^{\infty} \text{rect}\left(\frac{t - kT_s}{\tau}\right)$$

$x_s(t)$



$$x_s(t) = x(t) s(t)$$

(c) Plot of the natural sampling waveform

Natural Sampling (Gating): Spectrum

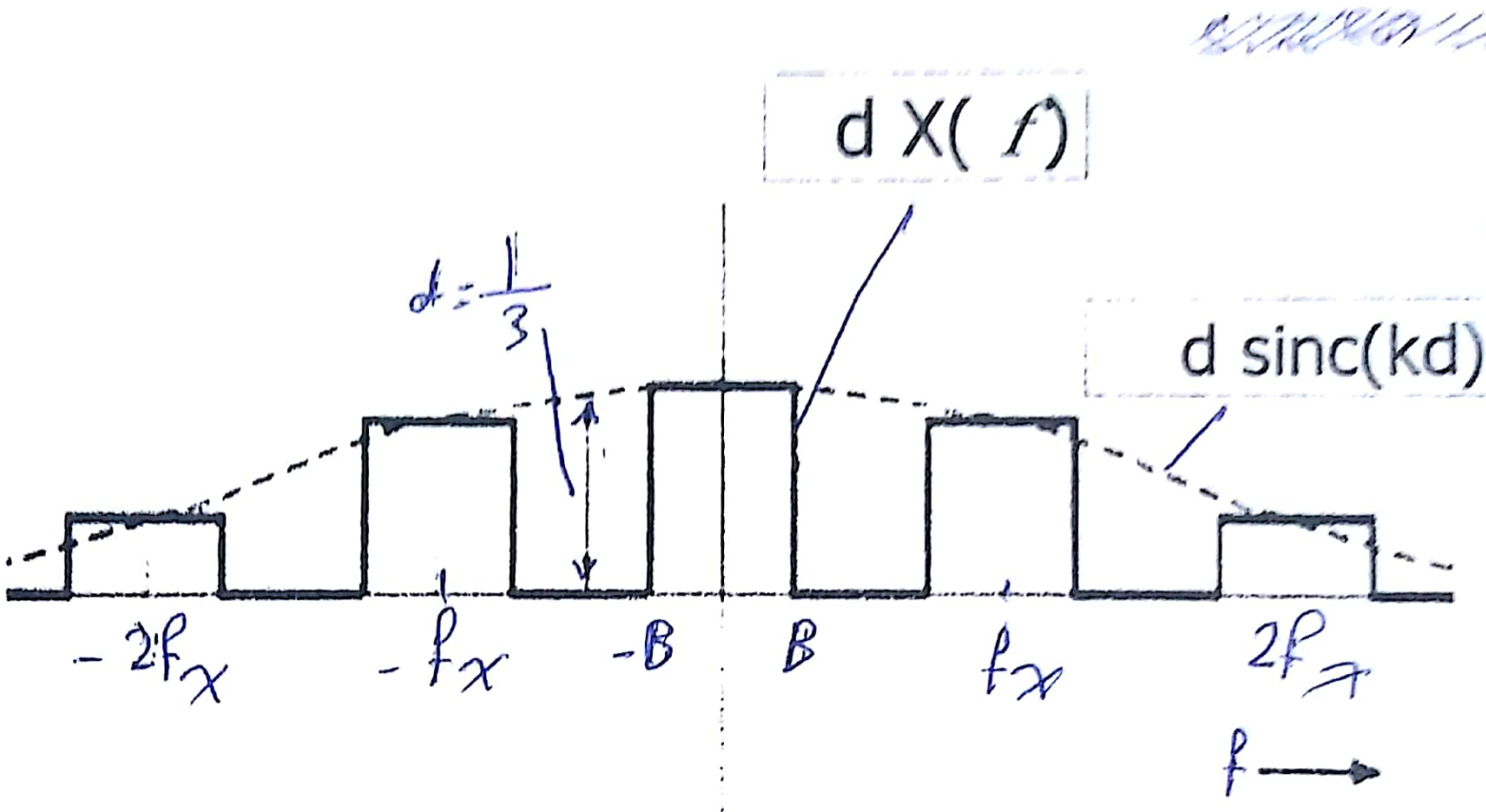
- The spectrum (FT) of the sampled (PAM) signal is

$$X_s(f) = d \sum_{k=-\infty}^{\infty} \text{sinc}(kd) X(f - kf_s)$$

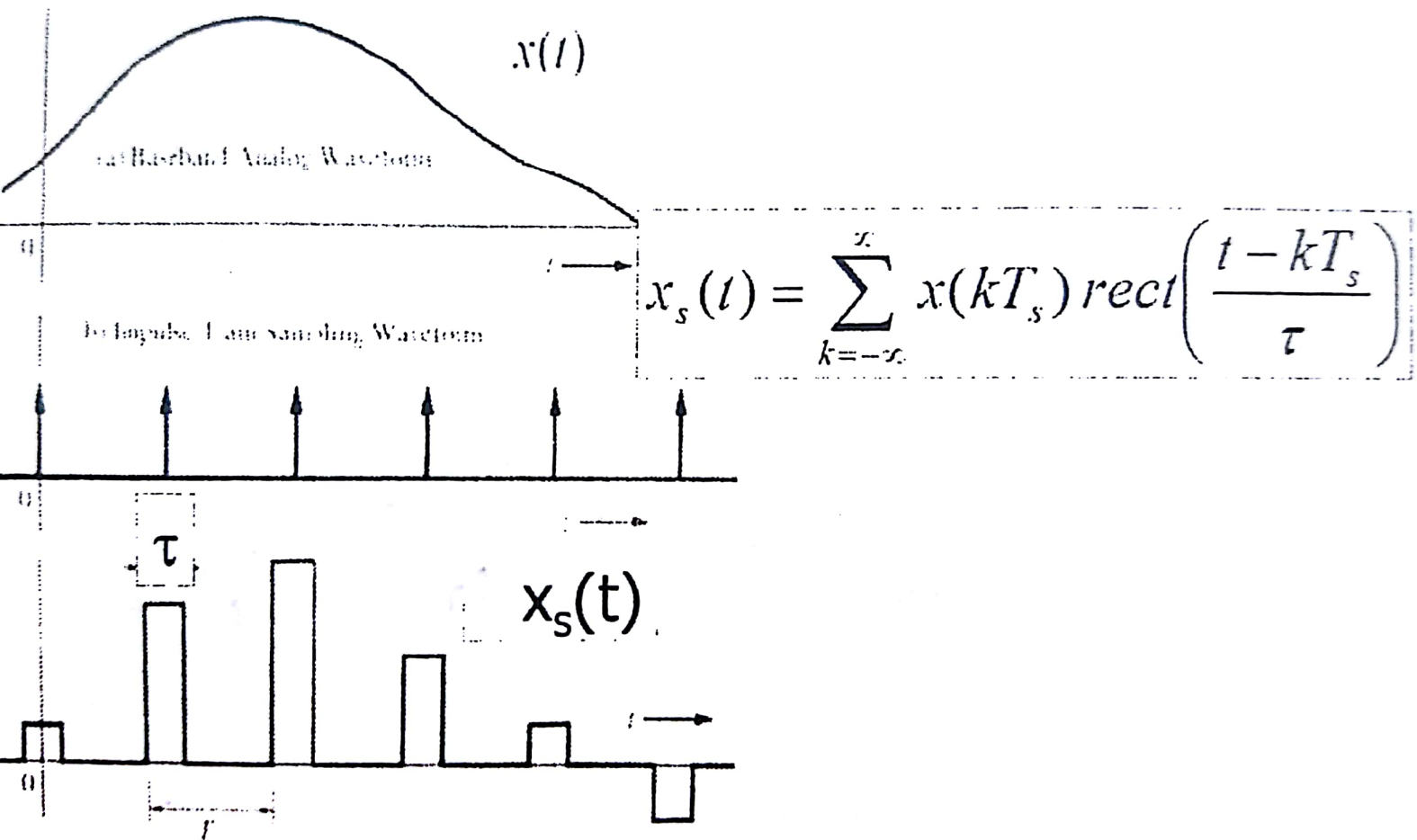
$$d = \frac{\tau}{T_s}$$

Duty cycle of $s(t)$

Natural Sampling (Gating): Spectrum



Sample-and-Hold(flat-top sampling)



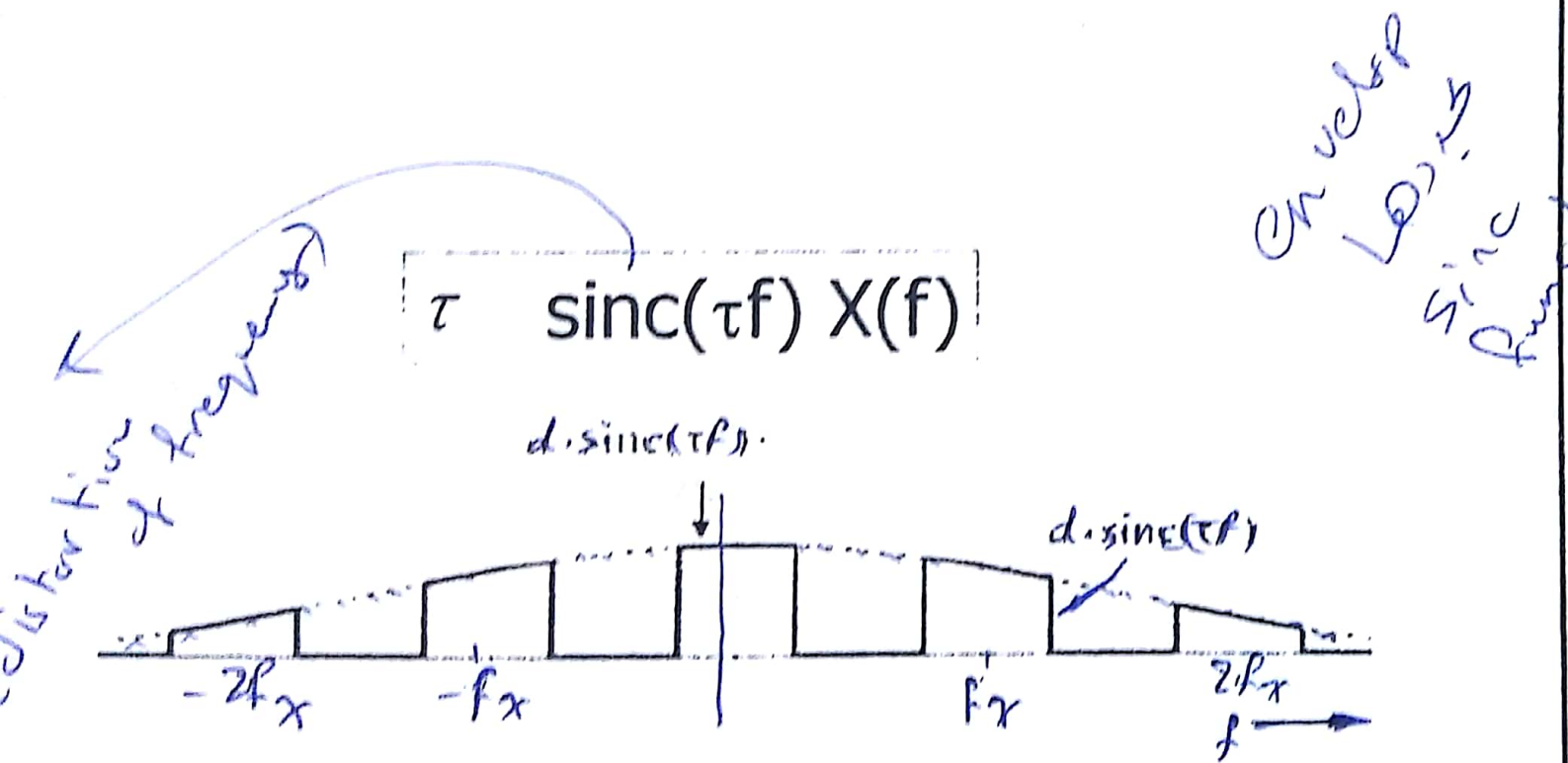
Sample and hold: Spectrum

$$\begin{aligned}
 x_s(t) &= \sum_{k=-r}^r x(kT_s) \operatorname{rect}\left(\frac{t - kT_s}{\tau}\right) \\
 &= \operatorname{rect}\left(\frac{t}{\tau}\right) * \sum_{k=-r}^r x(kT_s) \delta(t - kT_s)
 \end{aligned}$$

Handwritten annotations:
 - An arrow points from the $\operatorname{rect}\left(\frac{t - kT_s}{\tau}\right)$ term to the word "conv".
 - The asterisk $*$ in the second equation is circled, with an arrow pointing to the word "conv".
 - The word "rect" is written below the first $\operatorname{rect}\left(\frac{t}{\tau}\right)$ term.
 - The word "conv" is written below the asterisk.
 - The phrase "ideally sampling s" is written below the second equation.

$$X_s(f) = \tau \operatorname{sinc}(f\tau) \sum_{k=-r}^r X(f - kf_s)$$

Sample and hold: Spectrum



لہذا یہ لہذا ہو کر رہے
یا کہہ سکتے ہیں

- we see that by using flat-top sampling we have introduced amplitude distortion, and the primary effect is an attenuation of high-frequency components. This effect is known as the *aperture effect*.

- If $\tau \ll T_s$, then $H(f)$ represents a LPF.

- Else, we can use a LPF such that

$$H_{eq}(f) = 1/H(f)$$

The LPF is called an *equalization filter*.

Reasons for intentionally lengthening the duration of each sample are:

- Reduce the required transmission bandwidth: B is inversely proportional to pulse duration
- To get the exact signal value, the transient must fade away



EE325: Chapter 6 (Lec. #3)

Sampling and Pulse Code Modulation

M. A. Smadi

Pulse Modulation

- Pulse modulation results when some characteristic of a pulse is made to vary in one-to-one correspondence with the message signal.
- A pulse is characterized by three qualities:
 - Amplitude
 - Width \longrightarrow عرض
 - Position \longrightarrow A , t_s نَبْت
- Pulse amplitude modulation, Pulse width modulation, and Pulse position modulation

Pulse amplitude modulation (PAM)

- In Pulse Amplitude Modulation, a pulse is generated with an amplitude corresponding to that of the modulating waveform.
- There are two types of PAM sampling
 - Natural Sampling (Gating)
 - Flat-top or sample-and-hold.

PAM System

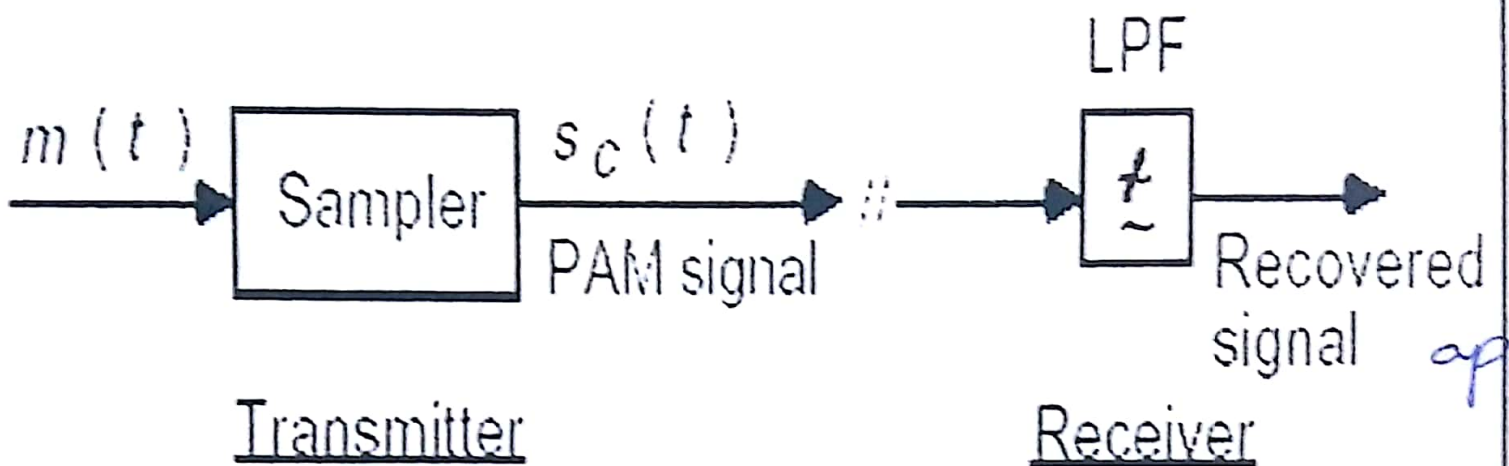



Figure 12.6 PAM system.

- A system transmitting sample values of the analog signal is called a *pulse-amplitude modulation (PAM)* system.

PAM

- Like AM, PAM is very sensitive to noise. 
- While PAM was deployed in early AT&T Dimension PBXs, there are no practical implementations in use today. However, PAM is an important first step in a modulation scheme known as Pulse Code Modulation.

Note

- PBX: Short for *private branch exchange*, a private telephone network used within an enterprise.
- Users of the PBX share a certain number of *outside lines* for making telephone calls external to the PBX.

Pulse Width Modulation (PWM)

- In PWM, pulses are generated at a regular rate. The length of the pulse is controlled by the modulating signal's amplitude.

سپیکٹرم میں پلسز کی چوڑائی کنٹرول کی جاتی ہے ← width پلس *
پلسز کی چوڑائی کنٹرول کی جاتی ہے ← BW اور اس کا

انہم کا شمار در T_c حصہ ہے مابقی شرح مدلل

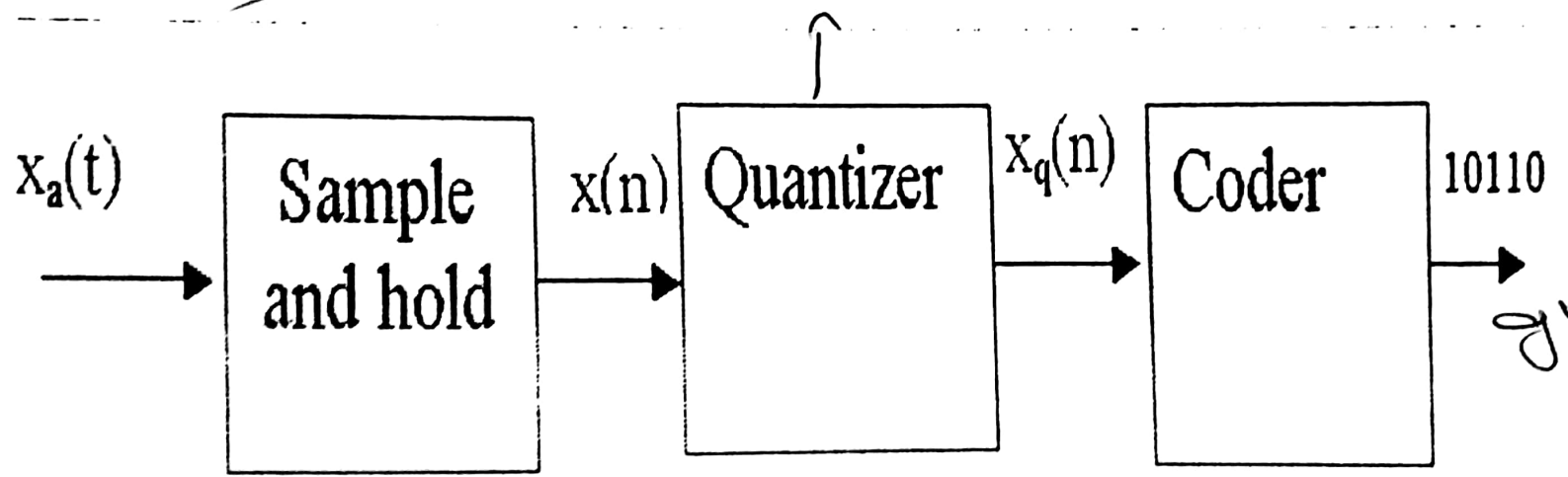
Pulse Position Modulation (PPM)

- PPM is a scheme where the pulses of equal amplitude are generated at a rate controlled by the modulating signal's amplitude.

10

← **Pulse Code Modulation**

Amplitude distortion

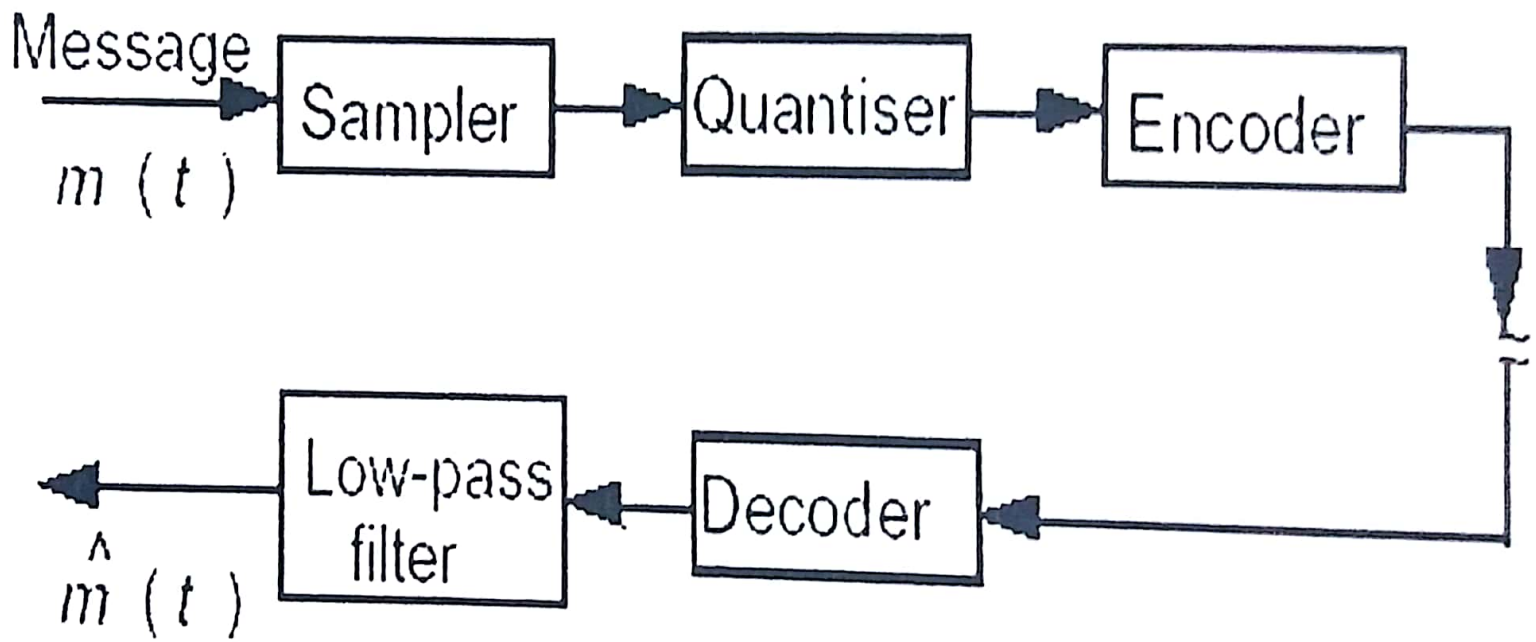


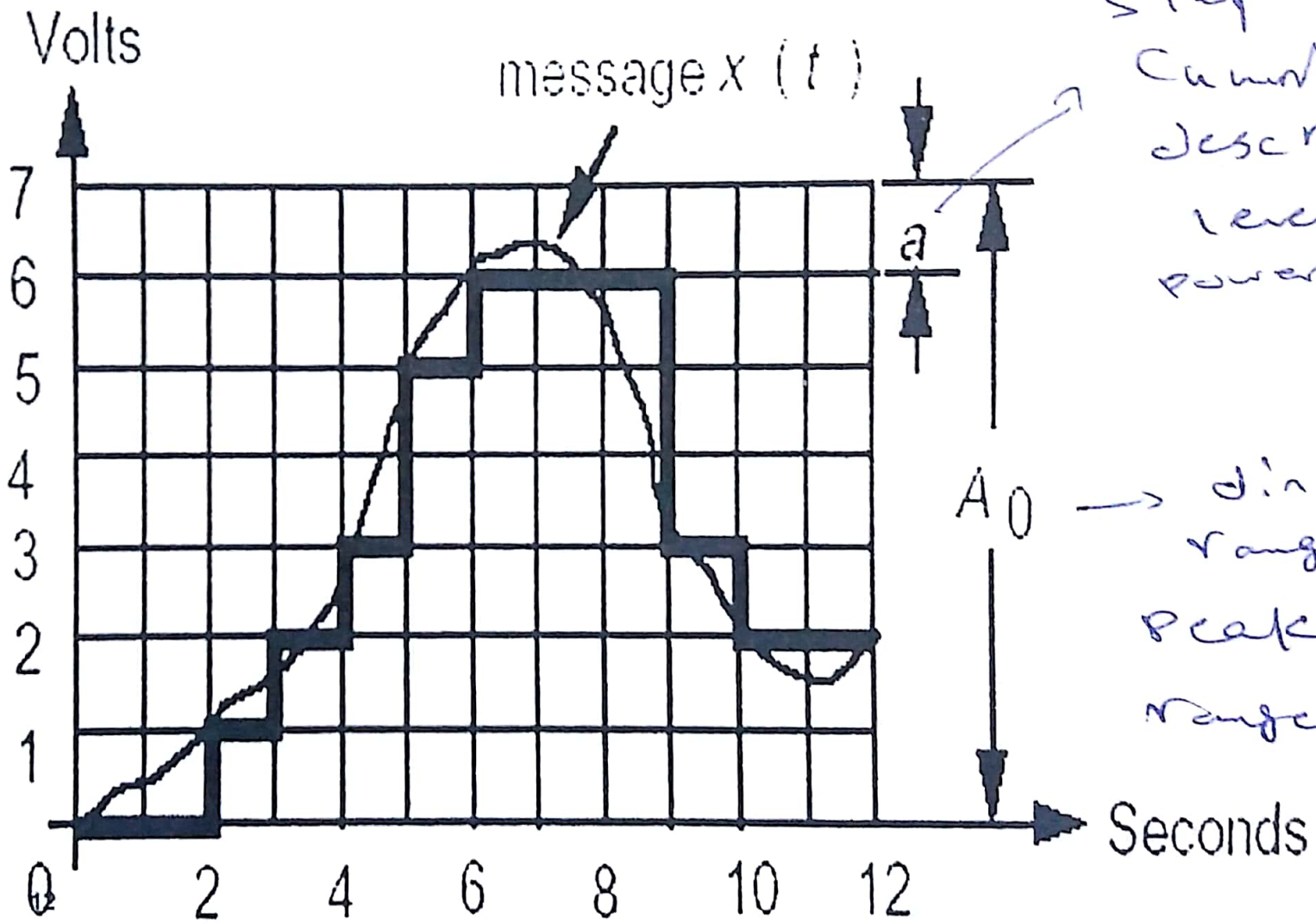
low → pure digital

Advantages of PCM

- Inexpensive digital circuitry may be used in the system.
- All-digital transmission.
- Further digital signal processing such as encryption is possible.
- Errors may be minimized by appropriate coding of the signals.
- Signals may be regularly reshaped or regenerated using *repeaters* at appropriate intervals.

