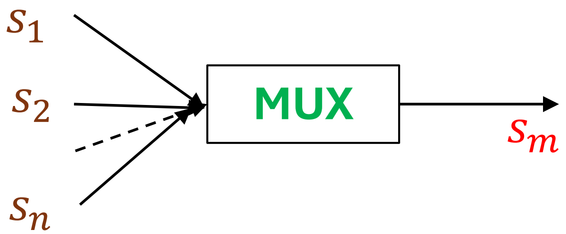
DIGITIZATION OF ONE-DIMENSION SIGNALS

ECE426 – DIGITAL COMMUNICATION

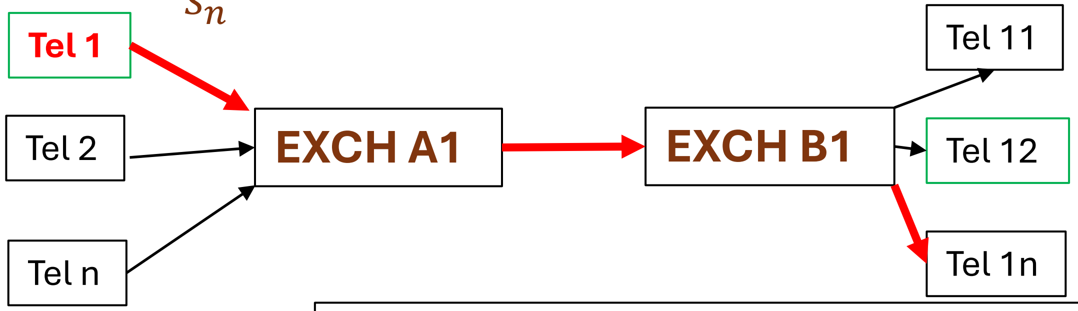
Monday, February 16, 2026

RECAP: ADVANTAGES OF DIGITAL COMMUNICATION NETWORKS /1

1. Ease of multiplexing



2. Ease of Signaling



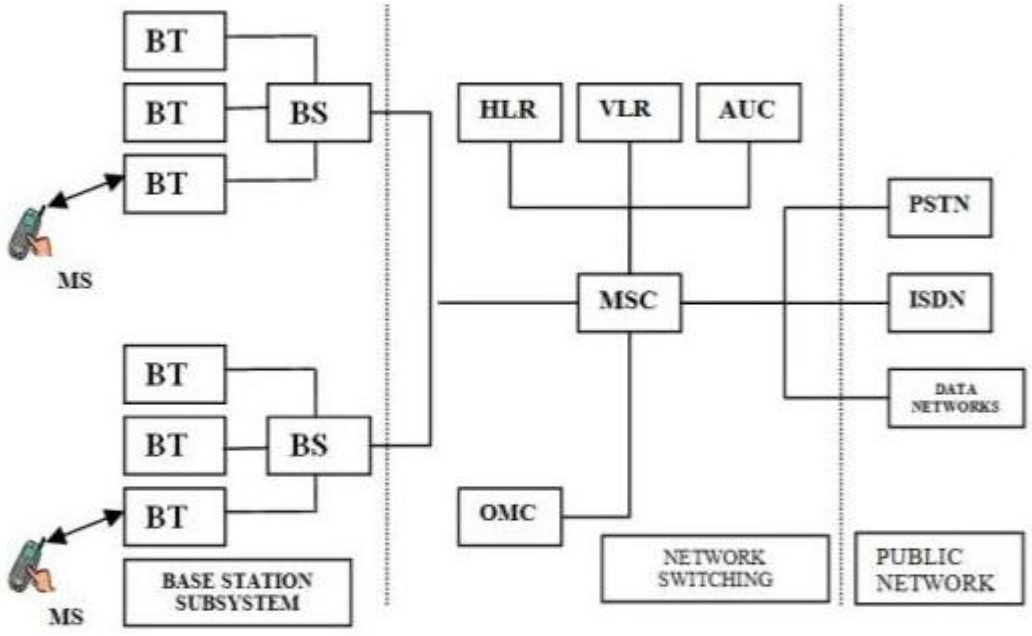
Signaling Issues

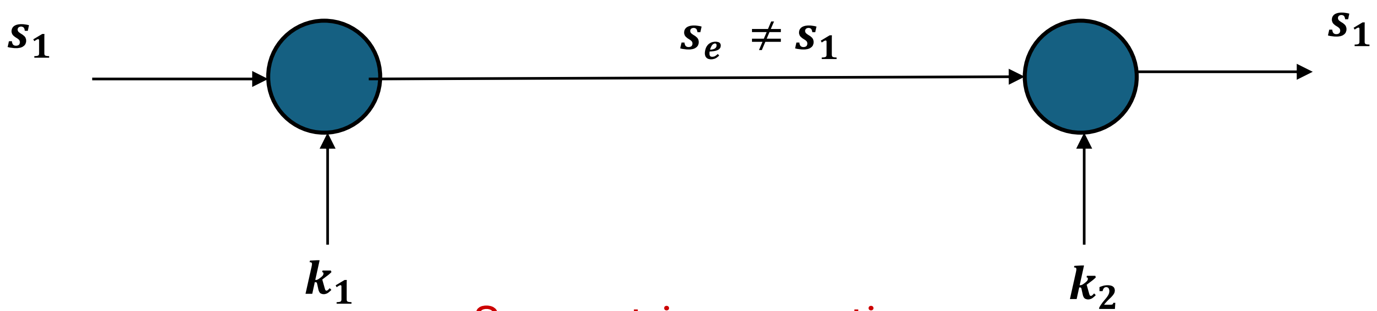
- Tel Exch B, I want Tel 1n
- Is there free link to 1n?
- Notify Tel 1 that Tel 1n is ringing
- Notify Exch. A1 that Tel 1n has disconnected.

3. Uses modern computer technology
4. Integration of transmission and switching
5. Signal Regeneration

RECAP: ADVANTAGES OF DIGITAL COMMUNICATION NETWORKS /2

6. Advanced Performance Monitoring.
7. Ability to integrate other services.
8. Ability to operate at low Signal-to-Noise Ratio (SNR).
9. Ease of encryption.