A single-channel PCM transmission system



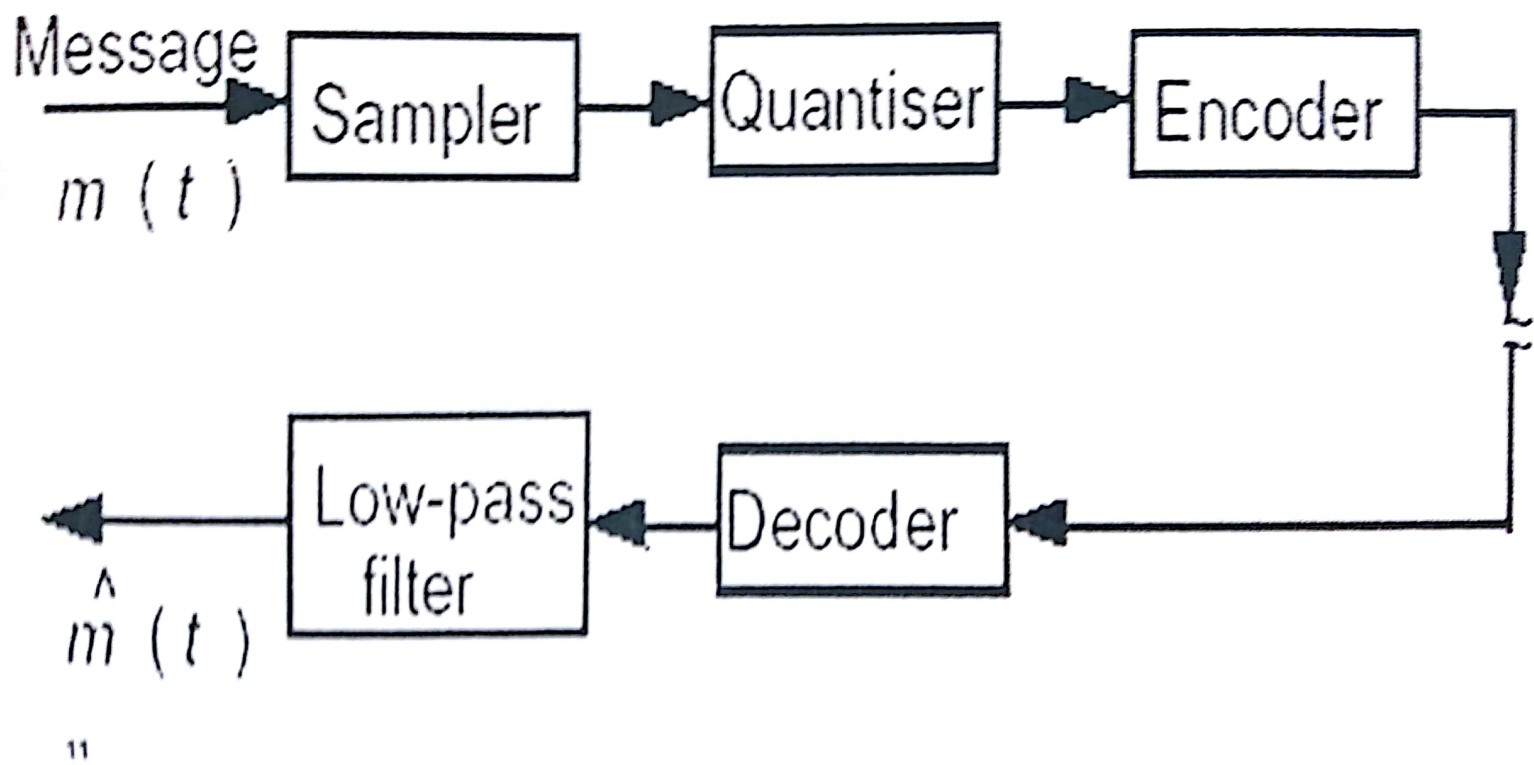


15

Advantages of PCM

- Inexpensive digital circuitry may be used in the system.
- All-digital transmission.
- Further digital signal processing such as *encryption* is possible.
- Errors may be minimized by appropriate coding of the signals.
- Signals may be regularly reshaped or regenerated using *repeaters* at appropriate intervals.

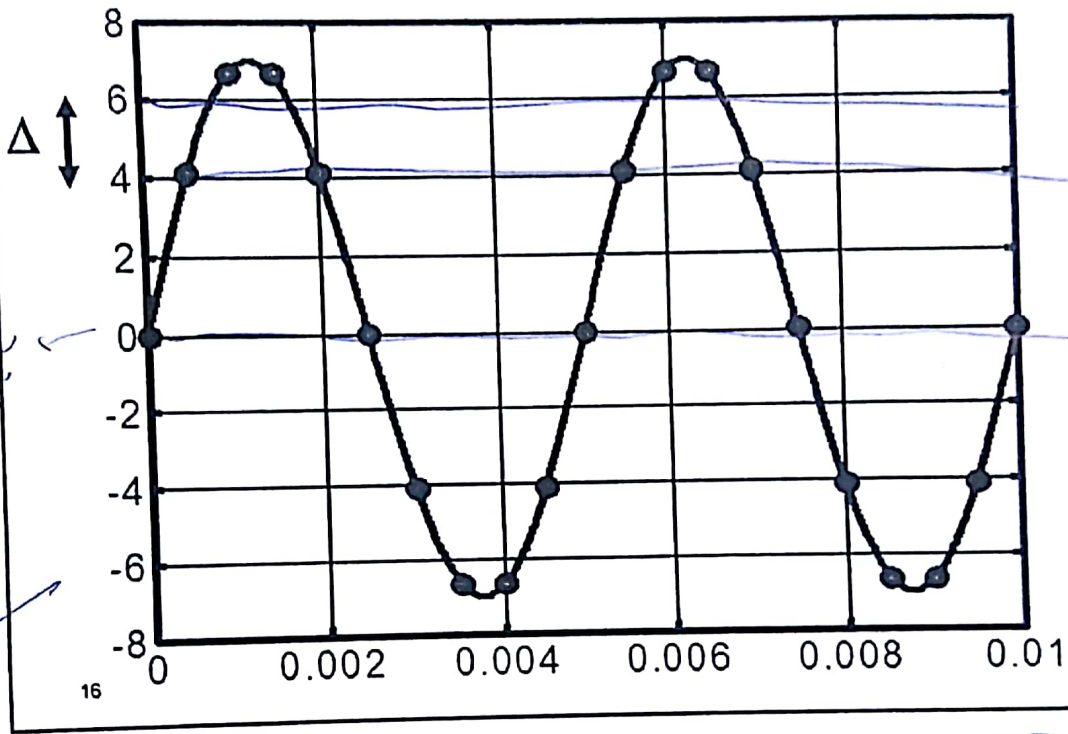
A single-channel PCM transmission system



Quantization

- Quantizer converts the discrete time signal into a sampled and quantized signal that is discrete in both time and amplitude

$m(t)$ and its sampled value $m(kT_s)$



Quantization
Step Size
Sample Rate

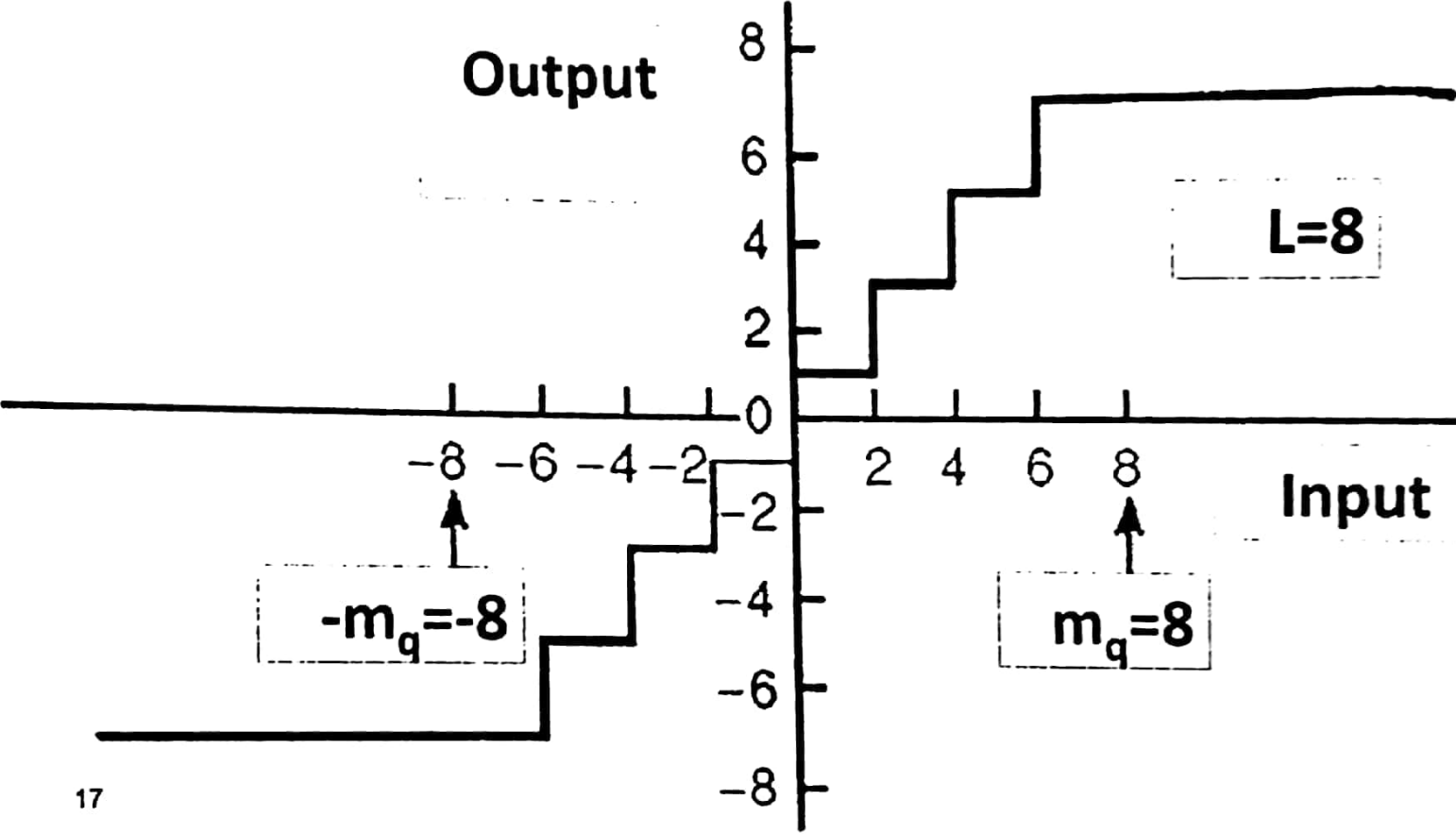
Signal Frequency
 5000 Hz

Sampling Rate
+ Quantize
Step Size
 $\Delta = 2 \text{ Volt}$

Quantization
results in
8 steps or
bits

$$\frac{16}{2} \rightarrow A_0$$

Input-output characteristics of the quantizer



- Quantization can be ***uniform*** and ***nonuniform***
- The quantization discussed so far is said to be ***uniform*** since all of the steps Δ are of equal size.
- Nonuniform quantization uses unequal steps

Uniform Quantization

- The amplitude of $m_s(t)$ can be confined to the range $[-m_q, m_q]$
- This range can be divided in L zones, each of step Δ such that

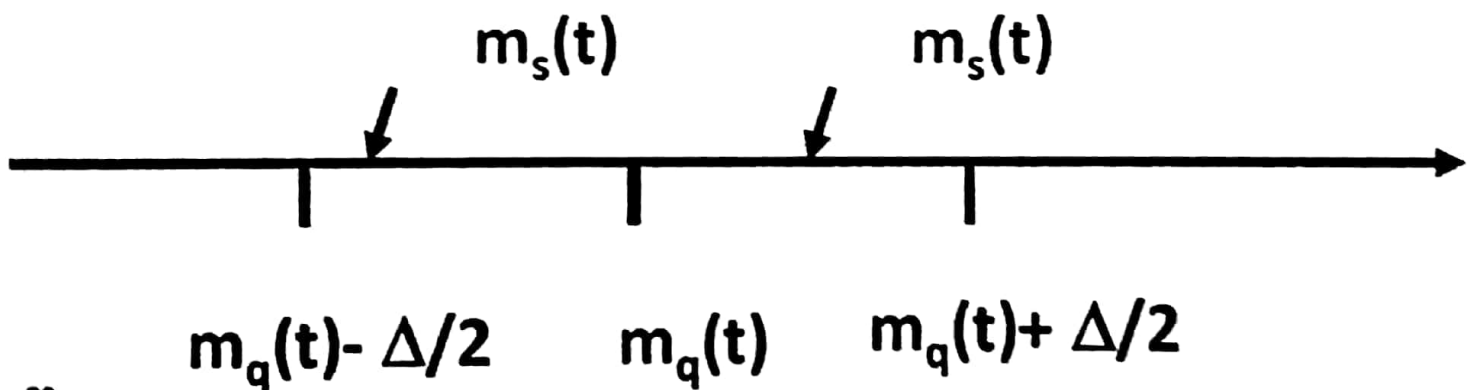
$$\Delta = 2 m_q / L$$

- The sample amplitude value is approximated by the midpoint of the interval in which it lies.

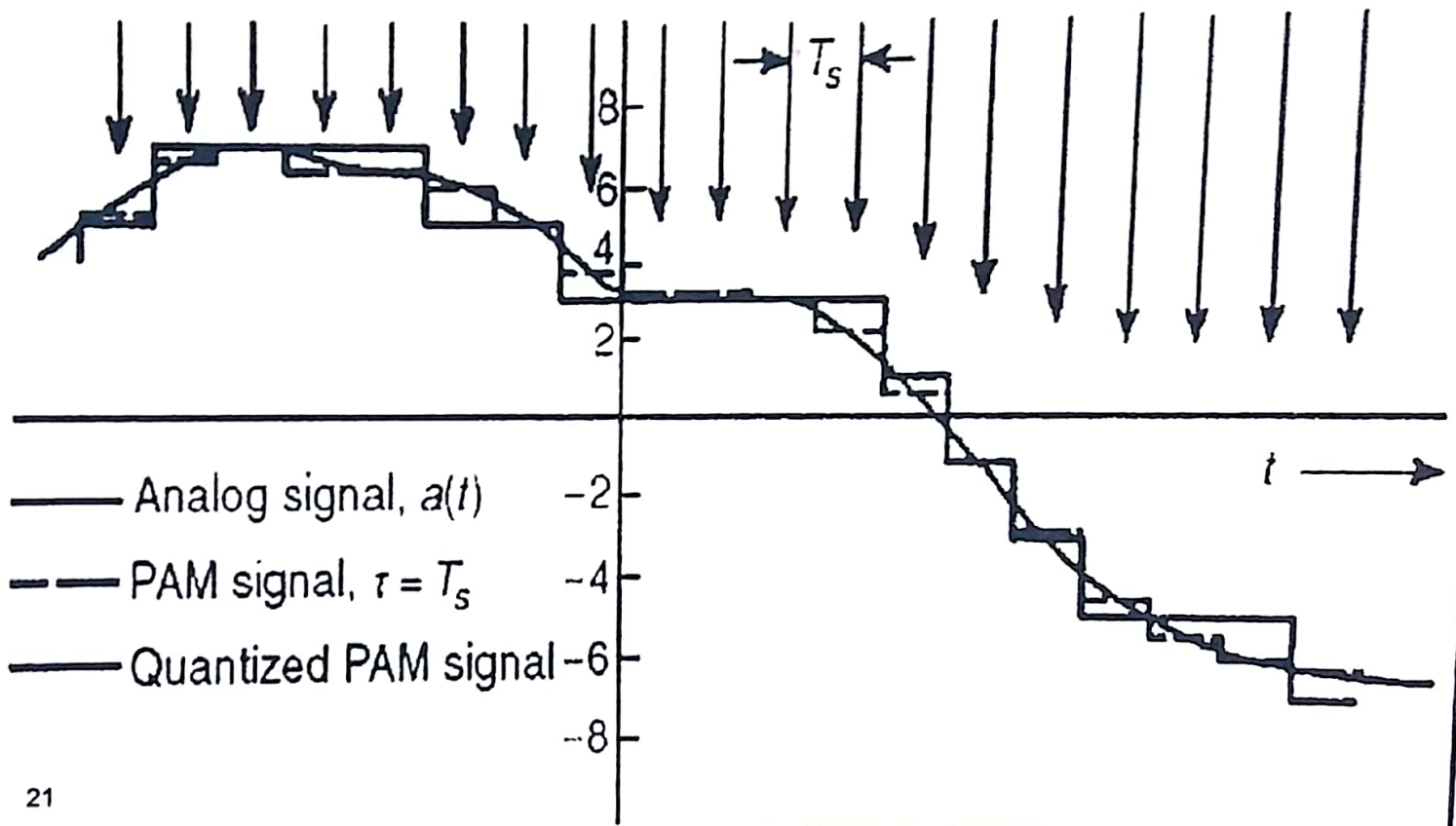
$q(t)$ is the quantization error in the
Quantization Noise

- The difference between the input and output signals of the quantizer becomes the **quantizing error** or **quantizing noise**

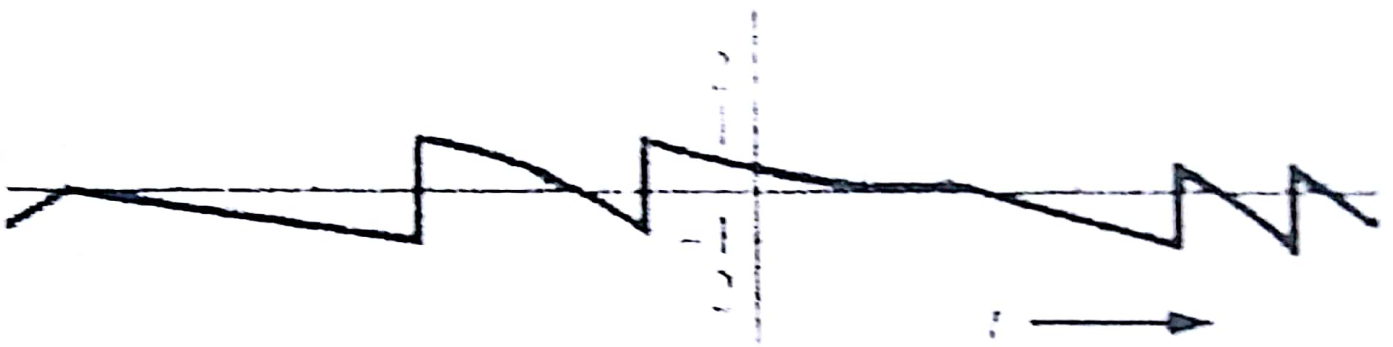
$$-\frac{\Delta}{2} \leq q(t) \leq \frac{\Delta}{2}$$



Sampling times



Quantization Error or Noise



- Assuming that the error is equally likely (uniform distributed) to lie anywhere in the range $(-\Delta/2, \Delta/2)$, the mean-square quantizing error is given by

$$\overline{q^2} = \frac{1}{\Delta} \int_{-\Delta/2}^{\Delta/2} q^2 dq = \frac{\Delta^2}{12}$$

$$\overline{q^2} = \frac{m_q^2}{3L^2}$$

$$\frac{S_o}{N_o} = 3L^2 \frac{\overline{m^2(t)}}{m_q^2}$$

SNR

Example

- For a full-scale sinusoidal modulating signal $m(t) = A \cos(\omega_m t)$, show that

$$\frac{S_o}{N_o} = \frac{3L^2}{2}$$

- or $\left(\frac{S_o}{N_o} \right)_{dB} = 1.76 + 20 \log_{10}(L) \quad (dB)$

$$\frac{S_o}{N_o} = 3L^2 \frac{S_o}{m_q^2}$$



EE325: Chapter 6 (Lec. #4)

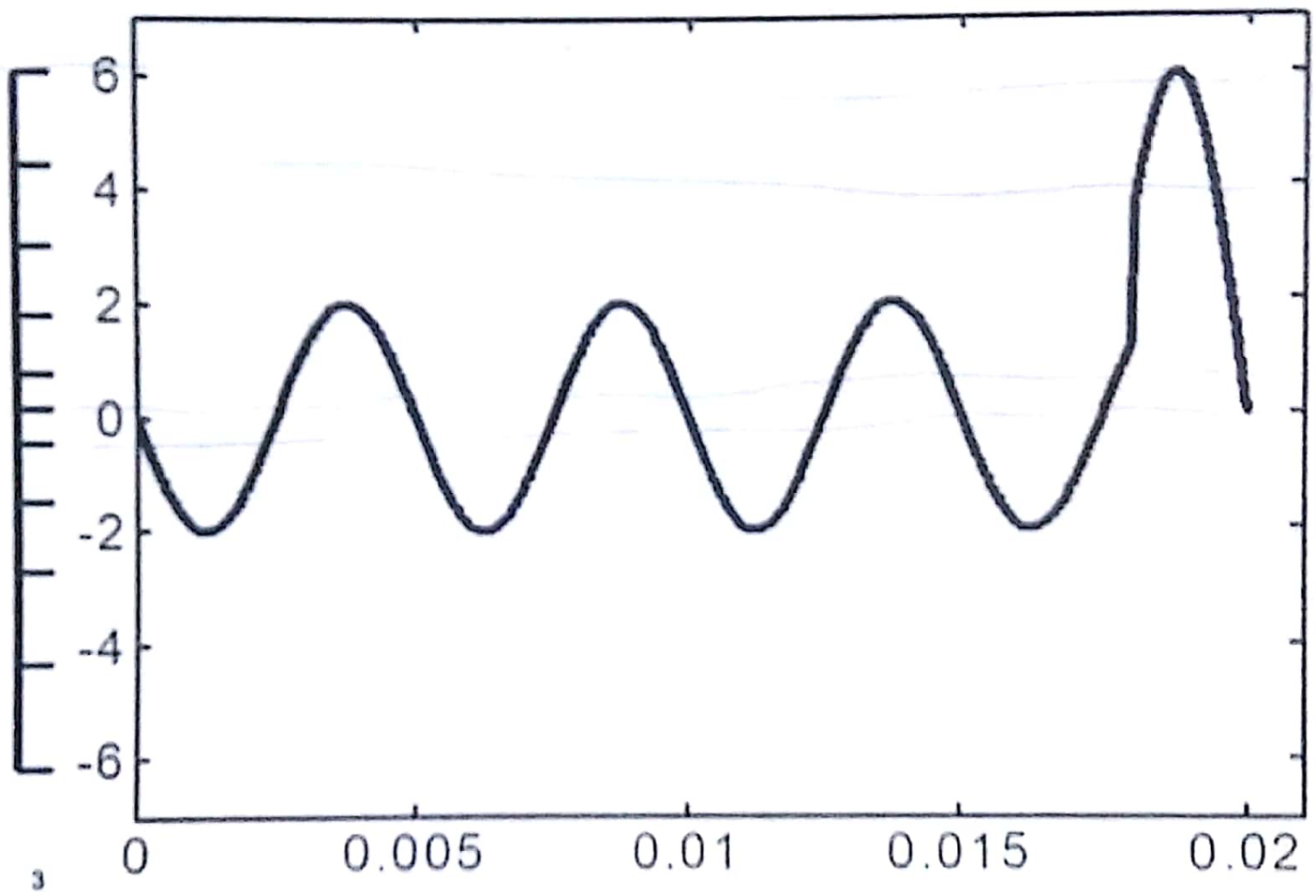
Sampling and Pulse Code Modulation

M. A. Smadi

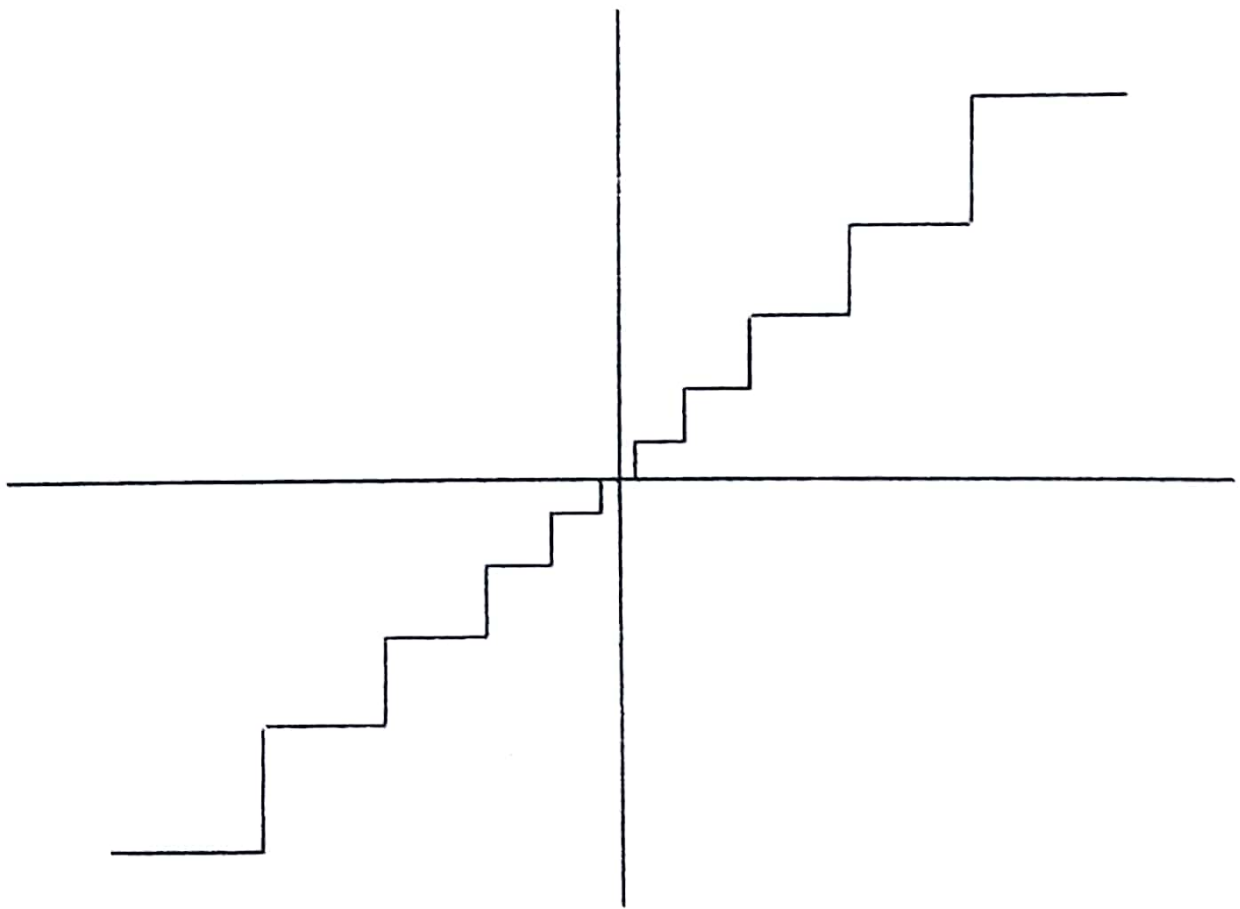
Nonuniform Quantization

- For many classes of signals the uniform quantizing is not efficient.
- Example: speech signal has large probability of small values and small probability of large ones.
- Solution: allocate more levels for small amplitudes and less for large. Thus, total quantizing noise is greatly reduced

Example of Nonuniform quantization



Nonuniform Quantization



- The effect of nonuniform quantizing can be obtained by first passing the analog signal through a compression (nonlinear) amplifier and then into the PCM circuit that uses a uniform quantizer.
- At the receiver end, demodulate uniform PCM and expand it.
- The technique is called **companding**.
- Two common techniques
 1. **μ -law companding**
 2. **A-law companding**

μ -law Compression Characteristic

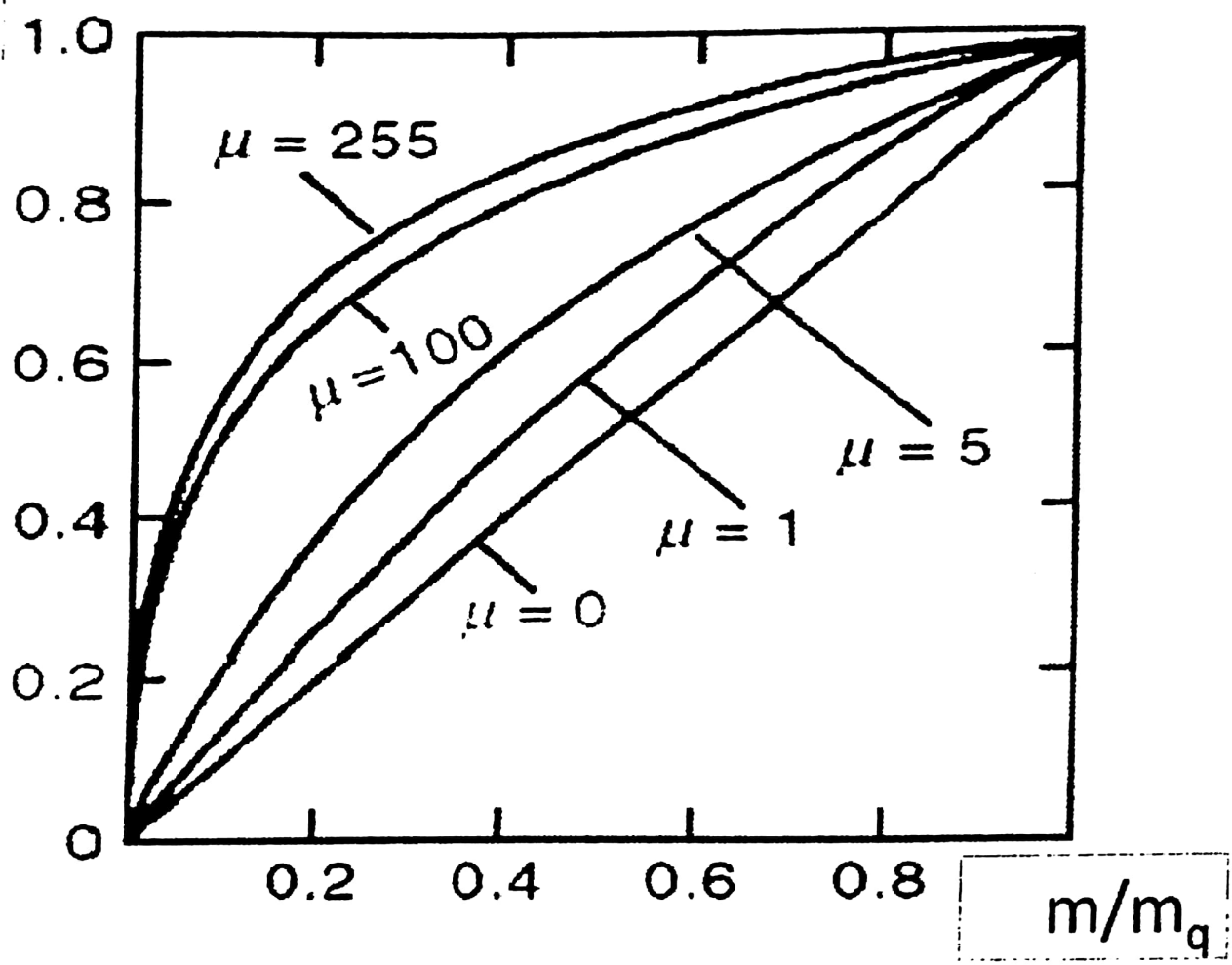
$$y = \frac{1}{\ln(1 + \mu)} \ln \left(1 + \mu \frac{m}{m_q} \right)$$

- where

$$0 \leq \frac{m}{m_q} \leq 1$$

μ -law Compression Characteristic

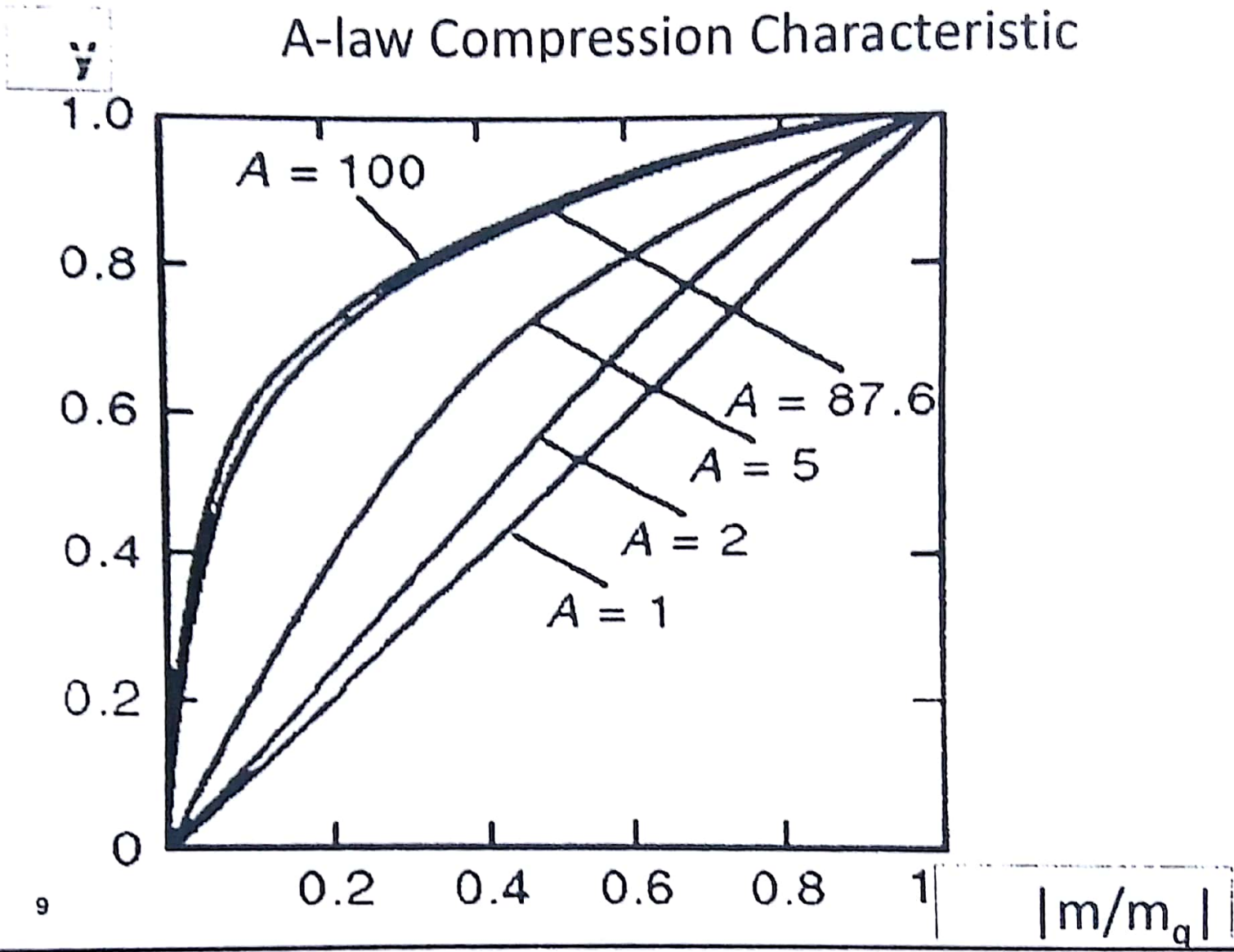
signal Y



A-law Compression Characteristic

$$y = \begin{cases} \frac{A}{1 + \ln A} \left(\frac{m}{m_q} \right), & 0 \leq \frac{m}{m_q} \leq \frac{1}{A} \\ \frac{1}{1 + \ln A} \left(1 + \ln \left[A \frac{m}{m_q} \right] \right), & \frac{1}{A} \leq \frac{m}{m_q} \leq 1 \end{cases}$$

A-law Compression Characteristic



Nonuniform Quantization

- The compressed samples must be restored to their original values at the receiver by using an expander with a characteristics complementary to that of the compressor.
- The combination of compression and expansion is called **companding**

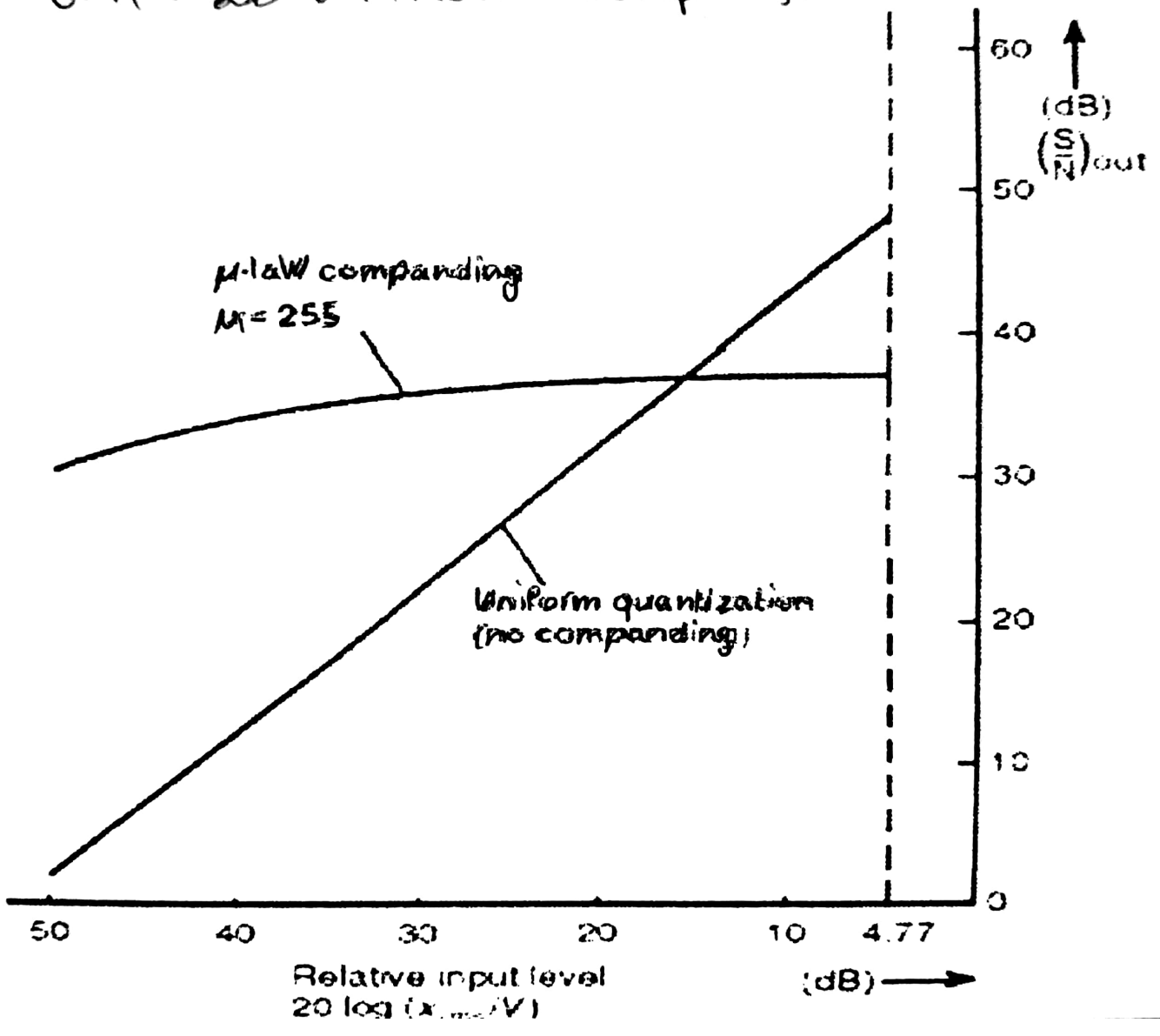
- It can be shown that when a μ -law compander is used, the output SNR is

$$\frac{S_o}{N_o} \approx \frac{3L^2}{[\ln(1 + \mu)]^2}$$

- where

$$\mu^2 \gg \frac{m_q^2}{m^2(t)}$$

with and without compression

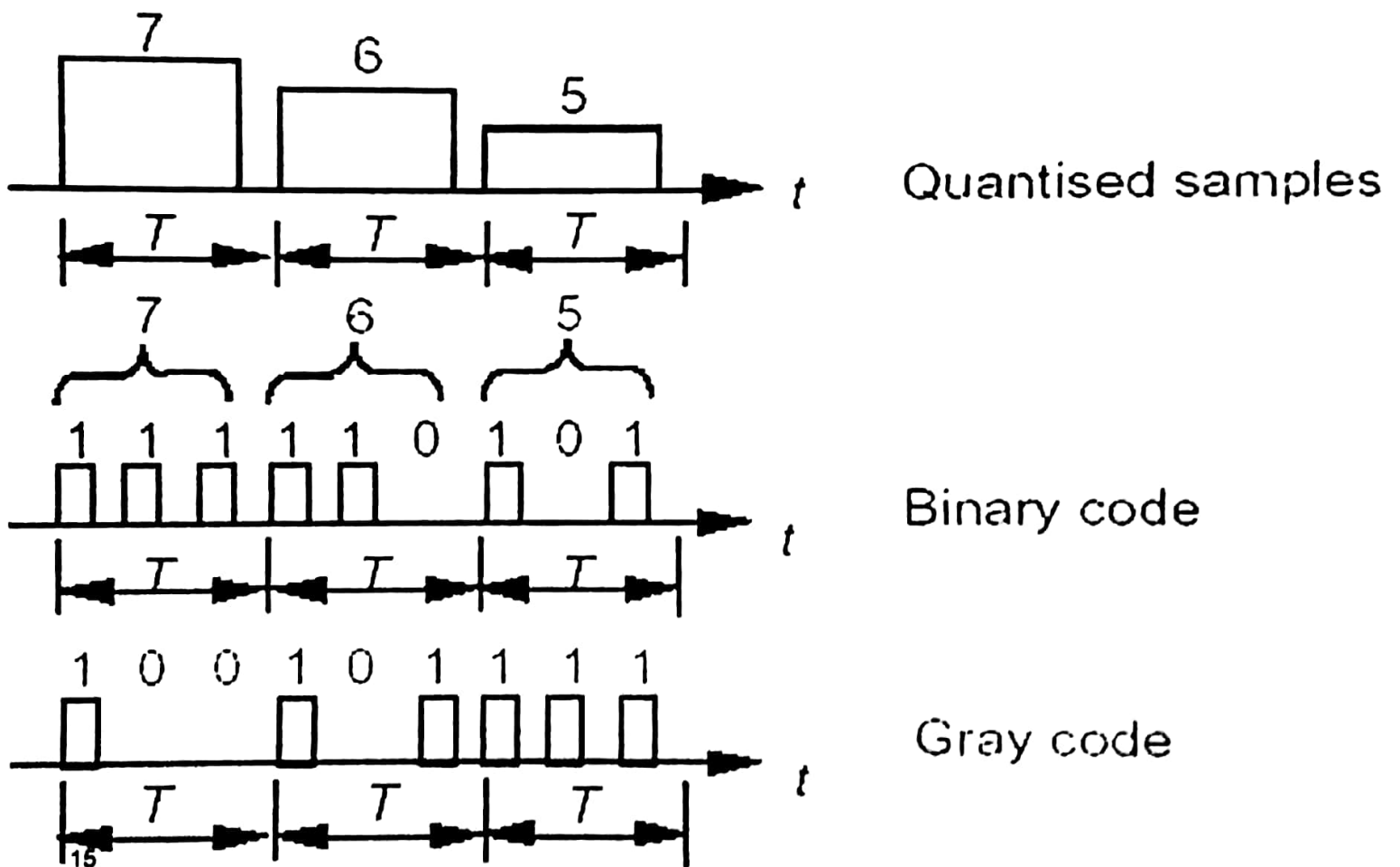


Coding of Quantized Samples

- The coding process in an A/D converter assigns a unique binary number to each quantization level. For example, we can use binary and gray coding.
- A word length of n bits can create $L = 2^n$ different binary numbers.
- The higher the number of bits, the finer the quantization and the more expensive the device becomes.

Digit	Binary Code	Gray Code
	$[b_1 \ b_2 \ b_3 \ b_4]$	$[g_1 \ g_2 \ g_3 \ g_4]$
0	0000	0000
1	0001	0001
2	0010	0011
3	0011	0010
4	0100	0110
5	0101	0111
6	0110	0101
7	0111	0100
8	1000	1100
9	1001	1101
10	1010	1111
11	1011	1110
12	1100	1010
13	1101	1011
14	1110	1001
15	1111	1000

Binary and Gray coding of samples.



Output SNR

- SNR is controlled by the PCM bandwidth

$$L = 2^n$$

$$\frac{S_o}{N_o} = cL^2 = c2^{2n}$$

$$\left(\frac{S_o}{N_o}\right)_{dB} = \alpha + 6n \text{ (dB)} \quad ; \quad \alpha = 10 \log_{10} c; \quad \log_{10}(x) = \frac{\ln(2)}{\ln(10)} \log_2(x)$$

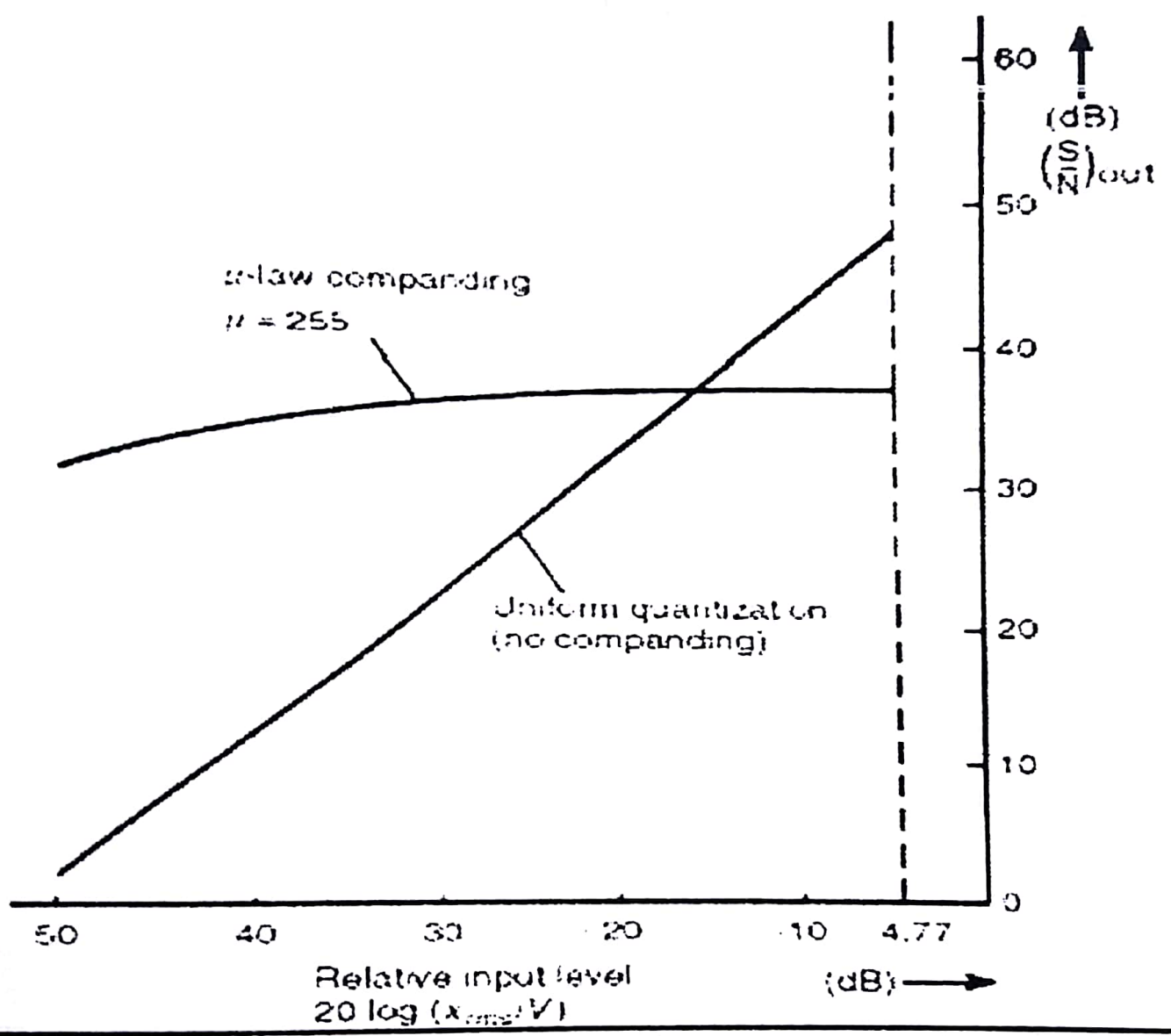
$$c = \begin{cases} \frac{3}{[\ln(1+\mu)]^2}, & \mu\text{-Law} \rightarrow \text{Compressed Case} \\ \frac{3m^2(r)}{m_g^2}, & \text{Uniform} \rightarrow \text{unCompressed Case} \end{cases}$$

Slide 12 $\leftarrow \frac{c}{L^2}$
 Slide 27 $\leftarrow \frac{c}{L^2}$

- However

$$B_T = nB$$

¹⁶ This is the theoretical minimum



Comments About dB Scale

- The decibel can be a measure of power ratio

$$\left(\frac{S}{N}\right)_{dB} = 10 \log_{10} \left(\frac{S}{N}\right)$$

- It can also be used for measuring power

$$P_{dBW} = 10 \log_{10} P_W$$
$$P_{dBm} = 10 \log_{10} \frac{P_W}{1 \text{ mW}}$$