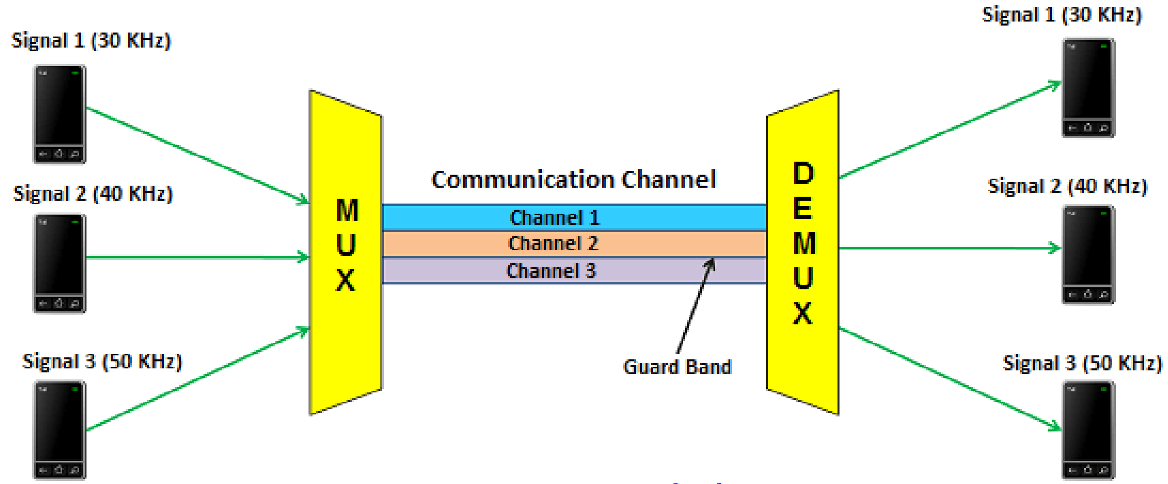


Symmetric encryption

$k_1 = k_2$

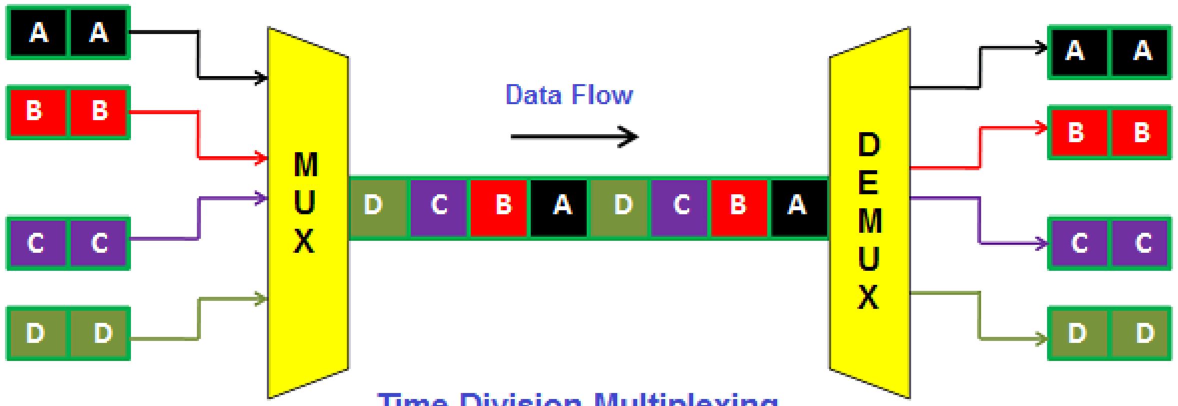
ANALOG Vs DIGITAL MULTIPLEXING

1. **A multiplexer combines (mux)** multiple signals into a single composite signal. The composite signal is transmitted over a shared medium, such as a fiber optic cable (WDM) or radio wave (FDM).
2. **A demultiplexer (demux)** separates the composite signal back into the original signals.
3. The Cost of digital multiplex systems, e.g. TDM is much lower that that of analogue multiplex systems, e.g. FDM.