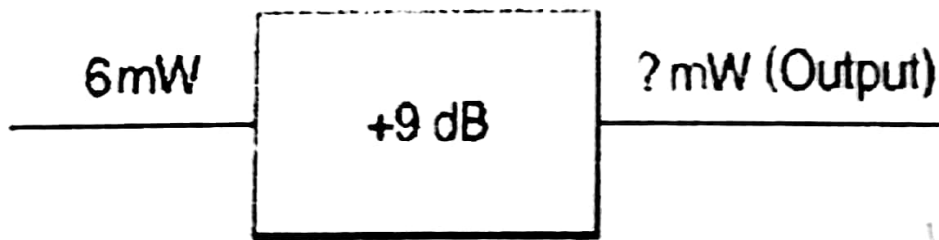
Examples

• Gain = $(P_{out}/P_{in}) = 2 = +3 \text{ dB}$



Handwritten notes showing the calculation of power gain in dB:

$$10 \log_{10} 2 = 3$$
$$P_{dB} = 10 \log_{10} P$$



$P_{out} = 48 \text{ mW}$

Handwritten equation: $9 = 10 \log_{10} \frac{x}{6}$



EE325: Chapter 6 (Lec. #5)

Sampling and Pulse Code Modulation

M. A. Smadi

Bandwidth of PCM

- What is the spectrum of a PCM signal?
- The spectrum of the PCM signal depends on the bit rate, the correlation of the PCM data, and on the PCM waveform pulse shape (usually rectangular) used to describe the bits.
- The dimensionality theorem [2] shows that the bandwidth of the PCM waveform is bounded by

$$B_{\text{PCM}} \geq R/2 = n f_s/2$$

where R: bit rate

Transmission Bandwidth

- For L quantization levels and n bits

$$L = 2^n \text{ or } n = \log_2 L$$

- The bandwidth of the PCM waveform

$$B_{\text{PCM}} \geq n B \text{ Hz}$$

بالتحديد
التردد
الذي
يحتوي
البيانات

- Minimum channel bandwidth or transmission bandwidth

$$B_T = n B \text{ Hz}$$

Example: PCM for Telephone System

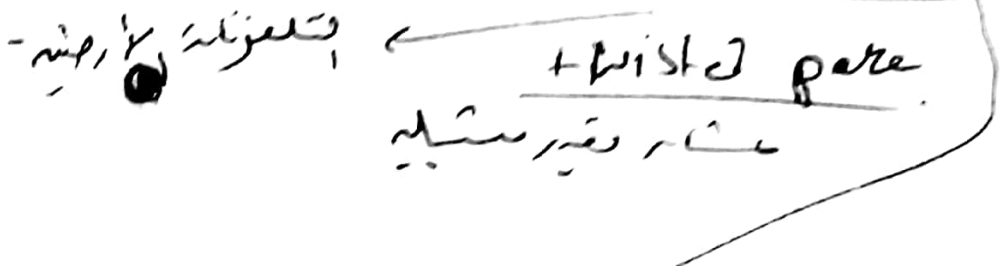
المثال الثاني

- Telephone spectrum: [300 Hz, 3400 Hz]
- Min. sampling frequency: $f_{s,min} = 2 f_{max} = 6.8 \text{ kHz}$
or value
- Some guard band is required:

$$f_s = 2 F_{max} + \Delta f_g = 8 \text{ kHz}$$

سبب الحاجة
 الى حيز
 800
 هرتز
 كحيز
 حرا

- n=8-bit codewords are used $\rightarrow L=256$
power of 2
- The transmission rate: $R = n * f_s = 64 \text{ kbits/s}$
- Minimum PCM bandwidth: $B_{PCM} = R/2 = 32 \text{ kHz}$



Example 6.3

نوع 6.3

- A signal $m(t)$ of bandwidth $B = 4$ kHz is transmitted using a binary companded PCM with $\mu = 100$. *non-linear*
Compare the case of $L = 64$ ($n = 6$) with the case of $L = 256$ ($n = 8$) from the point of view of transmission bandwidth and the output SNR.

$$\frac{S_o}{N_o} \approx \frac{3L^2}{[\ln(1 + \mu)]^2}$$

$$B_T = n B \text{ Hz}$$

أولاً

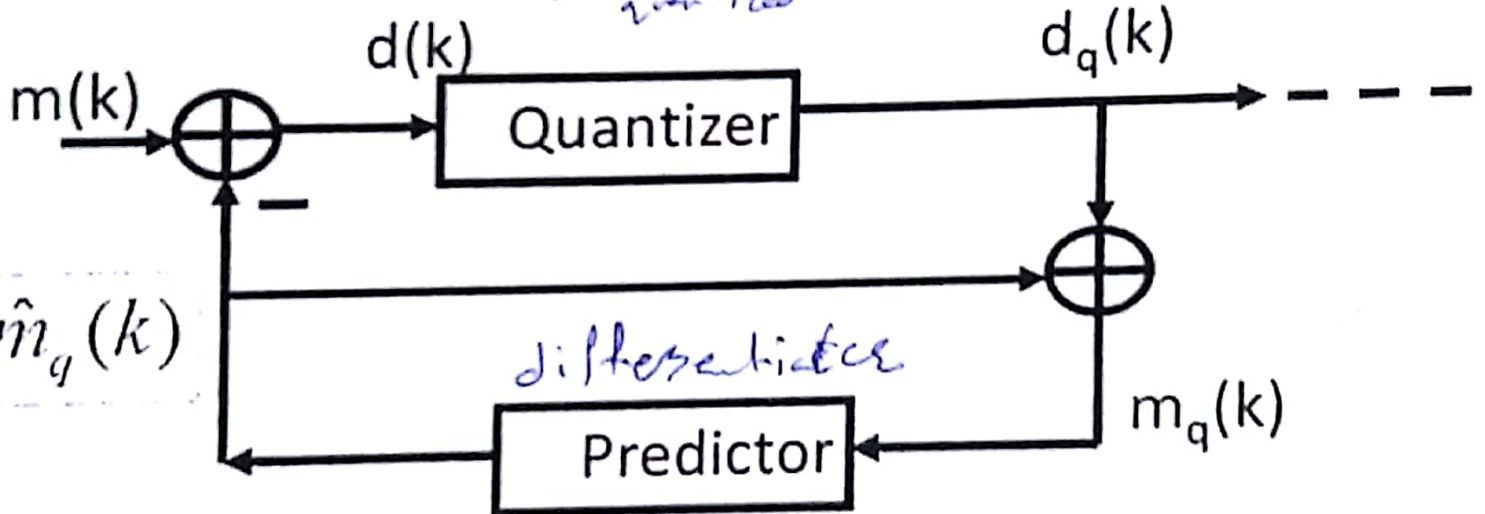
Differential PCM (DPCM)

- Samples of a band-limited signal are correlated.
- This can be used to improve PCM performance: to decrease the number of bits used (and, hence, the bandwidth) or to increase the quantization SNR for a given bandwidth.
- Main idea: quantize and transmit the ^{الفرق} difference between two adjacent samples rather than sample values.
- Since two adjacent samples are correlated, their difference is small and requires less bits to transmit.

DPCM System-Modulator

$$d(k) = m(k) - \hat{m}_q(k)$$

$$d_q(k) = d(k) + q(k)$$



$$m_q(k) = \hat{m}_q(k) + d_q(k)$$

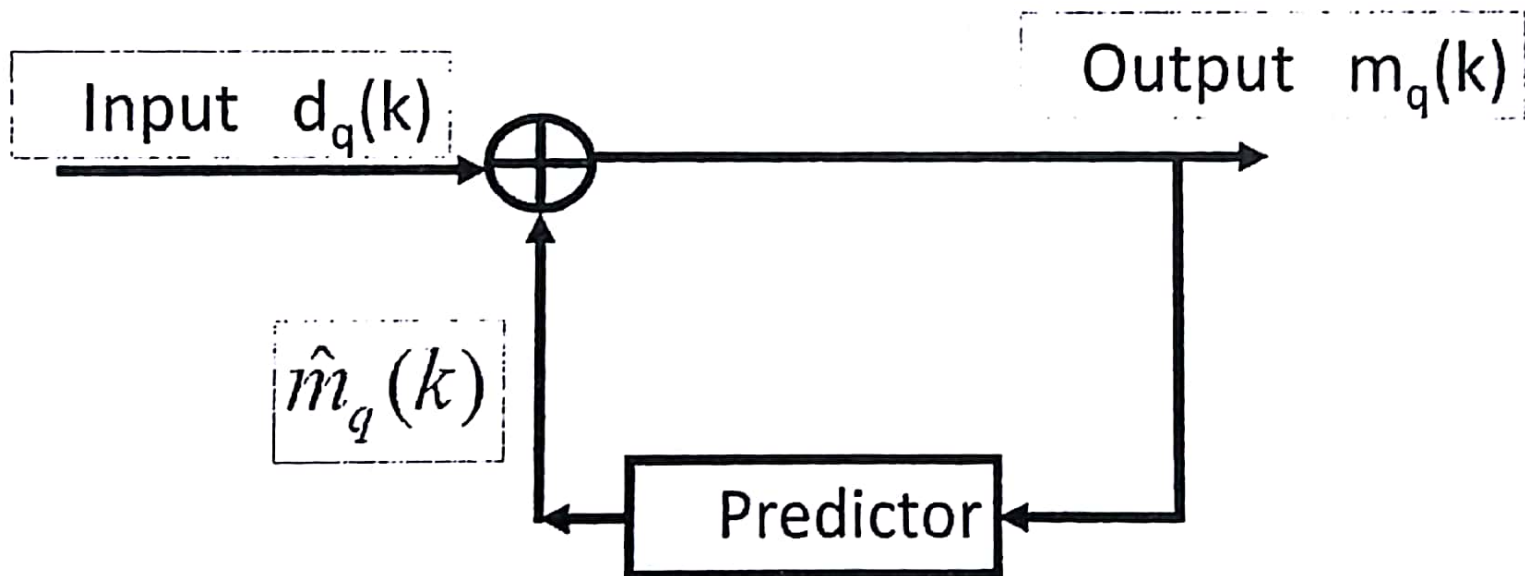
$$= m[k] + q[k]$$

$q(k)$: is the quantization error

DPCM System-Demodulator

$$d_q(k) = m_q(k) - \hat{m}_q(k)$$

$$m_q(k) = d_q(k) + \hat{m}_q(k)$$



or

Predictor

- From Taylor series

result

$$m(t + T_s) \approx m(t) + T_s m'(t), \quad T_s \ll$$

- Or \downarrow exact $\frac{1}{2} \Delta t$ *goes to zero*

T_s

$$m[k+1] \approx m[k] + T_s \left[\frac{m[k] - m[k-1]}{T_s} \right]$$

current sample

$$= 2m[k] - m[k-1]$$

etc. etc. main

صوت او

SNR improvement in DPCM

- Let m_q, d_q : Peak of $m(t)$ and $d(t)$

step size

- If we use the same L , $\frac{\Delta_{DPCM}}{\Delta_{PCM}} = \frac{d_q}{m_q} < 1$

$$L = \frac{2m}{\Delta}$$

$$L = \frac{2d}{\Delta}$$

- Or, quantization noise reduced by $\left(\frac{m_q}{d_q}\right)^2$

- Hence, SNR improvement will be

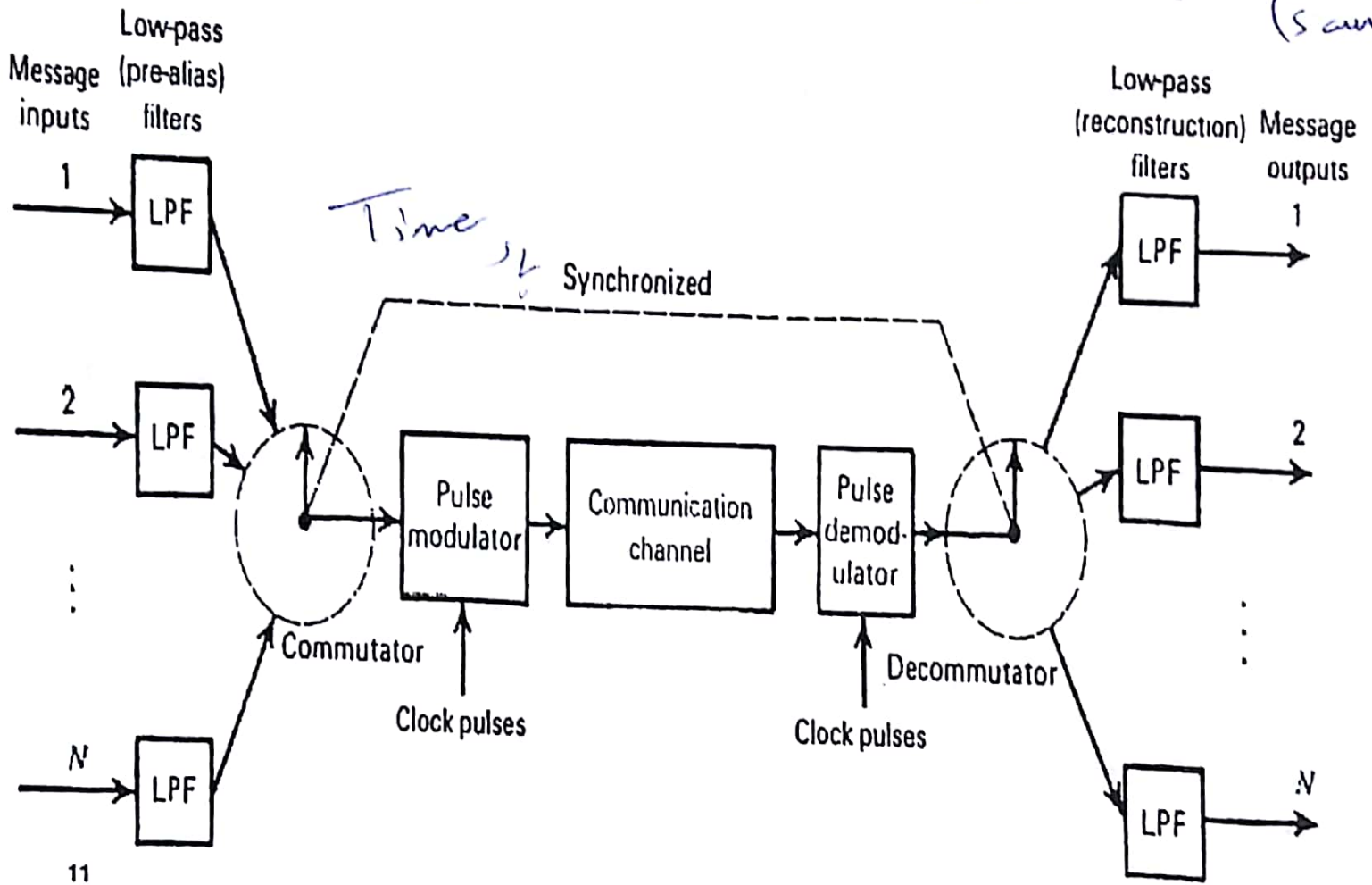
$$G_p = \frac{P_m}{P_d}$$

Power of $m(t)$ over $d(t)$

(SNR improvement due to prediction)

Time Division Multiplexing (TDM)

(5 amp)

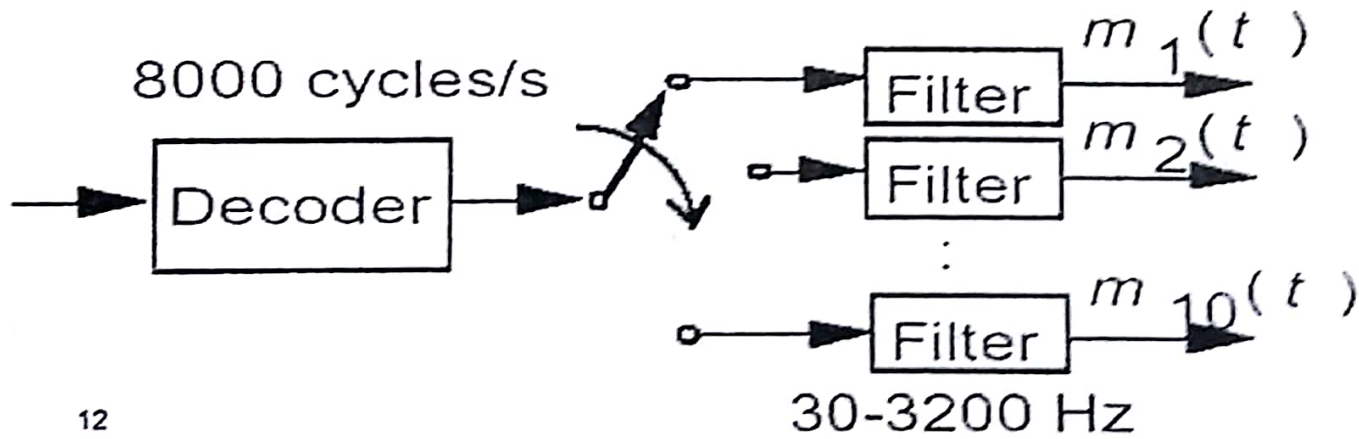
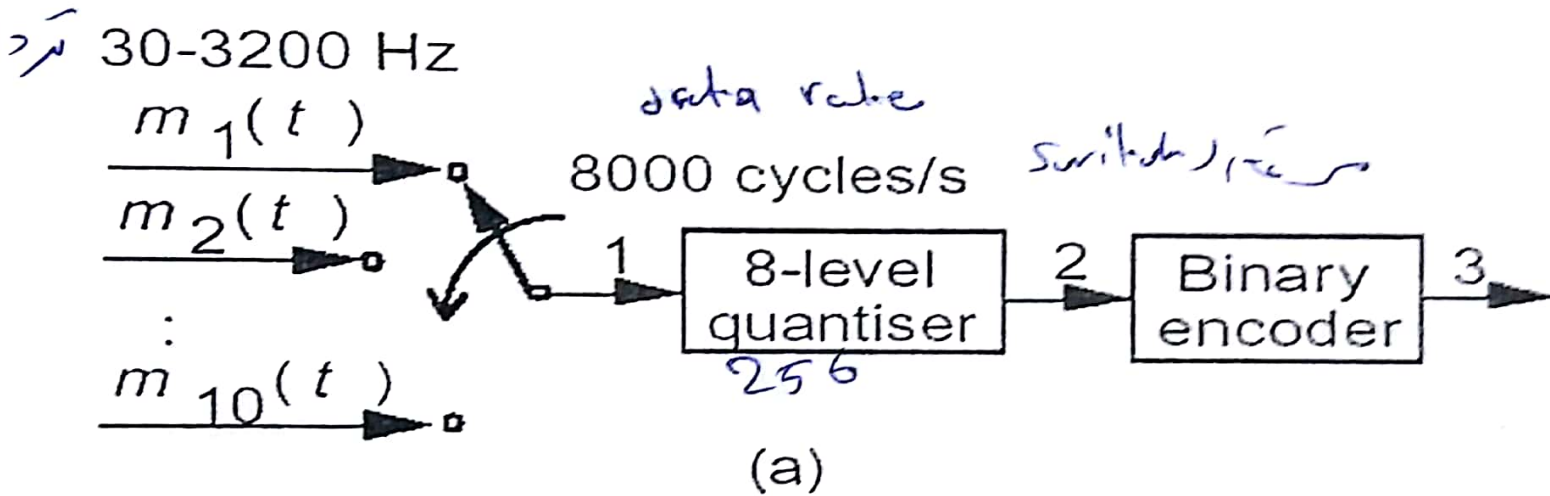


فقال على صوت
الاشارة

Ten-Channel PCM System

(a) Transmitter (b) Receiver

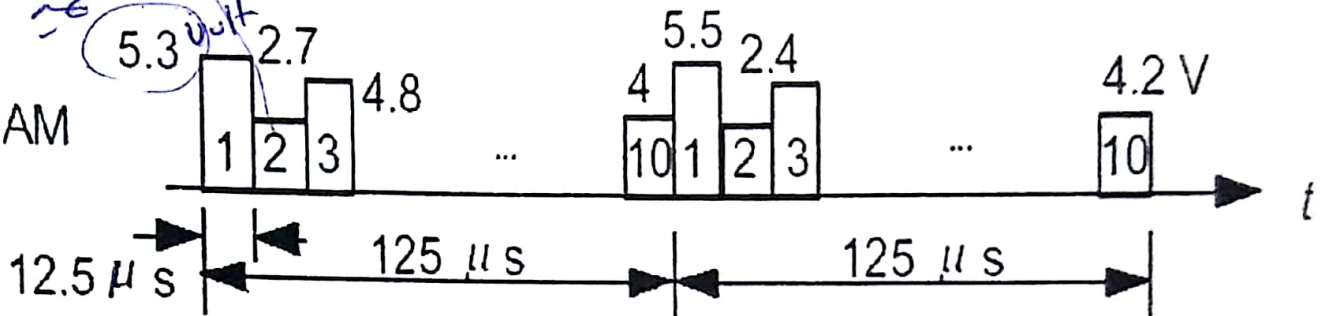
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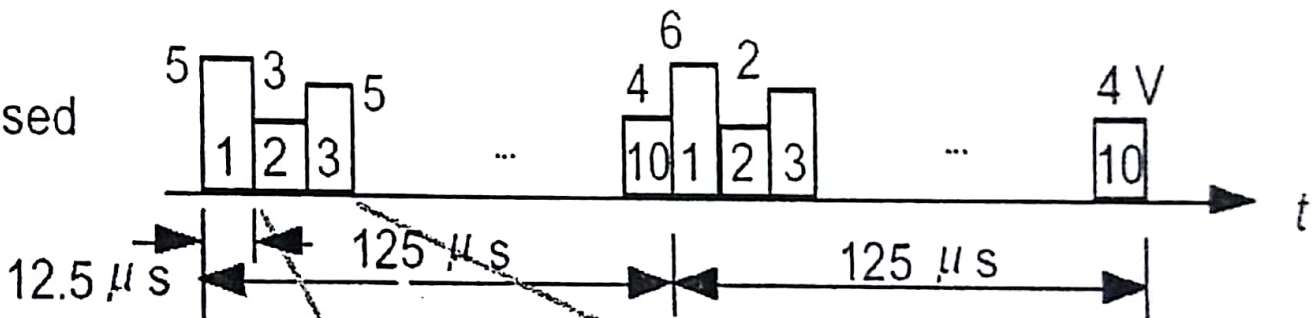
Signal Shapes

low pass
sample rate

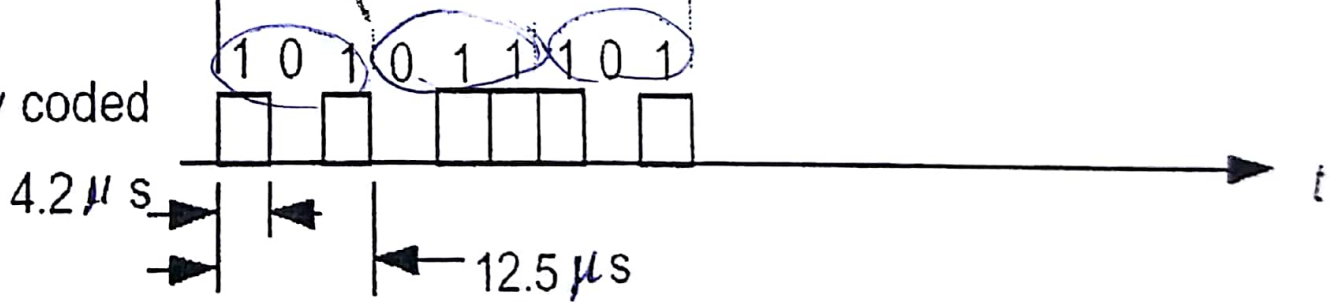
1. TDM PAM



2. Quantised



3. Binary coded



Bandwidth Requirements for TDM

- If N band-limited signals are multiplexed each with bandwidth B
- The minimum TDM sampling rate is

$$f_{\text{TDM}} = 2 N B$$

- If each sample is coded with n bits, then the minimum transmitted data rate is

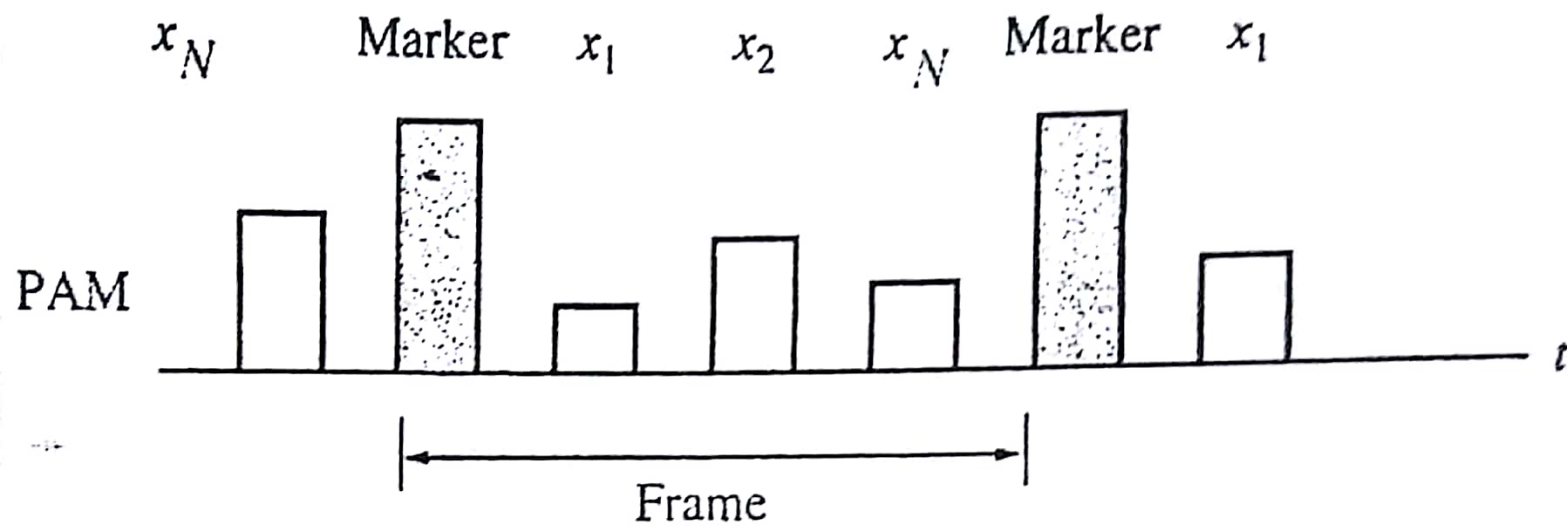
$$R = 2 n N B$$

- The minimum transmission bandwidth is

$$B_T = n N B$$

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TDM: Concept of Framing and Synchronization



- The time interval T_F containing *one sample from each message signal* is called a **frame**.
- an extra pulse (called **marker**) is transmitted for synchronization

Comparison of Time and Frequency Division Multiplexing

- ***Time division multiplexing:*** Individual TDM channels are assigned to *distinct time slots* but jumbled together in the frequency domain. Channels are separated in the *time domain*
- ***Frequency division multiplexing:*** Individual FDM channels are assigned to *distinct frequency regions* but jumbled together in the time domain. Channels are separated in the *frequency domain*

Comparison of Time and Frequency Division Multiplexing

- Many of the TDM advantages are technology driven. The digital circuits are much cheaper and easier to implement
- In FDM, imperfect bandpass filtering and nonlinear cross-modulation cause cross talk. TDM is not sensitive to these problems.

Example *bit rate*

- A binary channel with bit rate $R_b = 36000$ bits/s is available for PCM transmission. Find appropriate values of the sampling rate f_s , the quantizing level, and the binary digits n , assuming the signal bandwidth is $B = 3.2$ kHz.

○ $f_s \geq 2B = 6.4$ KHz

○ $R_b \geq n f_s$, $n = 5.6$, then we use $n = 5$, $L = 32$

○ $f_s = R_b / 5 = 7.2$ KHz

6 بیتی کوانٹرز

*integer
کوانٹرز*

نق

n

ر

SNR



EE325: Chapter 6 (Lec. #6)

Sampling and Pulse Code Modulation

M. A. Smad

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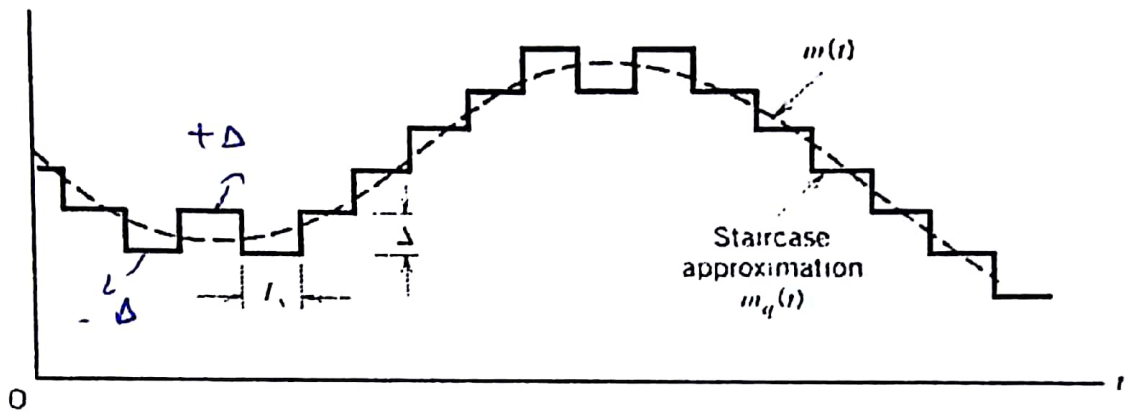
Delta Modulation (DM)

$\rightarrow n=1$
 $+ \Delta$
 $- \Delta$

- **Definition:** Delta Modulation is a technique which provides a staircase approximation to an over-sampled version of the message signal (analog input).
- Sampling is at a rate higher than the Nyquist rate – aims at increasing the correlation between adjacent samples; simplifies quantizing of the encoded signal

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Illustration of the DM process



(a)

Binary
sequence
at modulator
output

0 0 1 0 1 1 1 1 1 0 1 0 0 0 0 0 0

(b)

Principle Operation

- message signal is over-sampled Sampling rate $> \underline{\underline{2B}}$
- difference between the input and the approximation is quantized in two levels $\pm\Delta$
- these levels correspond to positive/negative differences
- provided signal does not change **very rapidly** the approximation remains within $\pm\Delta$



Assumptions and model

We assume that:

- $m(t)$ denotes the input message signal
- $m_q(t)$ denotes the staircase approximation
- $m[n] = m(nT_s)$, $n = +/-1, +/-2 \dots$ denotes a sample of the signal $m(t)$ at time $t=nT_s$, where T_s is the sampling period
- then

Cont.

- we can express the basic principles of the delta modulation in a mathematical form as follow:

- $e[n] = m[n] - m_q[n - 1]$: error signal

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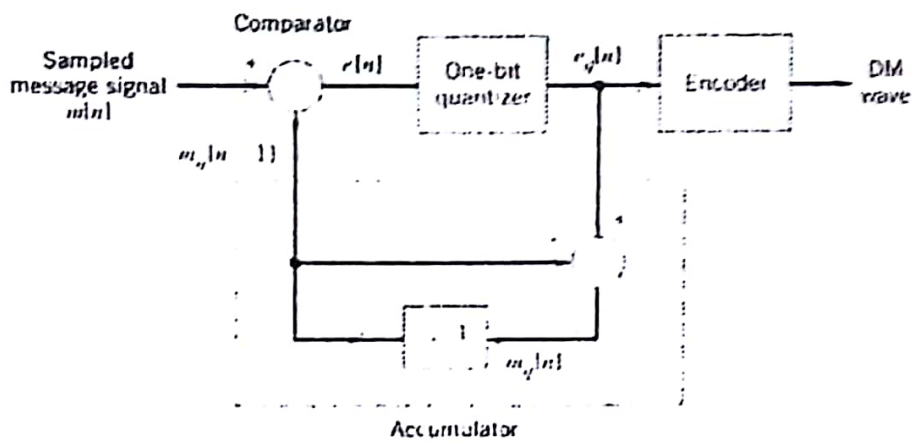
- $e_q[n] = \Delta \text{sgn}(e[n])$: quantized error signal
بعد ما اخذت الفرق تقريبي اعتمدت على الفرق ليعرف ان Δ او $-\Delta$

- $m_q[n] = m_q[n - 1] + e_q[n]$: quantized output

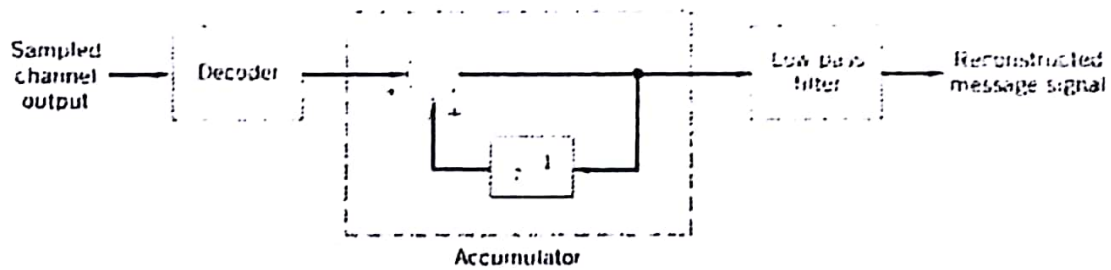
Pros and cons

- Main advantage – simplicity
 - Sampled version of the message is applied to a modulator (comparator, quantizer, accumulator)
 - delay in accumulator is “unit delay” = one sample period (z^{-1})
-

DM Block Diagram



(a)



(b)

Transmitter Side

- comparator – computes difference between input signal and one interval delayed version of it
- quantizer – includes a hard-limiter with an input-output relation a scaled version of the signum function
- accumulator – produces the approximation $m_q[n]$ (final result) at each step by adding either $+\Delta$ or $-\Delta$
- = tracking input samples by one step at a time

$$m_q[n] = \Delta \sum_{i=1}^n \text{sgn}(e[i])$$
$$= \sum_{i=1}^n e_q[i]$$

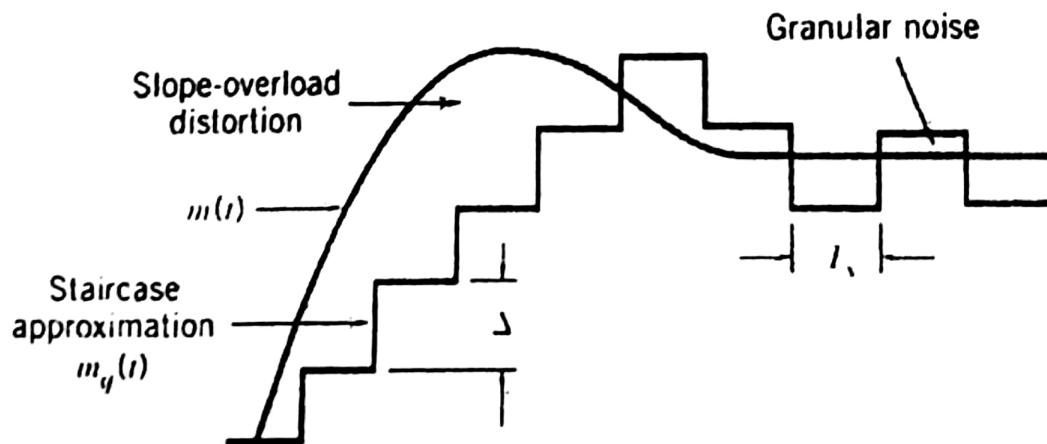
Handwritten note: $m_q[n]$ is the sum of $e_q[i]$

Receiver Side

- decoder – creates the sequence of positive or negative pulses
- accumulator – creates the staircase approximation $m_q[n]$ similar to Tx side
- out-of-band noise is cut off by low-pass filter (bandwidth equal to original message bandwidth)

Granular Noise

- In contrast to slope overhead
- Occurs when step size is too large
- Usually relatively flat segment of the signal
- Analogous to quantization noise in PCM systems



Slope Overhead Distortion

- The quantized message signal can be represented as:

$$m_q[n] = m[n] + q[n]$$
$$m_q[n-1] + e_q[n] = m[n] + q[n]$$

- where the input to the quantizer can be represented as:

$$e[n] = m[n] - m[n-1] - q[n-1]$$

So, (except for the quantization error) the **quantizer input** is the first backward difference (**derivative**) of the input **signal** = inverse of the digital integration process

Discussion

- Consider the max slope of the input signal $m(t)$
- To increase the samples $\{m_q[n]\}$ as fast as the input signal in its *max slope region* the following condition should be fulfilled:

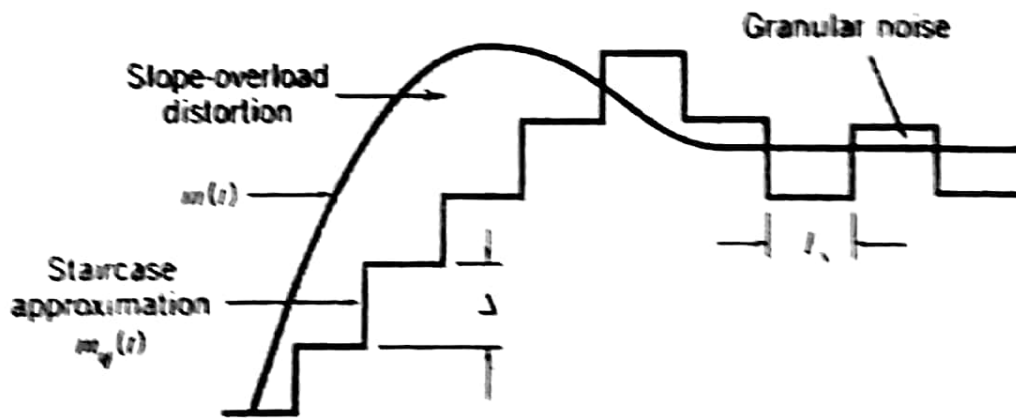
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$$\frac{\Delta}{T_s} \geq \max \left| \frac{dm(t)}{dt} \right| \quad (3.58)$$

otherwise the step-size Δ is too small

Granular Noise

- In contrast to slope overhead
- Occurs when step size is too large
- Usually relatively flat segment of the signal
- Analogous to quantization noise in PCM systems



Conclusion

- Large step-size is necessary to accommodate a wide dynamic range
- Small step-size is required for accuracy with low-level signals
- = compromise between slope overhead and granular noise
- = adaptive delta modulation, where the step size is made to vary with the input signal

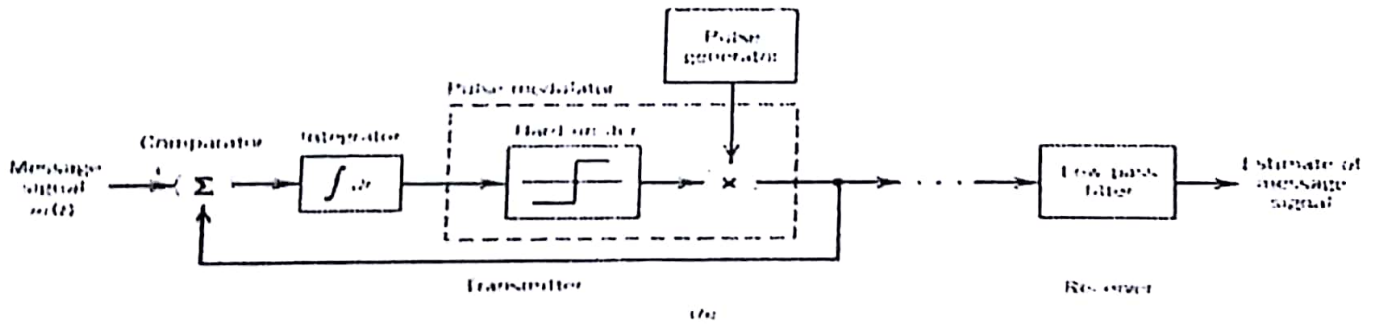
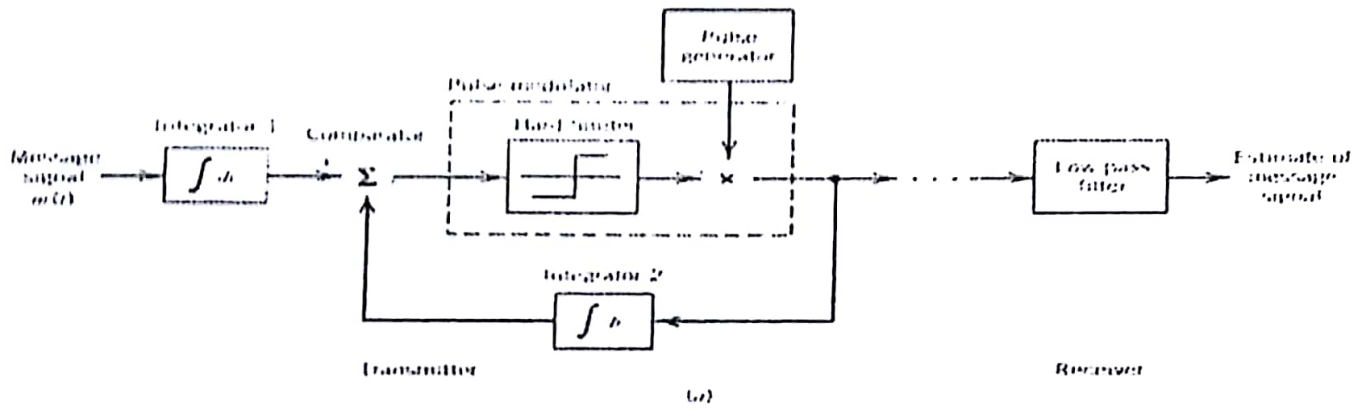
Delta Sigma Modulation (DSM)

- *Conventional* delta modulation - Quantizer input is an *approximation* of the **derivative** of the input message signal $m(t)$.
- Results in the accumulation of error (noise)
- Possible solution: integrating the message before delta modulation – called delta sigma modulation

Remark

- The message signal is defined in its continuous form – so pulse modulator contains a hard limiter and a pulse generator to produce a 1-bit encoded signal
- Integration at the Tx requires *differentiation* at the Rx side.
- **But:** As in conventional DM the message has to be *integrated* at the final stage this eliminates the need of differentiation here.

Block diagrams DSM



Pros and cons for DSM

- Simplicity of implementation both at the Tx and Rx side
- Requires sampling rate far in excess of the Nyquist rate (PCM) – increase in transmission and channel bandwidth
- If bandwidth is at a premium we have to choose increased system complexity (additional signal processing) to achieve reduced bandwidth.



EE 325: Chapter 7 (Lec.#1)

Principals of Digital Data Transmission

M. A. Smadi

Content

- Digital Communication Systems
- Line Coding
- Pulse Shaping
- M-ary Communication
- Digital Carrier Systems
- Digital Multiplexing

Principals of Digital Data Transmission

This chapter deals with the problems of transmitting digital data over a channel. Hence, the starting messages are assumed to be digital. To begin with we shall consider the binary case, where the data consists of only two symbols: 1 and 0. We assign a distinct waveform (pulse) to each of these two symbols. The resulting sequence of these pulses is transmitted over a channel. At the receiver, these pulses are detected and are converted back to (1's and 0's).

Principals of Digital Data Transmission

Source

- The input to a digital system is in the form of a sequence of digits. The input could be the output from such sources as a data set, a computer, a digitized voice signal (PCM or DM), a digital facsimile or television, or telemetry equipment. Most of the discussion in this is restricted to the binary case (communication schemes using only two symbols).
- General case of M-ary communication which uses M symbols is discussed later.

Principals of Digital Data Transmission

Multiplexer

- Generally speaking, the capacity of a practical channel transmitting data is much larger than the data rate of individual sources. To utilize this capacity effectively, we combine several sources through a digital multiplexer using the process of interleaving.
- Thus a channel is time shared by several messages simultaneously.

Principals of Digital Data Transmission

Line Coder

- The output of a multiplexer is coded into electrical pulses or waveforms for the purpose of transmission over the channel. This process is called line coding.
- There are many possible ways of assigning waveforms (pulses) to the digital data. In the next paragraph we consider the binary case (two symbols)

Principals of Digital Data Transmission

RZ and NRZ signals

- When the pulse duration is half the bit duration ($T_B / 2$) the resulting signal is called RZ (return to zero)
- When the pulse duration is equal to the bit duration (T_B) the resulting signal is called NRZ (non return to zero)

Principals of Digital Data Transmission

Regenerative repeater

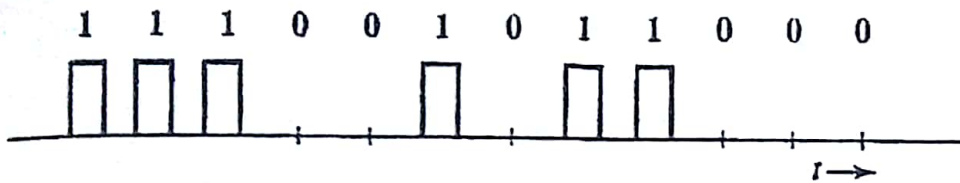
- Regenerative repeaters are used at regularly spaced intervals along a digital transmission line to detect the incoming digital signal and regenerate new clean pulses for further transmission along the line.
- This process periodically eliminates, and thereby combats, the accumulation of noise and signal distortion along the transmission path.

Line Coding

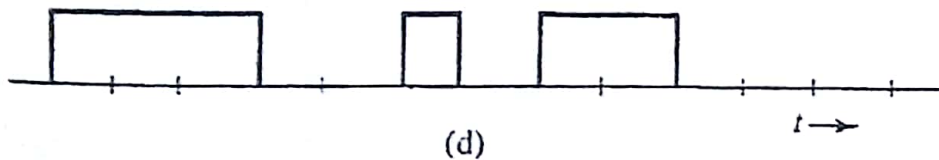
On-Off Line Coding

Bit = 1 → is transmitted by a pulse $p(t)$

Bit = 0 → is transmitted by no pulse (zero signal)



RZ on-off



NRZ on-off

(d)

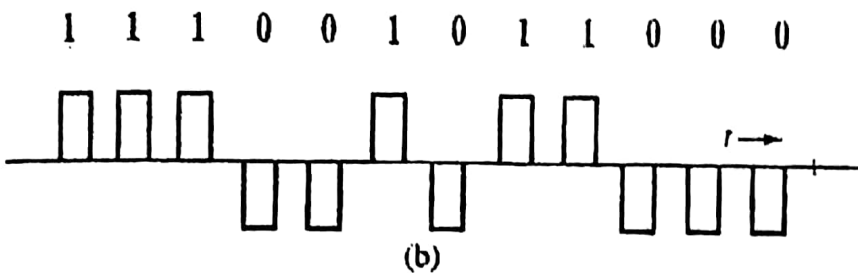
Line Coding

Polar Line Coding

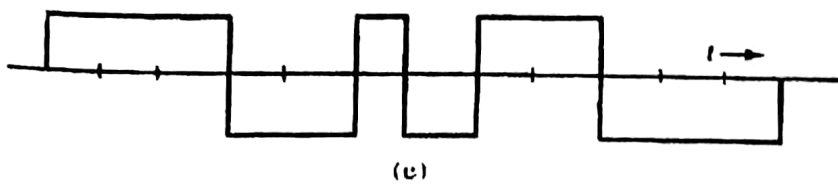
Bit = 1 \rightarrow is transmitted by a pulse $p(t)$

Bit = 0 \rightarrow is transmitted by $-p(t)$

Advantages: Power efficiency



RZ polar



NRZ polar

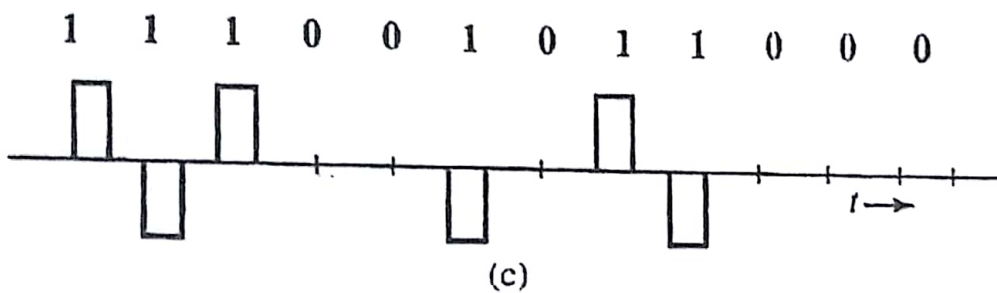
Line Coding

Bipolar Line Coding (pseudoternary or also alternate mark inversion AMI)

Bit = 0 \rightarrow is transmitted by no pulse

Bit = 1 \rightarrow is transmitted by $p(t)$ or $-p(t)$ depending whether the previous 1 is transmitted by $-p(t)$ or $p(t)$

Advantages: Error detection capability



RZ bipolar

11

Line Coding

Digital data can be transmitted by various transmission or line codes, such as on-off, polar, bipolar, and so on. Each has its advantages and disadvantages. Among other desirable properties, a line code should have the following properties:

1. Transmission bandwidth: It should be as small as possible.
2. Power efficiency: For a given bandwidth and a specified detection error probability, the transmitted power should be as small as possible.

Line Coding

3. Error detection and correction capability: It should be possible to detect, and preferably correct, detection errors. In a bipolar case, for example, a single error will cause bipolar violation and can easily be detected.

4. Favorable power spectral density: It is desirable to have zero PSD at $\omega = 0$ (dc), because ac coupling and transformers are used at the repeaters. The ac coupling is required because the dc paths provided by the cable pairs between the repeater sites are used to transmit the power required to operate the repeaters.

Line Coding

5. Adequate timing content: It should be possible to extract timing or clock information from the signal.
6. Transparency: It should be possible to transmit a digital signal correctly regardless of the pattern of 1's and 0's. A long string of 0's could cause errors in timing extraction in on-off and bipolar cases. If the data are so coded that for every possible sequence of data the coded signal is received faithfully, the code is transparent.

Line Coding

PSD of Various Line Codes

Procedure to find a general expression for PSD of a large class of line codes are as follows:

Consider the pulse train in Fig. 7.3b constructed from a basic pulse $p(t)$ (Fig. 7.3a) repeating at intervals of T_b with relative strength a_k for the pulse starting at $t = kT_b$. In other words, the k^{th} pulse in this pulse train $y(t)$ is $a_k p(t)$. The values a_k are arbitrary and random. This is a PAM signal. The on-off, polar, and bipolar line codes are all special case of this pulse train $y(t)$, where a_k takes on values 0, 1, or -1 randomly subject to some constraints. We therefore, analyze many line codes from the knowledge of the PSD of $y(t)$.