Frequency Division Multiplexing

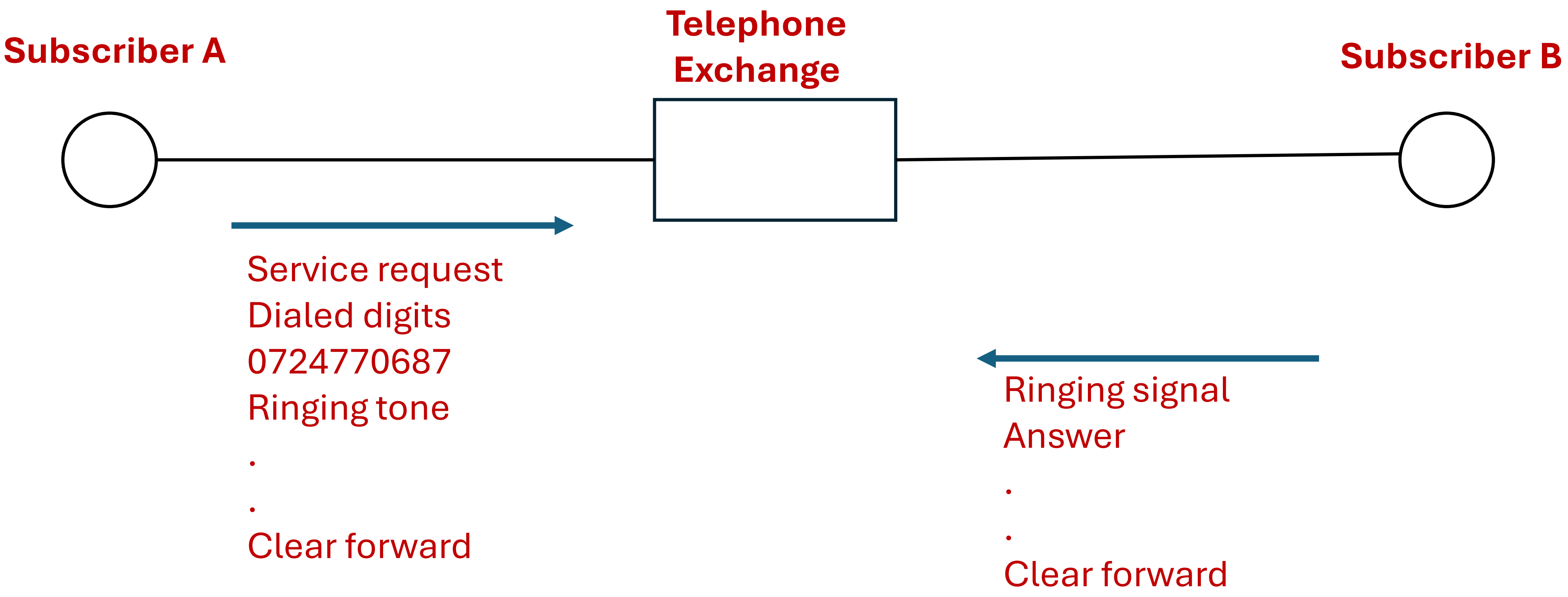
Physics and Radio-Electronics



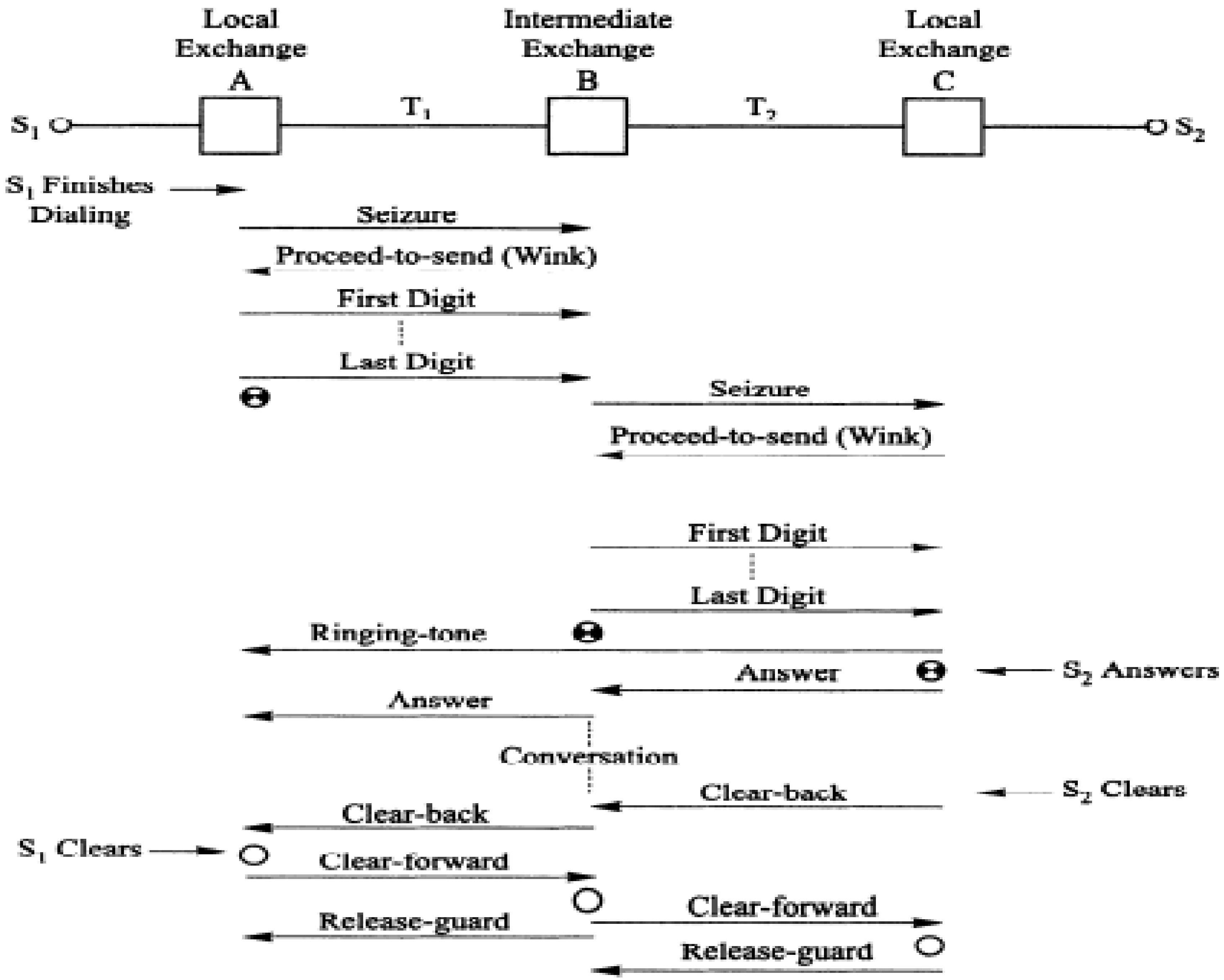
Time Division Multiplexing

SIGNALLING IN ANALOG TELECOMMUNICATION SYSTEM - LOCAL

1. **Signaling in telecommunications** is the process of using signals to control and manage communication networks.
2. **Telecommunication signals** include: dialed digits, status of the call e.g. telephone ringing, set-up and tear-down information.