Line Coding

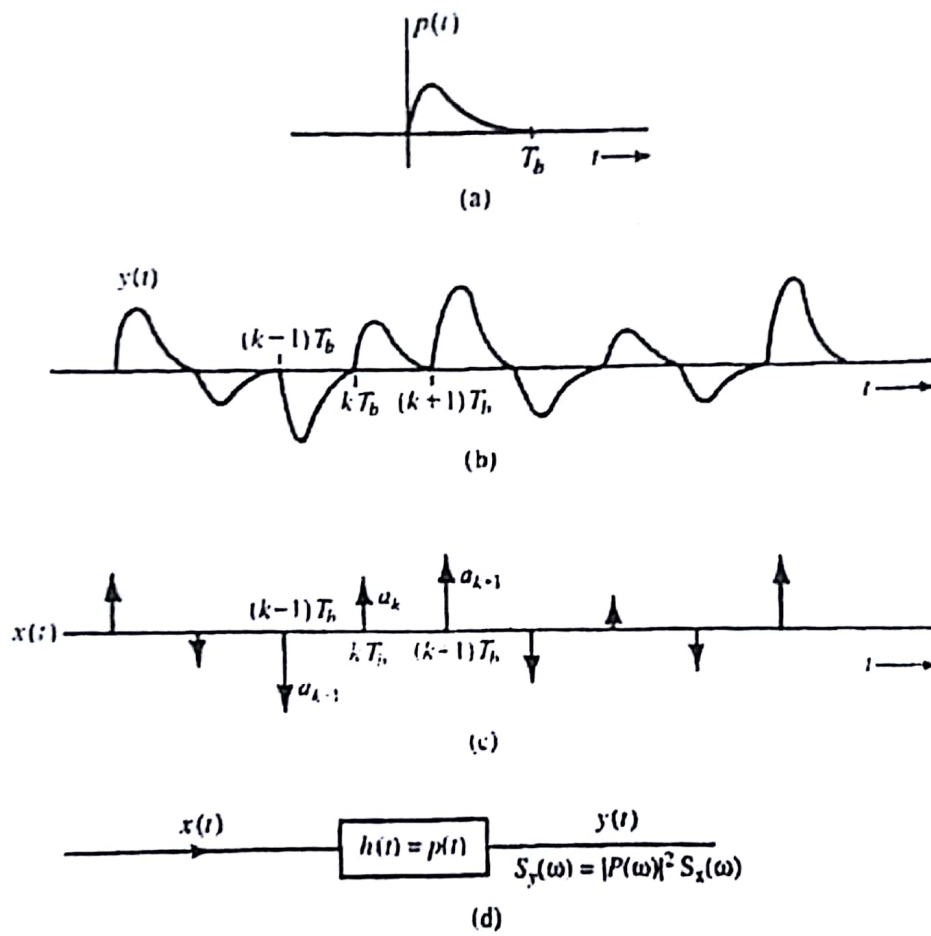


Figure 7.3 A random PAM signal and its generation from a PAM impulse sequence.

Line Coding

The input $x(t)$ to the filter with impulse response $h(t) = p(t)$ results in the output $y(t)$, as shown in Fig. 7.3d. If $p(t) \iff P(\omega)$, the transfer function of the filter is $H(\omega) = P(\omega)$, and according to Eq. (3.90),

$$\mathcal{R}_x(\tau) = \frac{1}{T_b} \sum_{n=-\infty}^{\infty} R_n \delta(\tau - nT_b) \quad (7.7)$$

The PSD $S_x(\omega)$ is the Fourier transform of $\mathcal{R}_x(\tau)$. Therefore,

$$S_x(\omega) = \frac{1}{T_b} \sum_{n=-\infty}^{\infty} R_n e^{-jn\omega T_b} \quad (7.8)$$

Recognizing the fact that $R_{-n} = R_n$ [because $\mathcal{R}(\tau)$ is an even function of τ], we have

$$S_x(\omega) = \frac{1}{T_b} \left(R_0 + 2 \sum_{n=1}^{\infty} R_n \cos n\omega T_b \right) \quad (7.9)$$

Line Coding

$$\begin{aligned} R_n &= \lim_{T \rightarrow \infty} \frac{T_b}{T} \sum_k a_k a_{k+n} \\ &= \lim_{N \rightarrow \infty} \frac{1}{N} \sum_k a_k a_{k+n} \\ &= \overline{a_k a_{k+n}} \end{aligned}$$

$$S_y(\omega) = |P(\omega)|^2 S_x(\omega) \quad (7.10a)$$

$$= \frac{|P(\omega)|^2}{T_b} \left(\sum_{n=-\infty}^{\infty} R_n e^{-jn\omega T_b} \right) \quad (7.10b)$$

$$= \frac{|P(\omega)|^2}{T_b} \left(R_0 + 2 \sum_{n=1}^{\infty} R_n \cos n\omega T_b \right) \quad (7.10c)$$

Using this result, we shall now find the PSDs of various line codes.

Line Coding

Example: Polar Line Coding

In polar signaling, 1 is transmitted by a pulse $p(t)$ and 0 is transmitted by $-p(t)$. In this case, a_k is equally likely to be 1 or -1 , and a_k^2 is always 1. Hence,

$$R_0 = \lim_{N \rightarrow \infty} \frac{1}{N} \sum_k a_k^2$$

There are N pulses and $a_k^2 = 1$ for each one. The summation on the right-hand side of this equation is N . Hence,

$$R_0 = \lim_{N \rightarrow \infty} \frac{1}{N} (N) = 1 \quad (7.11a)$$

Moreover, both a_k and a_{k+1} are either 1 or -1 . Hence, $a_k a_{k+1}$ is either 1 or -1 . Because the pulse amplitude a_k is equally likely to be 1 and -1 on the average, out of N terms the product $a_k a_{k+1}$ is equal to 1 for $N/2$ terms and is equal to -1 for the remaining $N/2$ terms. Therefore,

$$R_1 = \lim_{N \rightarrow \infty} \frac{1}{N} \left[\frac{N}{2} (1) + \frac{N}{2} (-1) \right] = 0 \quad (7.11b)$$

Line Coding

Polar Line Coding

Arguing this way, we see that the product $a_k a_{k+n}$ is also equally likely to be 1 or -1 . Hence,

$$R_n = 0 \quad n \geq 1 \quad (7.11c)$$

Therefore from Eq. (7.10c),

$$\begin{aligned} S_y(\omega) &= \frac{|P(\omega)|^2}{T_b} R_0 \\ &= \frac{|P(\omega)|^2}{T_b} \end{aligned} \quad (7.12)$$

For the sake of comparison of various schemes, we shall consider a specific pulse shape. Let $p(t)$ be a rectangular pulse of width $T_b/2$ (a half-width rectangular pulse), that is,

$$p(t) = \text{rect} \left(\frac{t}{T_b/2} \right) = \text{rect} \left(\frac{2t}{T_b} \right)$$

and

$$P(\omega) = \frac{T_b}{2} \text{sinc} \left(\frac{\omega T_b}{4} \right) \quad (7.13)$$

Line Coding

Polar Line Coding

T_b : bit width or bit duration (in second)

$R_b = 1/T_b$
transmission
rate (Hz)

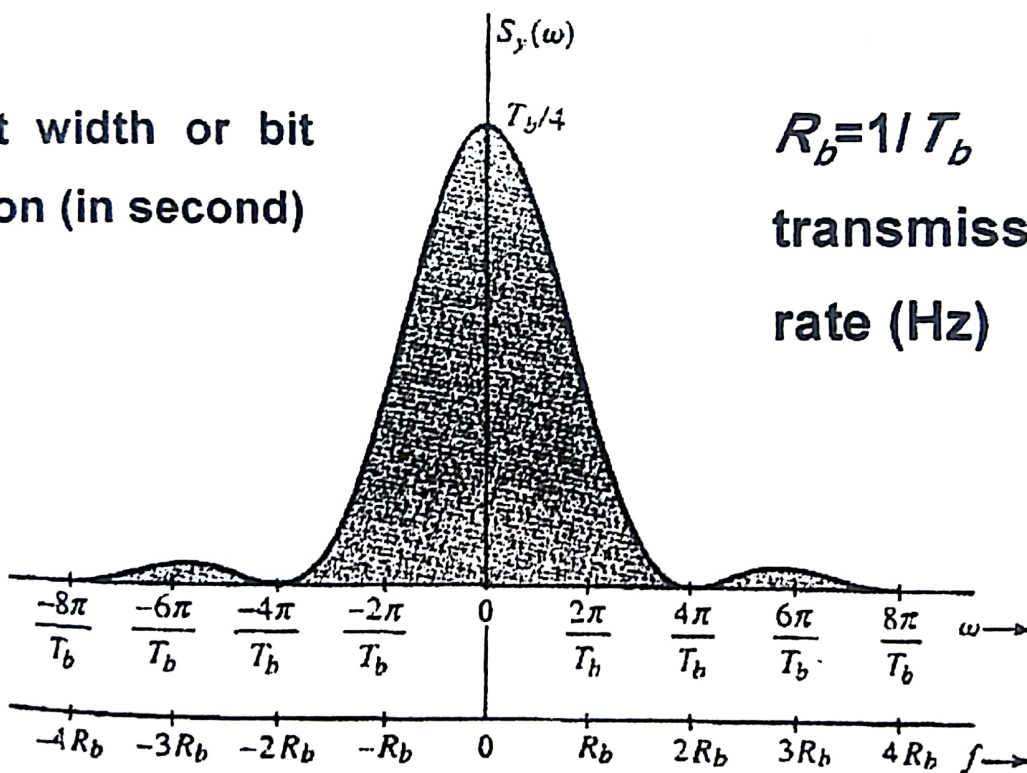


Figure 7.5 Power spectral density of a polar signal.

Line Coding

Polar Line Coding

Figure 7.5 shows the spectrum $S_y(\omega)$. From this spectrum, the essential bandwidth of the signal is seen to be $2R_b$ Hz (where R_b is the clock frequency). This is four times the theoretical bandwidth (Nyquist bandwidth) required to transmit R_b pulses per second. Increasing the pulse width reduces the bandwidth (expansion in the time domain results in compression in the frequency domain). For a full-width pulse (maximum possible pulse width), the essential bandwidth is half, that is, R_b Hz. This is still twice the theoretical bandwidth. Thus, polar signaling is not bandwidth efficient.

Second, polar signaling has no error-detection or error-correction capability. A third disadvantage of polar signaling is that it has nonzero PSD at dc ($\omega = 0$). This will rule out the use of ac coupling during transmission. The ac coupling, which permits transformers and blocking capacitors to aid in impedance matching and bias removal, and which allows dc powering of the line repeaters over the cable pairs, is very important in practice. Later, we shall show how a PSD of a line code may be forced to zero at dc by properly shaping $p(t)$.

On the positive side, polar signaling is the most efficient scheme from the power requirement viewpoint. For a given power, it can be shown that the detection-error probability for a polar scheme is the smallest possible (see Sec. 7.6). Polar signaling is also transparent because there is always some pulse (positive or negative) regardless of the bit sequence. There is no discrete clock frequency component in the spectrum of the polar signal. Rectification of the polar signal, however, yields a periodic signal of the clock frequency and can readily be used to extract timing.

Line Coding

Polar Line Coding

Achieving a DC Null in PSD by Pulse Shaping

Because $S_y(\omega)$, the PSD of a line code, contains a factor $|P(\omega)|^2$, we can force the PSD to have a dc null by selecting a pulse $p(t)$ such that $P(\omega)$ is zero at dc ($\omega = 0$). Because

$$P(\omega) = \int_{-\infty}^{\infty} p(t)e^{-j\omega t} dt$$

we have

$$P(0) = \int_{-\infty}^{\infty} p(t) dt$$

Hence, if the area under $p(t)$ is made zero, $P(0)$ is zero, and we have a dc null in the PSD. For a rectangular pulse, one possible shape of $p(t)$ to accomplish this is shown in Fig. 7.6a. When we use this pulse with polar line coding, the resulting signal is known as Manchester, or split-phase (also twinned-binary) signal.

Line Coding

Polar Line Coding

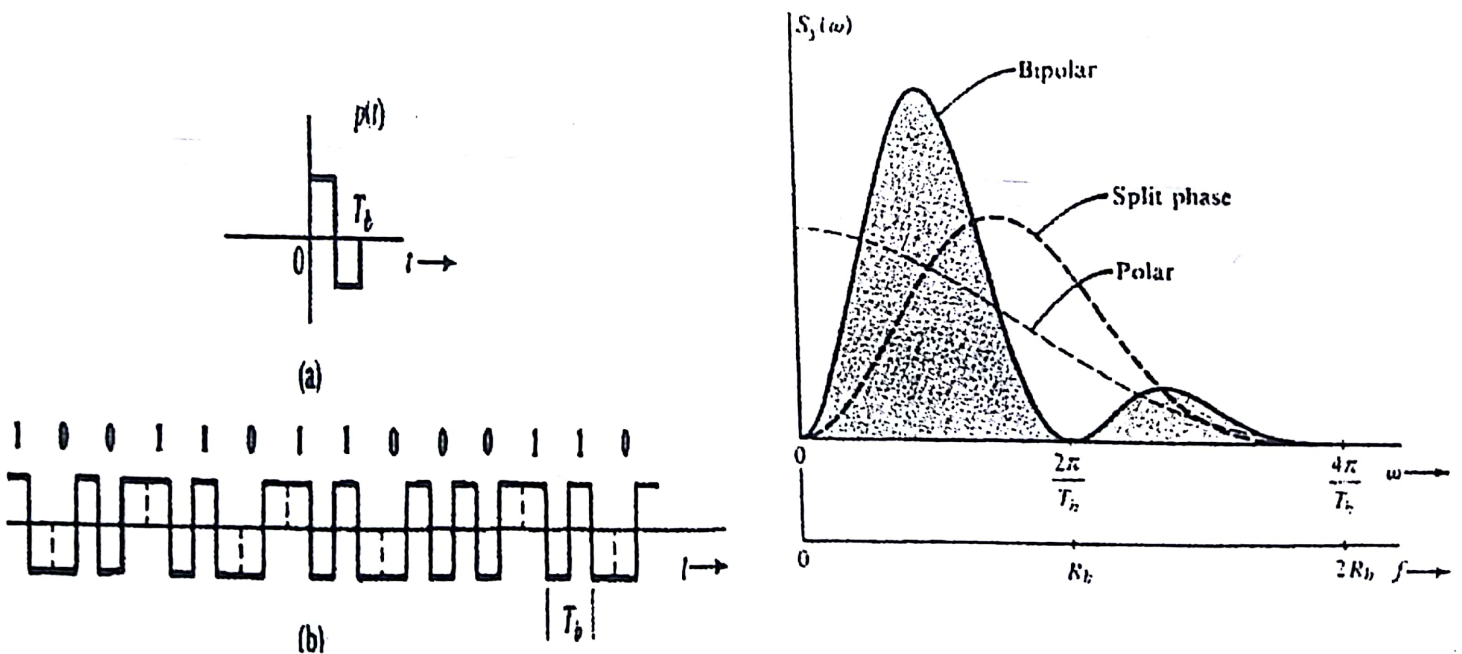


Figure 7.6 Split-phase (Manchester or twinned-binary) signal. (a) Basic pulse $p(t)$ for Manchester signaling. (b) PSD of Manchester signaling.

Line Coding

Example: On-off Line Coding

In this case a 1 is transmitted by a pulse $p(t)$ and a 0 is transmitted by no pulse. Hence, a pulse strength a_k is equally likely to be 1 or 0. Out of N pulses in the interval of T seconds, a_k is 1 for $N/2$ pulses and 0 for the remaining $N/2$ pulses on the average. Hence,

$$R_0 = \lim_{N \rightarrow \infty} \frac{1}{N} \left[\frac{N}{2}(1) + \frac{N}{2}(0) \right] = \frac{1}{2} \quad (7.15)$$

To compute R_n we need to consider the product $a_k a_{k+n}$. Since a_k and a_{k+n} are equally likely to be 1 or 0, the product $a_k a_{k+n}$ is equally likely to be 1×1 , 1×0 , 0×1 , or 0×0 , that is, 1, 0, 0, 0. Therefore, on the average, the product $a_k a_{k+n}$ is equal to 1 for $N/4$ terms and 0 for $3N/4$ terms, and

$$R_n = \lim_{N \rightarrow \infty} \frac{1}{N} \left[\frac{N}{4}(1) + \frac{3N}{4}(0) \right] = \frac{1}{4} \quad n \geq 1 \quad (7.16)$$

Line Coding

On-off Line Coding

Therefore [Eq. (7.8)],

$$S_x(\omega) = \frac{1}{2T_b} + \frac{1}{4T_b} \sum_{\substack{n=-\infty \\ n \neq 0}}^{\infty} e^{-jn\omega T_b} \quad (7.17a)$$

$$= \frac{1}{4T_b} + \frac{1}{4T_b} \sum_{n=-\infty}^{\infty} e^{-jn\omega T_b} \quad (7.17b)$$

Equation (7.17b) is obtained from Eq. (7.17a) by splitting the term $1/2T_b$ corresponding to R_0 into two: $1/4T_b$ outside the summation and $1/4T_b$ inside the summation (corresponding to $n = 0$). We now use the formula (see the footnote for a proof)*

$$\sum_{n=-\infty}^{\infty} e^{-jn\omega T_b} = \frac{2\pi}{T_b} \sum_{n=-\infty}^{\infty} \delta\left(\omega - \frac{2\pi n}{T_b}\right)$$

Line Coding

On-off Line Coding

Substitution of this result into Eq. (7.17b) yields

$$S_x(\omega) = \frac{1}{4T_b} + \frac{2\pi}{4T_b^2} \sum_{n=-\infty}^{\infty} \delta\left(\omega - \frac{2\pi n}{T_b}\right) \quad (7.18a)$$

and the desired PSD of the on-off waveform $y(t)$ is [Eq. (7.10a)]

$$S_y(\omega) = \frac{|P(\omega)|^2}{4T_b} \left[1 + \frac{2\pi}{T_b} \sum_{n=-\infty}^{\infty} \delta\left(\omega - \frac{2\pi n}{T_b}\right) \right] \quad (7.18b)$$

For the case of a half-width rectangular pulse [see Eq. (7.13)]

$$S_y(\omega) = \frac{T_b}{16} \text{sinc}^2\left(\frac{\omega T_b}{4}\right) \left[1 + \frac{2\pi}{T_b} \sum_{n=-\infty}^{\infty} \delta\left(\omega - \frac{2\pi n}{T_b}\right) \right] \quad (7.19)$$

Line Coding

On-off Line Coding

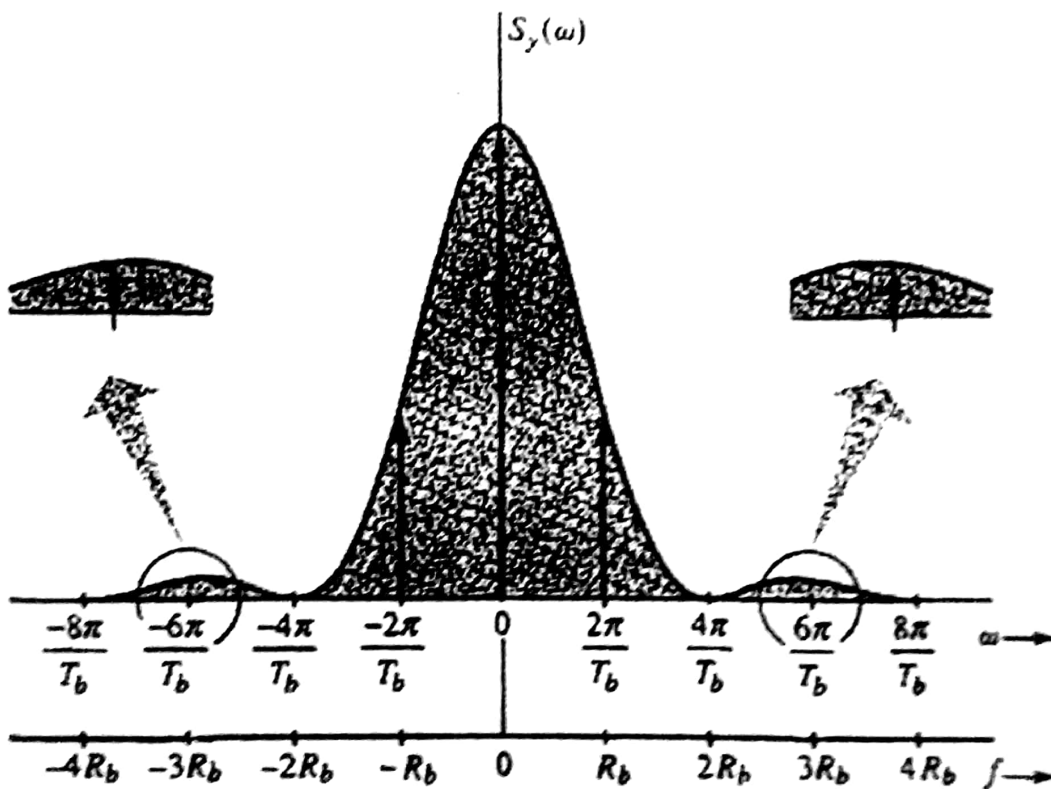


Figure 7.7 Power spectral density of an on-off signal.

Line Coding

Example: Bipolar Line Coding

This is the signaling scheme used in PCM these days. A 0 is transmitted by no pulse, and a 1 is transmitted by a pulse $p(t)$ or $-p(t)$, depending on whether the previous 1 was transmitted by $-p(t)$ or $p(t)$. With consecutive pulses alternating, we can avoid the dc wander and thus cause a dc null in the PSD. Bipolar signaling actually uses three symbols $\{p(t), 0, \text{and } -p(t)\}$, and, hence, it is in reality ternary rather than binary signaling.

To calculate the PSD, we have

$$R_0 = \lim_{N \rightarrow \infty} \frac{1}{N} \sum_k a_k^2$$

Line Coding

Bipolar Line Coding

On the average, half of the a_k 's are 0, and the remaining half are either 1 or -1, with $a_k^2 = 1$. Therefore,

$$R_0 = \lim_{N \rightarrow \infty} \frac{1}{N} \left[\frac{N}{2} (\pm 1)^2 + \frac{N}{2} (0) \right] = \frac{1}{2}$$

To compute R_1 , we consider the pulse strength product $a_k a_{k+1}$. There are four possible equally likely sequences of two bits: 11, 10, 01, 00. Since bit 0 is encoded by no pulse ($a_k = 0$), the product $a_k a_{k+1} = 0$ for the last three of these sequences. This means that, on the average, $3N/4$ combinations have $a_k a_{k+1} = 0$ and only $N/4$ combinations have nonzero $a_k a_{k+1}$. Because of the bipolar rule, the bit sequence 11 can only be encoded by two consecutive pulses of opposite polarities. This means the product $a_k a_{k+1} = -1$ for the $N/4$ combinations. Therefore,

$$R_1 = \lim_{N \rightarrow \infty} \frac{1}{N} \left[\frac{N}{4} (-1) + \frac{3N}{4} (0) \right] = -\frac{1}{4}$$

To compute R_2 in a similar way, we need to observe the product $a_k a_{k+2}$. For this, we need to consider all possible combinations of three bits in sequence. There are eight equally likely combinations: 111, 101, 110, 100, 011, 010, 001, 000. The last six combinations have either the first or the last bit 0, or both. Hence, $a_k a_{k+2} = 0$ for all these six combinations. The first two combinations are the only ones that yield nonzero $a_k a_{k+2}$. Using the bipolar rule, the first and the third pulses in the combination 111 are of the same polarity, yielding $a_k a_{k+2} = 1$. But for 101, the first and third pulses are of opposite polarity, yielding $a_k a_{k+2} = -1$. Thus, on the average, $a_k a_{k+2} = 1$ for $N/8$ terms, -1 for $N/8$ terms, and 0 for $3N/4$ terms. Hence,

$$R_2 = \lim_{N \rightarrow \infty} \frac{1}{N} \left[\frac{N}{8} (1) + \frac{N}{8} (-1) + \frac{3N}{8} (0) \right] = 0$$

Line Coding

Bipolar Line Coding

In general,

$$R_n = \lim_{N \rightarrow \infty} \frac{1}{N} \sum_k a_k a_{k+n}$$

For $n > 1$, the product $a_k a_{k+2}$ can be 1, -1, or 0. Moreover, an equal number of combinations have values 1 and -1. This causes $R_n = 0$. Thus,

$$R_n = 0 \quad n > 1$$

and [see Eq. (7.10c)]

$$S_y(\omega) = \frac{|P(\omega)|^2}{2T_b} [1 - \cos \omega T_b] \quad (7.20a)$$

$$= \frac{|P(\omega)|^2}{T_b} \sin^2 \left(\frac{\omega T_b}{2} \right) \quad (7.20b)$$

Note that $S_y(\omega) = 0$ for $\omega = 0$ (dc), regardless of $P(\omega)$. Hence, the PSD has a dc null, which is desirable for ac coupling. Moreover, $\sin^2(\omega T_b/2) = 0$ at $\omega = 2\pi/T_b$, that is, at $1/T_b = R_b$ Hz. Thus, regardless of $P(\omega)$, we are assured of a bandwidth of R_b Hz. For the half-width pulse,

$$S_y(\omega) = \frac{T_b}{4} \operatorname{sinc}^2 \left(\frac{\omega T_b}{4} \right) \sin^2 \left(\frac{\omega T_b}{2} \right) \quad (7.21)$$

Line Coding

Bipolar Line Coding (Advantages)

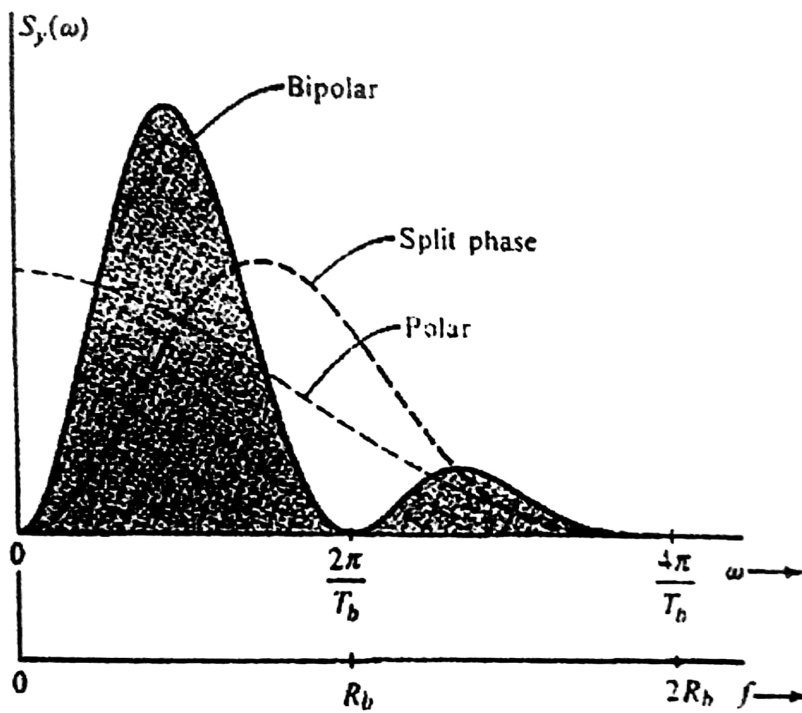


Figure 7.8 PSD of bipolar, polar, and split-phase signals normalized for equal powers. Half-width rectangular pulses are used.