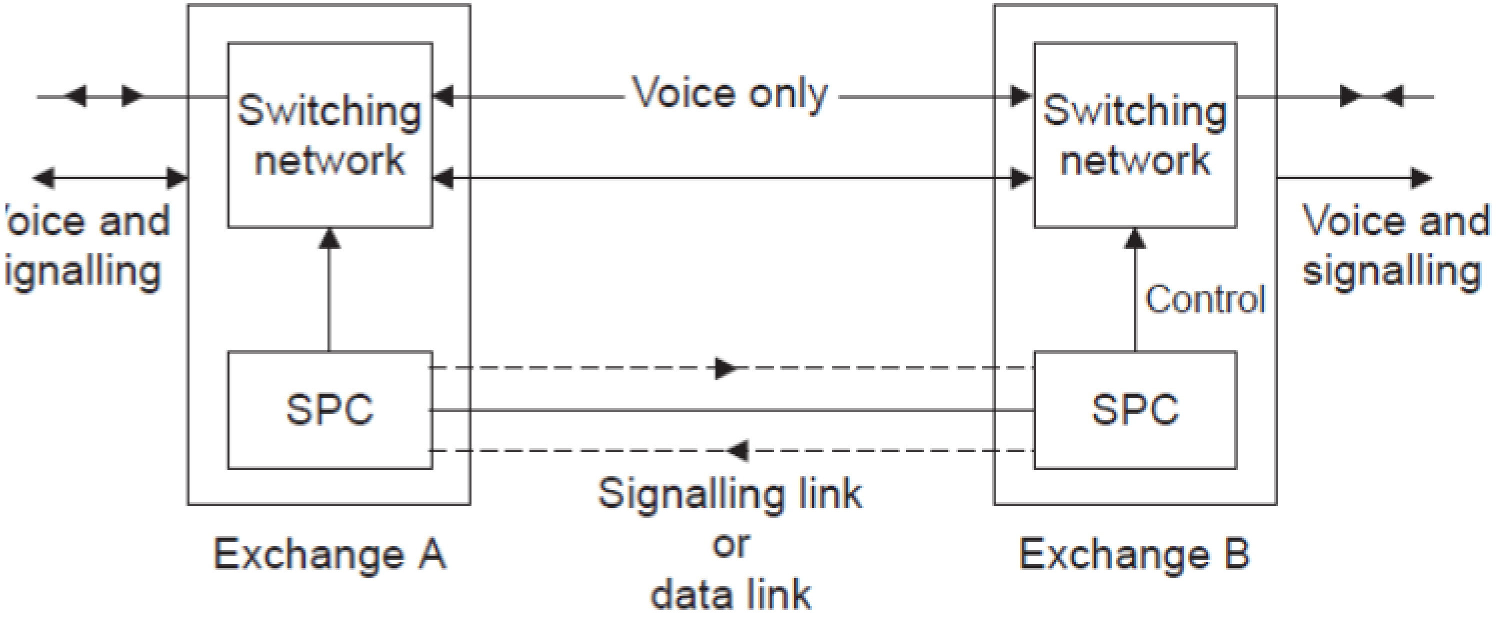


SIGNALLING IN ANALOG SYSTEM WITH TRANSIT EXCHANGE



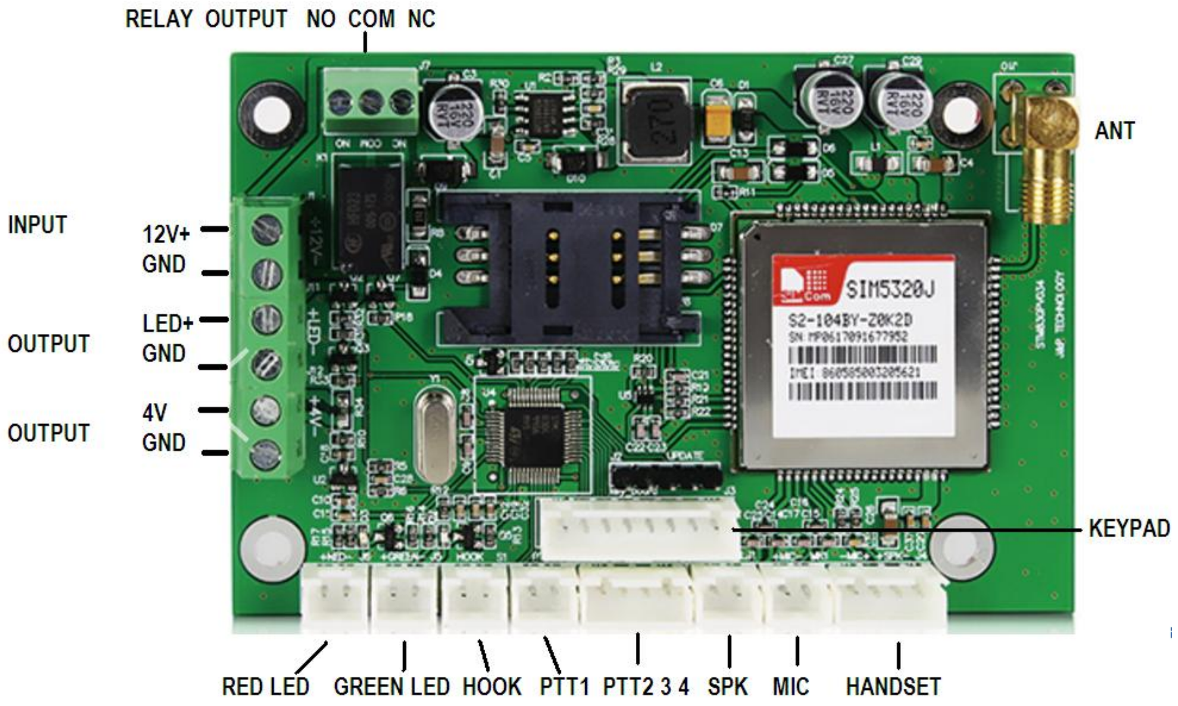
SIGNALLING IN DIGITAL SYSTEMS

1. **Digital Signaling systems** allow control information to be inserted into and extracted from a message stream independent of the mode of transmission.
2. **Signaling equipment** is designed separate from transmission systems allowing control functions and formats to be designed and modified independently.



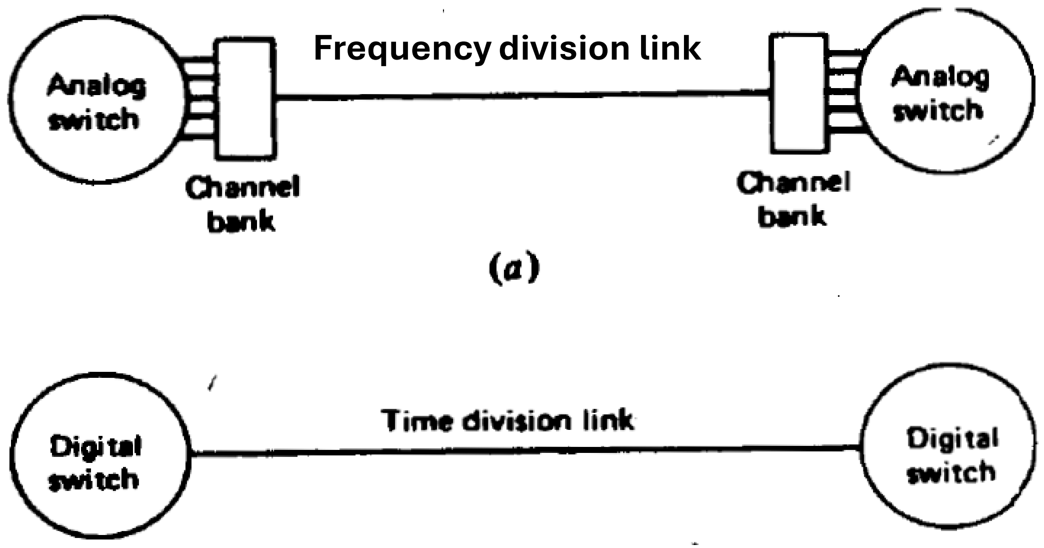
USE OF MODERN TECHNOLOGY-LSI & VLSI

1. **Multiplexer and switching matrix** for digital systems are implemented with the same basic circuits used in computers.
2. **Special LSI Circuits** have been developed specifically for telecommunication functions e.g Voice Codecs, Multiplexing, DSPs, etc.
3. **Low-cost of digital circuitry** allow for implementations that would be very expensive if developed on analogue platforms, e.g large non-blocking exchanges.
4. **Digital technology** provides easier and cheaper interfaces to fibre-optic cable systems.



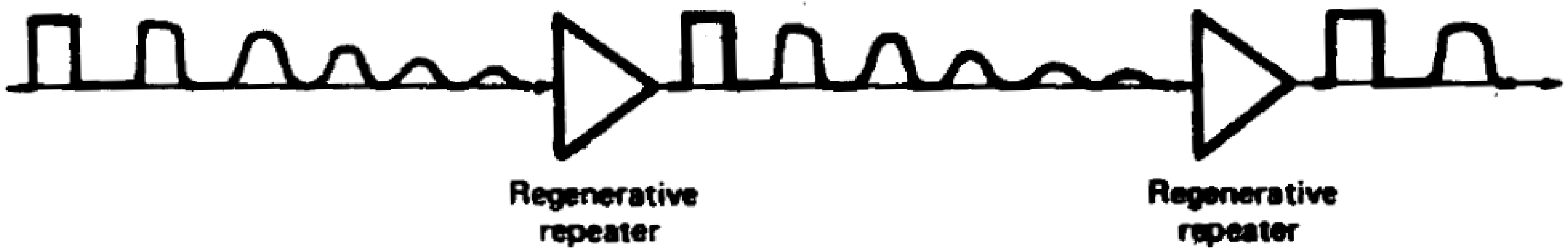
INTEGRATION OF TRANSMISSION AND SWITCHING`