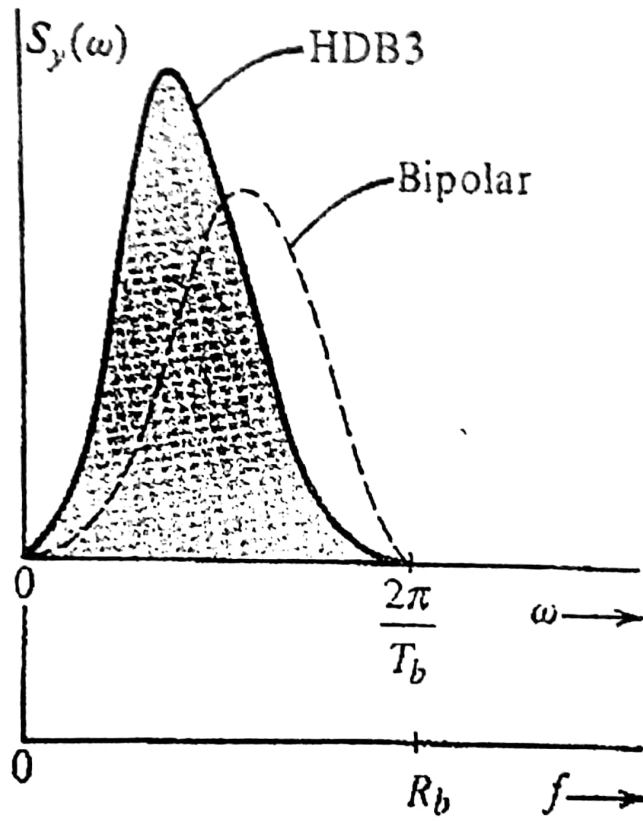
Line Coding

High Density Bipolar

- To solve the non transparency problem of the bipolar line any sequence of $n+1$ successive "0" is replaced by a special code.
- HBD3 is adopted as international standard. Any sequence "0000" is replaced by "000V" or "B00V"
 - B is regular "1"
 - V is a "1" that violates the bipolar rule
- The choice of "000V" or "B00V" is made such that consecutive "V" pulses alternate signs to avoid dc wander and maintain dc null PSD. How?
- "B00V" is used when the number of "1" after the last special sequence is even. Otherwise the sequence "000V" is used.

Line Coding

High Density Bipolar



Line Coding

Binary with 8 zero Substitution (B8ZS)

Any sequence of 8 zeros is replaced by "000VB0VB". This code is unlikely to happen which make its detection at the receiver easy. This signaling is used with the DS1 signal.

Binary with 6 zero Substitution (B6ZS)

Any sequence of 6 zeros is replaced by "0VB0VB". This code is unlikely to happen which make its detection at the receiver easy. This signaling is used with the DS2 signal.

Line Coding

Binary with 8 zero Substitution (B8ZS)

Any sequence of 8 zeros is replaced by "000VB0VB". This code is unlikely to happen which make its detection at the receiver easy. This signaling is used with the DS1 signal.

Binary with 6 zero Substitution (B6ZS)

Any sequence of 6 zeros is replaced by "0VB0VB". This code is unlikely to happen which make its detection at the receiver easy. This signaling is used with the DS2 signal.



EE 325: Chapter 7 (Lec.#2)

Principals of Digital Data Transmission

M. A. Smadi

Pulse Shaping

- The PSD $S_{\mu}(\omega)$ is strongly and directly influenced by the pulse shape $p(t)$ because $S_{\mu}(\omega)$ contains the term $|P(\omega)|^2$. Thus, compared to the nature of the line code, the pulse shape is a much more potent factor in terms of shaping the PSD $S_{\mu}(\omega)$.

Pulse Shaping

Problem

- We need to transmit a pulse every T_b interval, the k^{th} pulse being $a_k p(t - kT_b)$. The channel has a finite bandwidth, and we are required to detect the pulse amplitude a_k correctly (that is, without ISI).
- In our discussion so far, we are considering time-limited pulses. Since such pulses cannot be band-limited, part of their spectra is suppressed by a band-limited channel. This causes pulse distortion (spreading out) and, consequently, Intersymbol Interference (ISI).

Pulse Shaping

We can try to resolve this difficulty by using pulses which are band-limited to begin with so that they can be transmitted intact over a band-limited channel. But band-limited pulses cannot be time-limited. Obviously, various pulses will overlap and cause an inter-symbol Interference (ISI). Thus, whether we begin with time-limited pulses or band-limited pulses, it appears that ISI cannot be avoided

Solution

Pulse amplitudes can be detected correctly despite pulse spreading (or overlapping) if there is no ISI at the decision-making instants. This can be accomplished by a properly shaped band-limited pulse. To eliminate ISI, **Nyquist** proposed three different criteria for pulse shaping.

Pulse Shaping

Nyquist Criterion for Zero ISI

In the first method, Nyquist achieves zero ISI by choosing a pulse shape that has a nonzero amplitude at its center (say $t = 0$) and zero amplitudes at $t = nT_b$ ($n = 1, 2, 3, \dots$), T_b is the separation between successive transmitted pulses (Fig. 7.10a),

$$p(t) = \begin{cases} 1 & t = 0 \\ 0 & t = \pm nT_b \end{cases} \quad \left(T_b = \frac{1}{R_b} \right) \quad (7.22)$$

Pulse Shaping

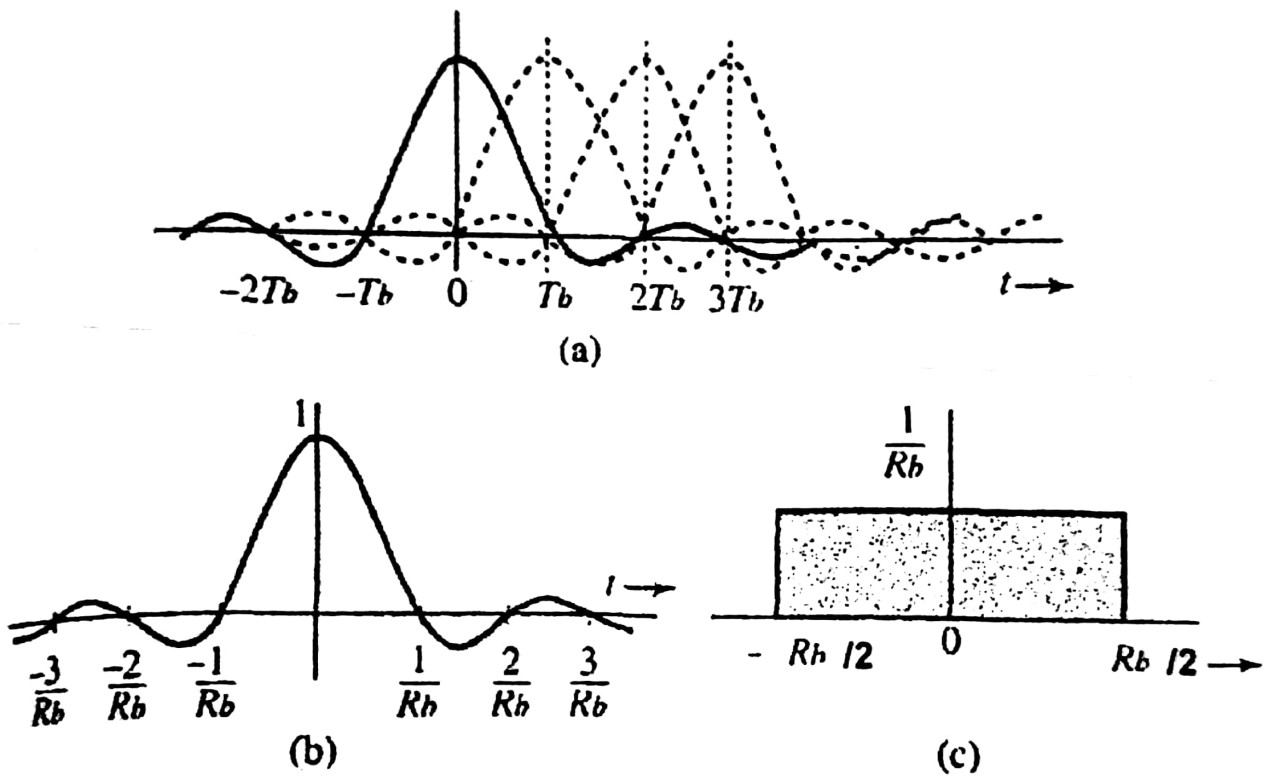


Figure 7.10 Minimum bandwidth pulse that satisfies the Nyquist criterion and its spectrum.

Pulse Shaping

Question

Now transmission of R_b bit/s requires a theoretical minimum bandwidth of $R_b/2$ Hz. It would be nice if a pulse satisfying Nyquist's criterion had this minimum bandwidth $R_b/2$ Hz. Can we find such a pulse $p(t)$?

Answer

Yes, This pulse, $p(t) = \text{sinc}(\pi R_b t)$, (see Fig. 7.10b) has the property

$$\text{sinc}(\pi R_b t) = \begin{cases} 1 & t = 0 \\ 0 & t = \pm n T_b \end{cases} \quad \left(T_b = \frac{1}{R_b} \right) \quad (7.23a)$$

Moreover, the Fourier transform of this pulse is

$$P(\omega) = \frac{1}{R_b} \text{rect} \left(\frac{\omega}{2\pi R_b} \right) \quad (7.23b)$$

Pulse Shaping

Problem

- This scheme shows that we can attain the theoretical limit of performance by using a *sinc* pulse. Unfortunately, this pulse is impractical because it is non causal.
- We will have to wait an infinite time to generate it. Any attempt to truncate it would increase its bandwidth beyond $R_b/2$ Hz. Also it has the undesirable feature that it decays too slowly at a rate of $1/t$. This causes some serious practical problems, specially when any deviation from R_b rate occurs at the transmitter or the receiver sides.

Pulse Shaping

Solution

The solution is to find a pulse $p(t)$ that satisfies Eq. (7.22) but decays faster than $1/t$. Nyquist shown that such a pulse have excess bandwidth. i.e. the bandwidth of $P(\omega)$ is $(\omega_b/2) + \omega_x$, where ω_x , is the bandwidth in excess of the theoretical minimum bandwidth. Let r be the ratio of the excess bandwidth ω_x , to the theoretical minimum bandwidth $\omega_b/2$.

$$r = \frac{\text{excess bandwidth}}{\text{theoretical minimum bandwidth}} = \frac{\omega_x}{\omega_b/2} = \frac{2\omega_x}{\omega_b}$$

Pulse Shaping

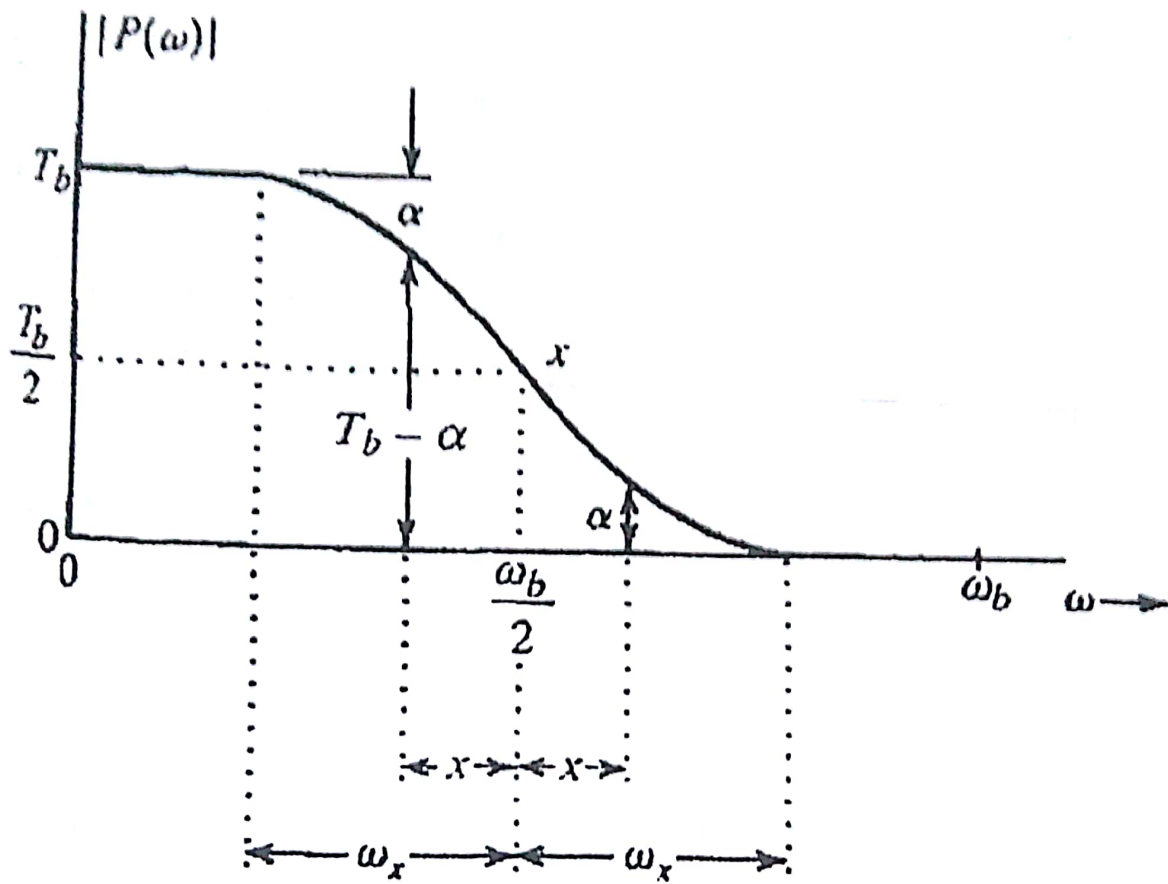


Figure 7.12 Vestigial spectrum.

Pulse Shaping

- One family of spectra that satisfies the Nyquist criterion is :

$$P(\omega) = \begin{cases} \frac{1}{2} \left\{ 1 - \sin \left(\frac{\pi [\omega - (\omega_b/2)]}{2\omega_x} \right) \right\} & \left| \omega - \frac{\omega_b}{2} \right| < \omega_x \\ 0 & \left| \omega \right| > \frac{\omega_b}{2} + \omega_x \\ 1 & \left| \omega \right| < \frac{\omega_b}{2} - \omega_x \end{cases} \quad (7.33)$$

Figure 7.13a shows three curves, corresponding to $\omega_x = 0$ ($r = 0$), $\omega_x = \omega_b/4$ ($r = 0.5$), $\omega_x = \omega_b/2$ ($r = 1$). The respective impulse responses are shown in Fig. 7.13b. It can be seen that increasing ω_x , (or r) improves $p(t)$; that is, more gradual cutoff reduces the oscillation nature of $p(t)$ and causes it to decay more rapidly.

Pulse Shaping

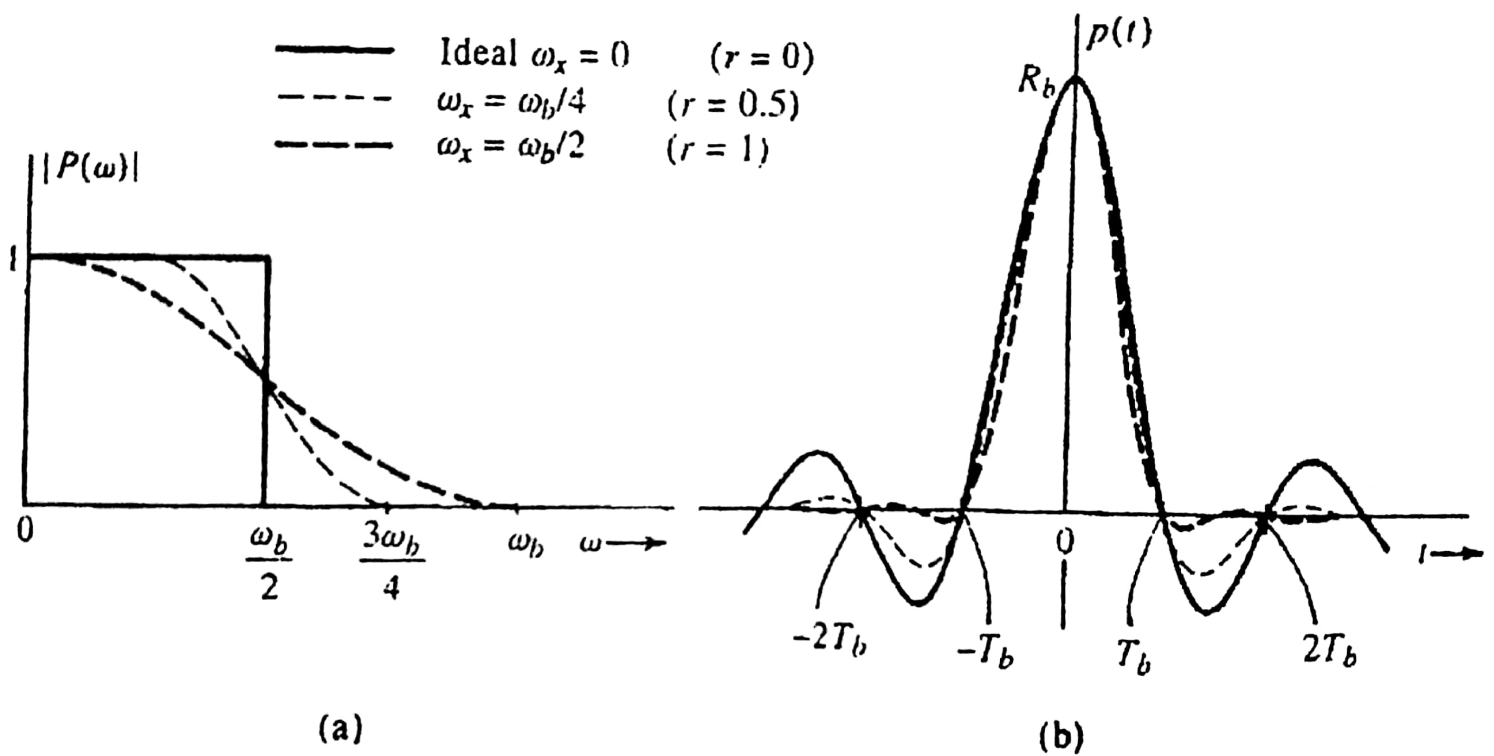


Figure 7.13 Pulses satisfying the Nyquist criterion.

Pulse Shaping

For the case of the maximum value of $\omega x = \omega b/2$ ($r = 1$), Eq. (7.33) reduces to

$$P(\omega) = \frac{1}{2} \left(1 + \cos \frac{\omega}{2R_b} \right) \text{rect} \left(\frac{\omega}{4\pi R_b} \right) \quad (7.34a)$$

$$= \cos^2 \left(\frac{\omega}{4R_b} \right) \text{rect} \left(\frac{\omega}{4\pi R_b} \right) \quad (7.34b)$$

This characteristic is known in the literature as the *raised-cosine* characteristic, because it represents a cosine raised by its peak amplitude. It is also known as the *full-cosine roll off*.

Pulse Shaping

- The inverse Fourier transform of this spectrum is readily found as

$$p(t) = R_b \frac{\cos \pi R_b t}{1 - 4R_b^2 t^2} \text{sinc}(\pi R_b t) \quad (7.35)$$

Characteristics

1. Bandwidth = R_B
2. *Amplitude* = R_B at $t = 0$, zero at signaling instant and zero at midway
3. Decays with $1/t^2$ rate.

Pulse Shaping

Signaling with Controlled ISI: Partial Response Signals

- The Nyquist criterion pulse results in a bandwidth somewhat larger than the theoretical minimum. If we wish to reduce the pulse bandwidth further, we must somehow widen pulse $p(t)$
- Consider the following pulse

$$p(nT_b) = \begin{cases} 1 & n = 0, 1 \\ 0 & \text{for all other } n \end{cases} \quad (7.36)$$

Pulse Shaping

Signaling with Controlled ISI: Partial Response Signals

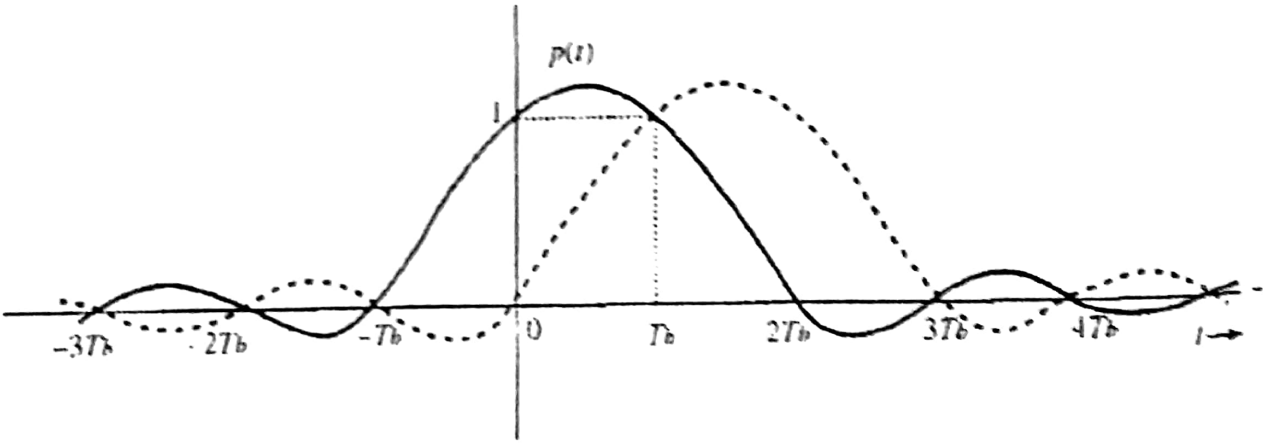


Figure 7.14 Communication using duobinary pulses.

Pulse Shaping

Signaling with Controlled ISI: Partial Response Signals

- We use polar signaling using this pulse. Thus, 1 is transmitted by $p(t)$ and 0 is transmitted using the pulse $-p(t)$. The received signal is sampled at $t = n T_b$, and the pulse $p(t)$ has value at all n except $n = 0$ and 1, where its value is 1.
- Clearly, such a pulse causes zero ISI with all the pulses except the succeeding pulse. Therefore, we need worry about the ISI with the succeeding pulse only.

Pulse Shaping

Signaling with Controlled ISI: Partial Response Signals

- Consider two such successive pulses located at 0 and T_b respectively. If both pulses were positive, the sample value of the resulting signal at $t = T_b$ would be 2 . If the both pulses were negative, the sample value would be -2 . But if the pulses were of opposite polarity, the sample value would be 0 . This clearly allows us to correct decisions at the sampling instants.
- The decision rule is as follows. If the sample value is positive, the present bit is 1 and the previous bit is also 1 . If the sample value is negative, the present bit is 0 and the previous bit is also 0 . If the sample value is zero, the present bit is the complement of the previous bit. The knowledge of the previous bit then allows the determination of the present bit.

Pulse Shaping

Signaling with Controlled ISI: Partial Response Signals

Transmitted sequence	1	1	0	1	1	0	0	0	1	0	1	1	1
Samples of $x(t)$	1	2	0	0	2	0	-2	-2	0	0	0	2	2
Detected sequence	1	1	0	1	1	0	0	0	1	0	1	1	1

Figure 7.15 Transmitted bits and the received samples in controlled ISI signaling.