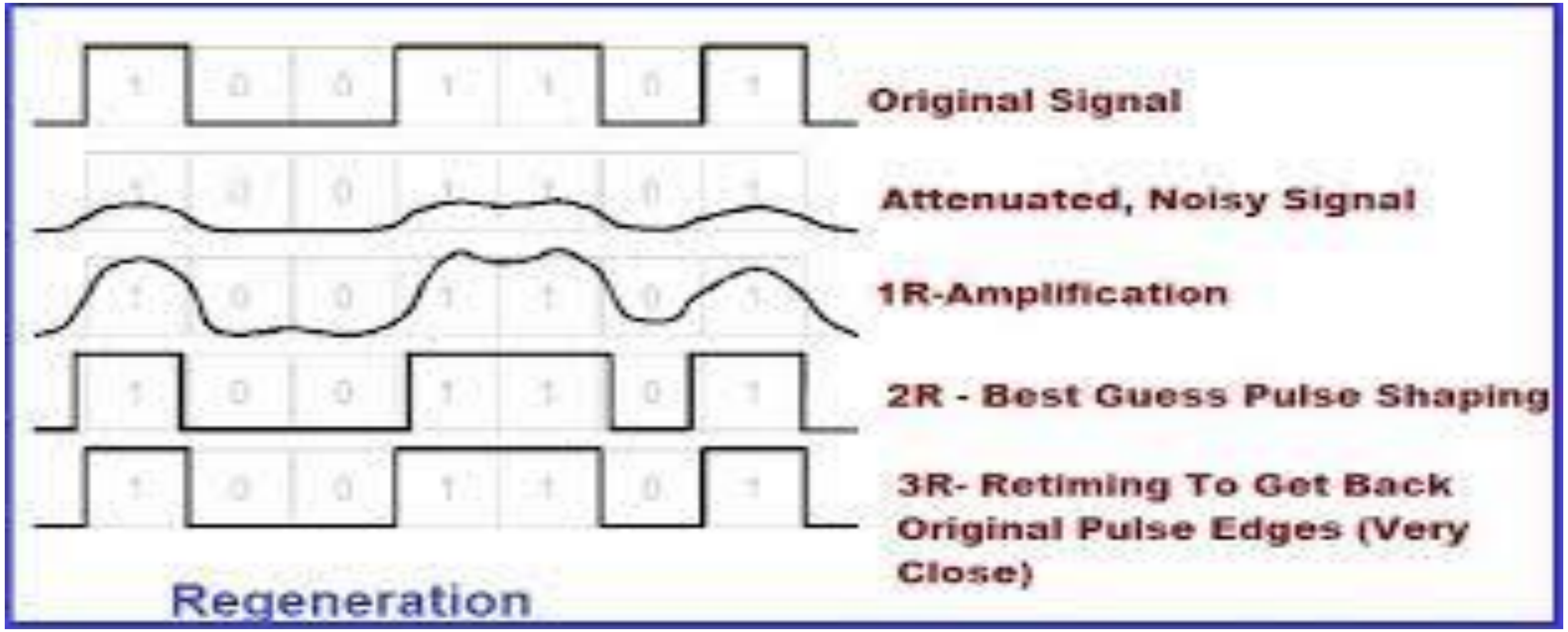
1. **Digital communication** allows data to be terminated directly onto telephone switches unlike analog systems which require modulation and demodulation in separate channel banks.
2. **Cable entrance requirements** and distribution of wire pairs is greatly reduced because all trunks are implemented as sub-channels in a TDM signal.



SIGNAL REGENERATION

Digital Signals can be regenerated at suitable intervals unlike analogue signals.





BENEFITS OF FULLY INTEGRATED DIGITAL COMMUNICATION NETWORKS

1. **Long-distance and local voice quality** are identical in terms of noise, signal level, and distortion.
   a) **At physical layer**, repeaters regenerate clean pulses from distorted ones.
   b) **At data link layer**, error detection and correction are used.

2. Since digital baseband circuits are inherently four-wire, **network-generated echoes** are eliminated, and true full-duplex, four-wire digital circuits are available.

RECAP: OPEN SYSTEMS INTERCONNECT MODEL

1. **Open Systems Interconnection (OSI) model** is a conceptual framework that describes how data travels across a network.
2. **OSI** was created by the International Standards (ISO) and published in 1984.