Example of a Duobinary Pulse

If we restrict the pulse bandwidth to $R_b/2$, then following the procedure of Example 6.1, we can show that (see Prob. 7.3-9) only the following pulse $p(t)$ meets the requirement in Eq. (7.36) for the duobinary pulse,

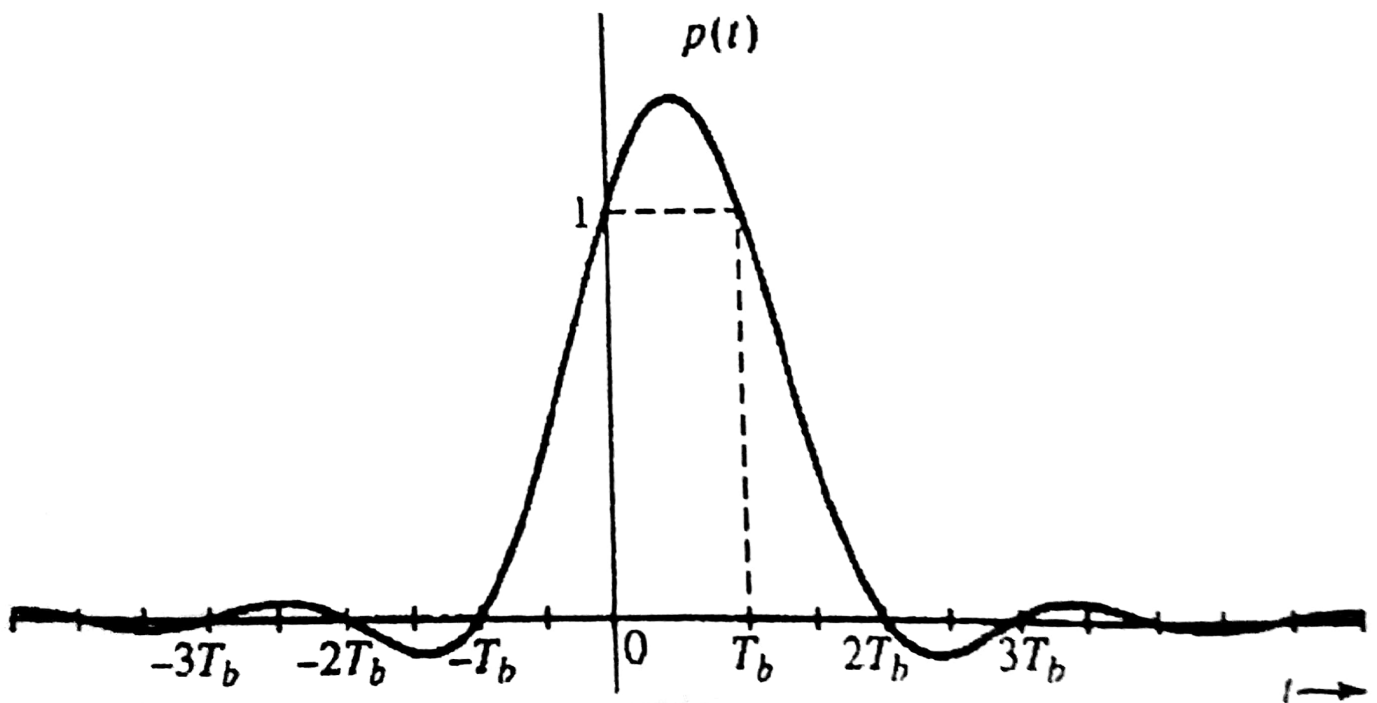
$$p(t) = \frac{\sin(\pi R_b t)}{\pi R_b t (1 - R_b t)} \quad (7.37)$$

The Fourier transform $P(\omega)$ of the pulse $p(t)$ is given by (see Prob. 7.3-8)

$$P(\omega) = \frac{2}{R_b} \cos\left(\frac{\omega}{2R_b}\right) \text{rect}\left(\frac{\omega}{2\pi R_b}\right) e^{-j\frac{\omega}{2R_b}} \quad (7.38)$$

Pulse Shaping

Signaling with Controlled ISI: Partial Response Signals



(a)

Pulse Shaping

Signaling with Controlled ISI: Partial Response Signals

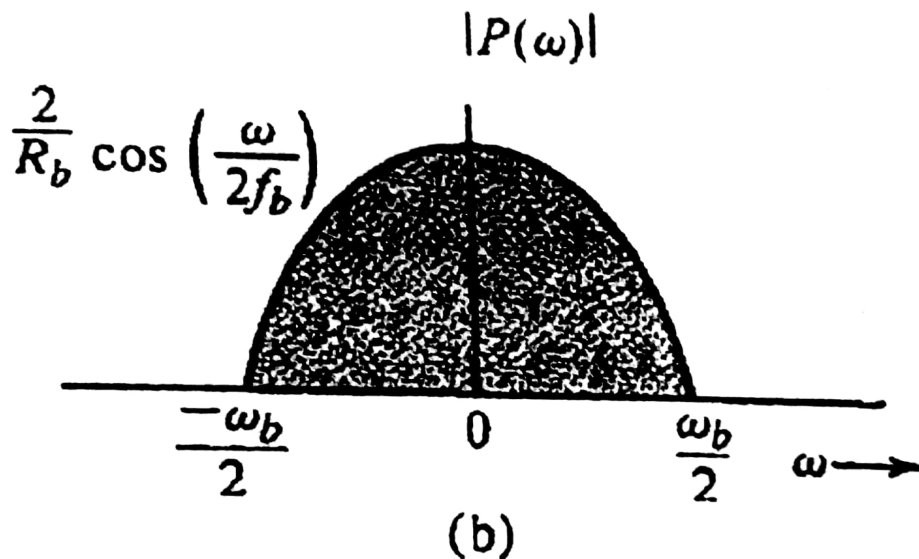


Figure 7.16 Minimum-bandwidth pulse that satisfies the duobinary pulse criterion and its spectrum.

Pulse Shaping

Problem

When we receive a signal 0V \rightarrow the value of the corresponding bit depends on the previous bit. If we make error in the previous bit we will have error in the current bit \rightarrow error propagation

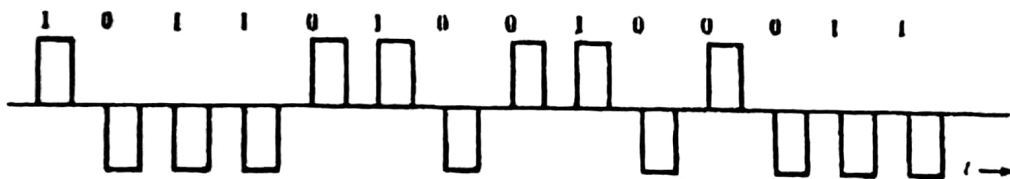
Solution

Use of the differential coding with the duobinary pulse.

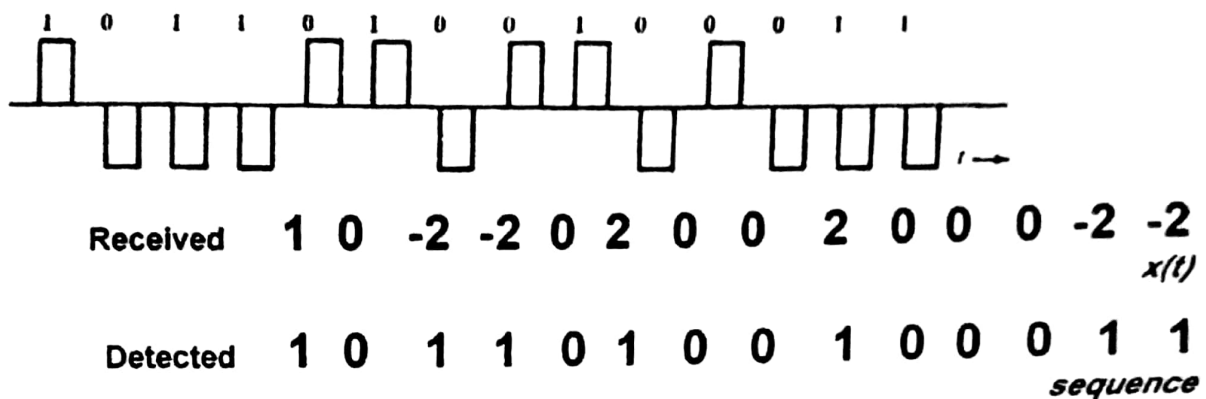
Use of Differential Coding

1 \rightarrow same pulse as previous bit.

0 \rightarrow opposite pulse compared to previous bit



Pulse Shaping



Advantages

1. Independent decision making for each bit → no error propagation
2. Simplified decision making circuit:

$(2V, -2V) \rightarrow 1$ logic; $0V \rightarrow 0$ logic

M-ary Communication

- In the past we considered binary signals that is each symbol represent 1 bit with 2 levels ($M=2$)
- In this paragraph we discuss the aspect of M-ary communication

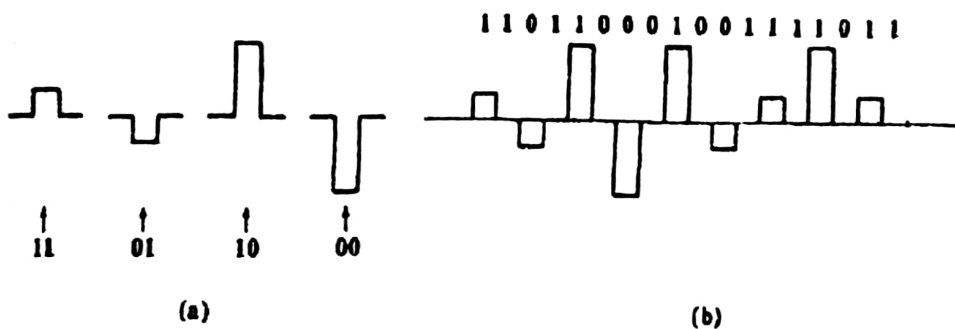


Figure 7.25 4-ary multi-amplitude signal.

4-ary (quaternary) symbols or pulses

M-ary Communication

A 4-ary symbol represents 2 bits

A 16-ary symbol represents 4 bits

An M -ary symbol represents I_M bits

$$M = 2^{(I_M)} \text{ and } I_M = \log_2(M) \quad \text{binary digits or bits}$$

Pulse Shaping in the Multi-amplitude Case:

1. Pulses satisfying zero ISI
2. Pulses satisfying controlled ISI
3. Orthogonal pulses
4. Huge number of choices

M-ary Communication

Orthogonal pulses

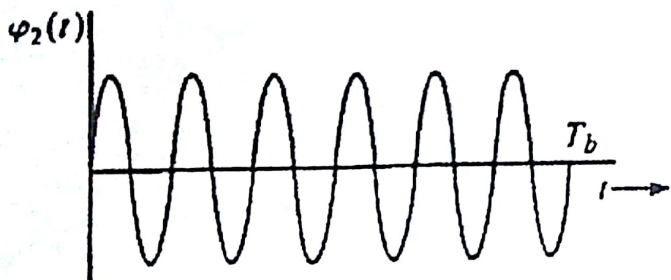
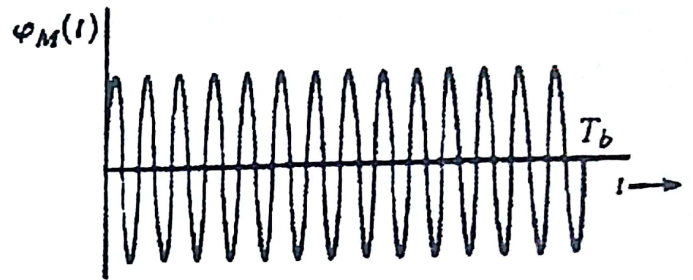
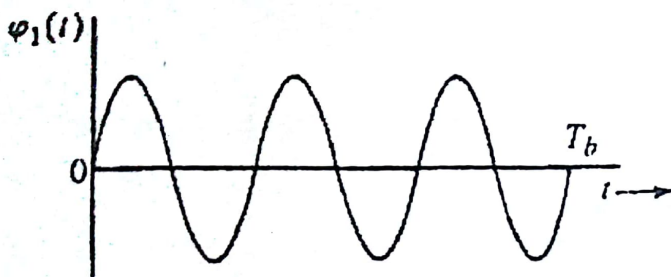
Consider M orthogonal pulses $\phi_1(t), \phi_2(t), \dots, \phi_M(t)$

$$\int_0^{T_b} \phi_i(t)\phi_j(t) dt = \begin{cases} C & i = j \\ 0 & i \neq j \end{cases}$$

Pulse example

$$\phi_k(t) = \begin{cases} \sin \frac{2\pi kt}{T_b} & 0 < t < T_b \\ 0 & \text{otherwise} \end{cases} \quad k = 1, 2, \dots, M$$

M-ary Communication



Question

Why orthogonal pulses

Answer

Essential for symbol detection. Explain

Digital Carrier Systems

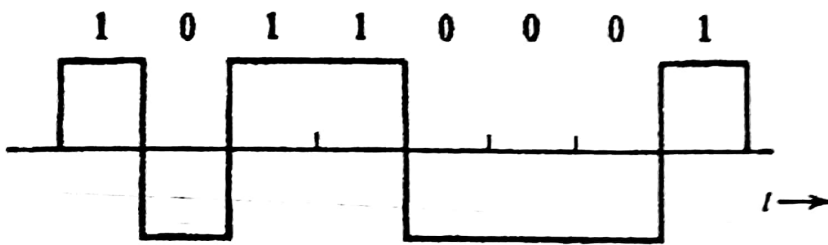
- So far we have discussed baseband communication :
- No shift (translation) of the message signal spectrum. This is good for two-wire, coaxial or optical fiber communication. But not good for wireless transmission.
- As seen in chapter 3, frequency shifting using carrier modulation allows
 1. *wireless transmission*
 2. *Frequency division multiplexing (FDM)*
- Similar as in the analog communication we distinguish three main modulation techniques
 1. *Amplitude shift keying (ASK), equivalent to AM*
 2. *Phase shift keying (PSK), equivalent to PM*
 3. *Frequency shift keying (FSK), equivalent to FM*

Digital Carrier Systems

- So far we have discussed baseband communication :
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 2. *Phase shift keying (PSK), equivalent to PM*
 3. *Frequency shift keying (FSK), equivalent to FM*

Digital Carrier Systems

2&3. Phase shift keying (PSK) and frequency shift keying (FSK)



(a)

(a) Modulating signal $m(t)$



(b)

(b) PSK modulated signal $m(t)\cos(\omega t)$



(c)

(c) FSK modulated signal

Digital Carrier Systems

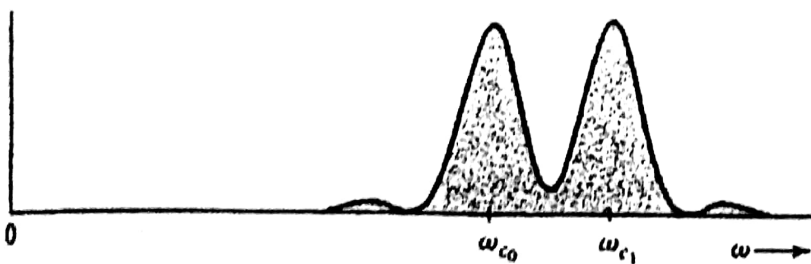
Spectra of ASK, PSK and FSK signals



ASK



PSK



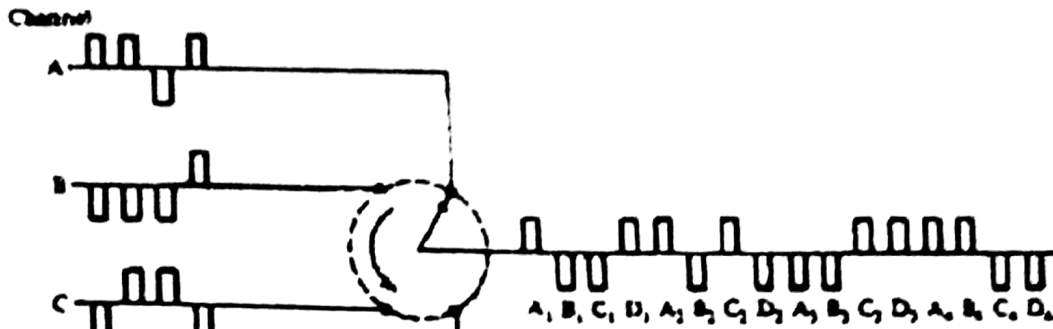
FSK

Digital Carrier Systems

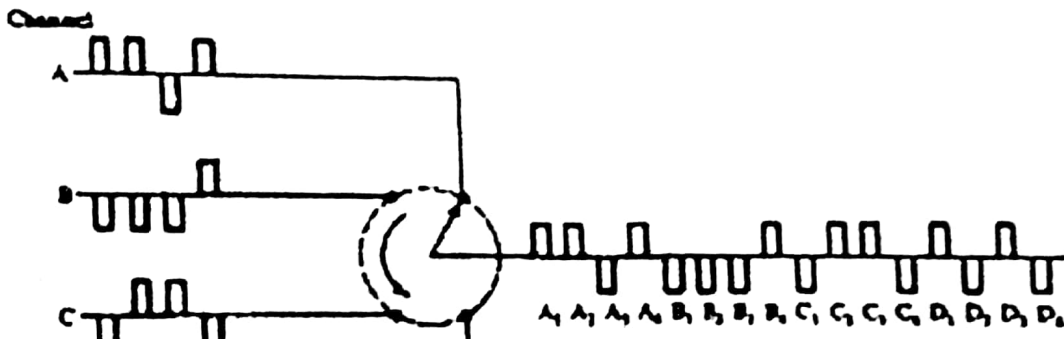
Demodulation

- 1. ASK signals can be demodulated coherently (synchronous detection) or non coherently (envelop detector)*
- 2. PSK signals cannot be demodulated using envelop detectors since 1 and 0 have the same envelop. → only coherently (synchronous detection) is possible.*
- 3. FSK signals can be demodulated coherently or non coherently but using two frequencies*

Digital Multiplexing

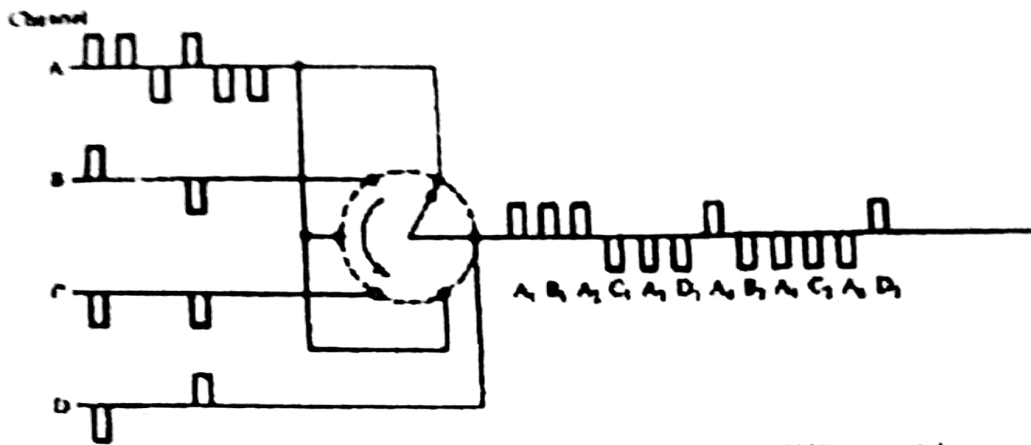


(a) Digit interleaving

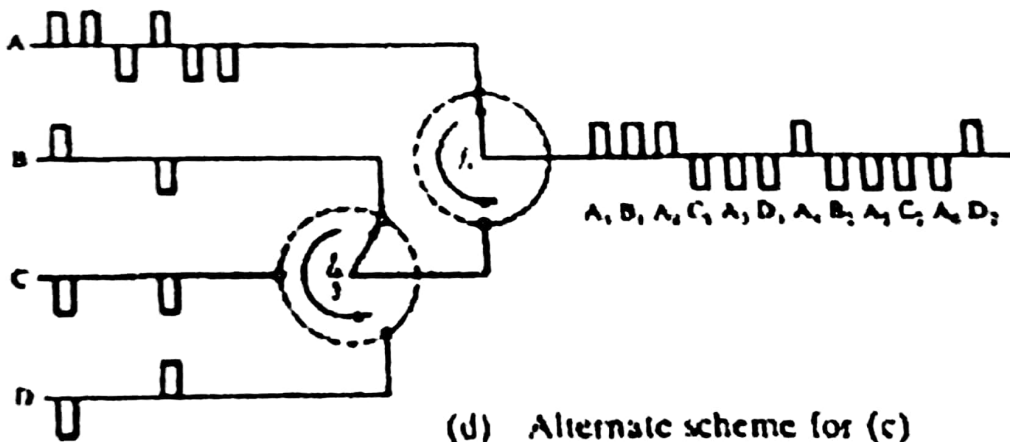


(b) Word (or byte) interleaving

Digital Multiplexing



(c) Interleaving channel having different bit rate

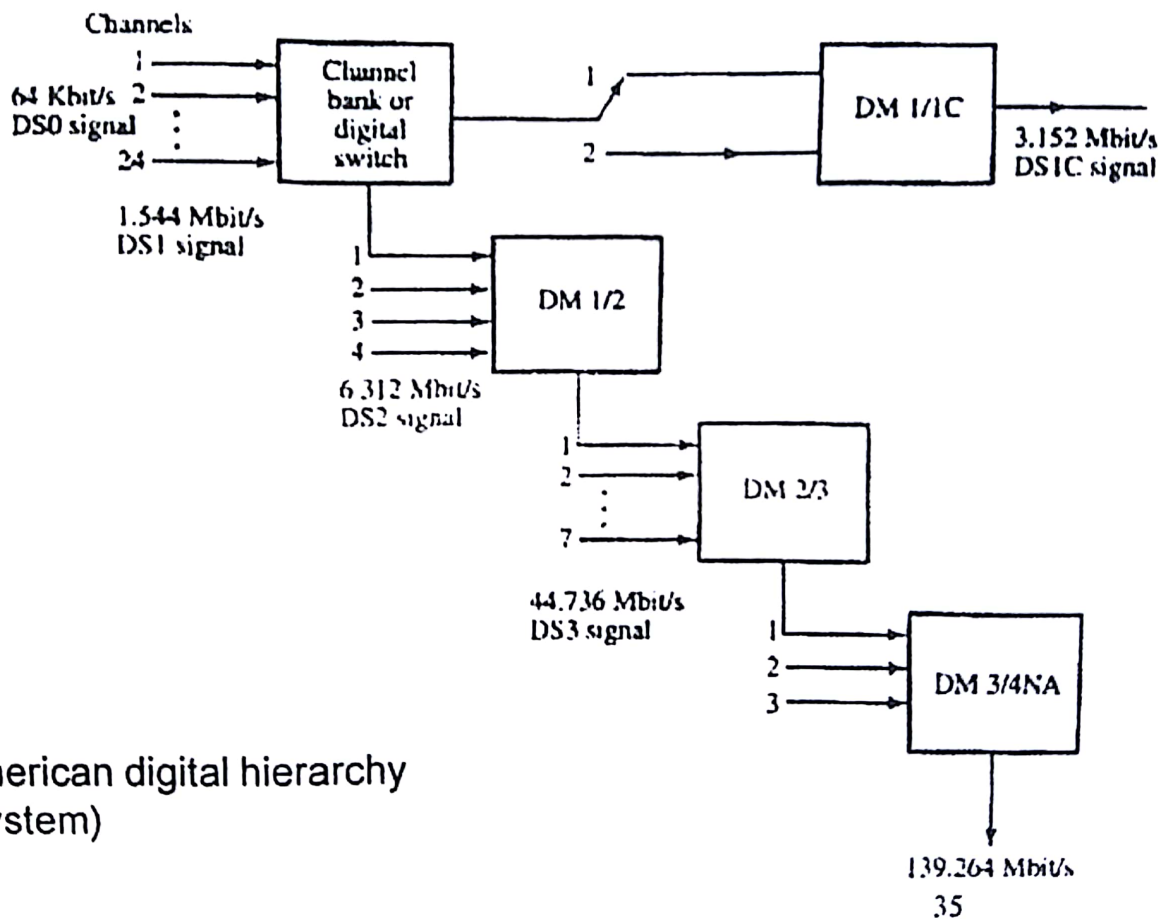


(d) Alternate scheme for (c)

Figure 7.33 Time-division multiplexing of digital signals.

Digital Multiplexing

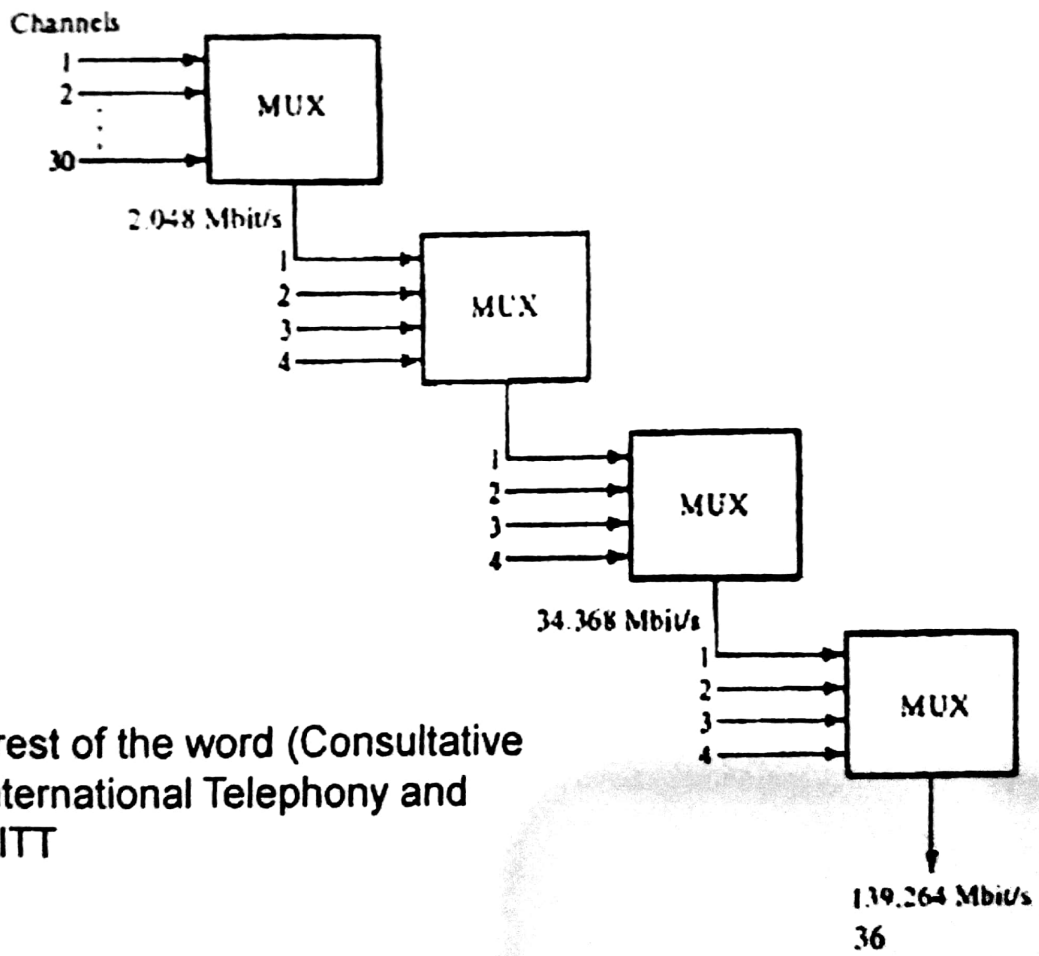
Digital Hierarchy



North American digital hierarchy (AT&T system)

Digital Multiplexing

Digital Hierarchy



Europe and the rest of the world (Consultative Committee on International Telephony and Telegraphy), CCITT