OSI Layer	Purpose
Application	Application Program
Presentation	Data Interpretation
Session	Remote Actions
Transport	End-to-End Reliability
Network	Destination Addressing
Data Link	Media Access & Framing
Physical	Electrical Interconnect

ENCRIPTION: SUBSTITUTION & TRANSPOSITION CIPHERS

Substitution ciphers replace each group of letters in the message with another group of letters to disguise it

plaintext:	a	b	c	d	e	f	g	h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z
ciphertext:	Q	W	E	R	T	Y	U	I	O	P	A	S	D	F	G	H	J	K	L	Z	X	C	V	B	N	M

Simple single-letter substitution cipher

Transposition ciphers reorder letters to disguise them

M	E	G	A	B	U	C	K
7	4	5	1	2	8	3	6
p	l	e	a	s	e	t	r
a	n	s	f	e	r	o	n
e	m	i	l	l	i	o	n
d	o	l	l	a	r	s	t
o	m	y	s	w	i	s	s
b	a	n	k	a	c	c	o
u	n	t	s	i	x	t	w
o	t	w	o	a	b	c	d

← Key gives column order

Plaintext

pleasetransferonemilliondollarsto
myswissbankaccountsixtwotwo

Ciphertext

AFLLSKSOSELAWAIATOOSSCTCLNMOMANT
$\underbrace{\text{ESILYNTW}}_{\text{Column 5}}\underbrace{\text{RNNTSOWD}}_{6}\underbrace{\text{PAEDOBUO}}_{7}\underbrace{\text{ERIRICXB}}_{8}$

Simple column transposition cipher

DIGITIZATION OF SPEECH

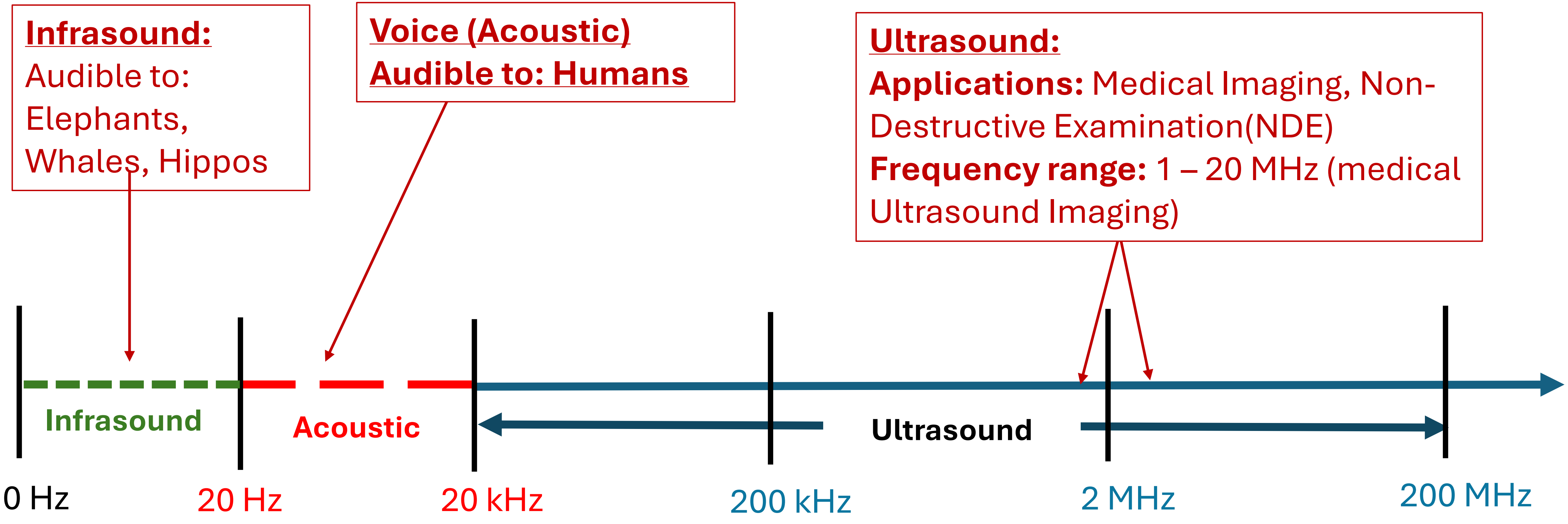
EEEN 464 – DIGITAL COMMUNICATION

Friday, February 6, 2026

HUMAN HEARING & VOICING FREQUENCY RANGE

1. **Human ear** can detect sounds with frequencies between 20 Hz and 20,000 Hz.
2. Sound with frequencies below the human hearing range is referred to as **infrasound**.
3. Sound with frequencies above is called **ultrasound**.
4. Humans produce two types of sounds:
   a) **Voiced sounds** are made when the vocal folds vibrate.
   b) **Voiceless** sounds are made when the vocal folds do not vibrate.

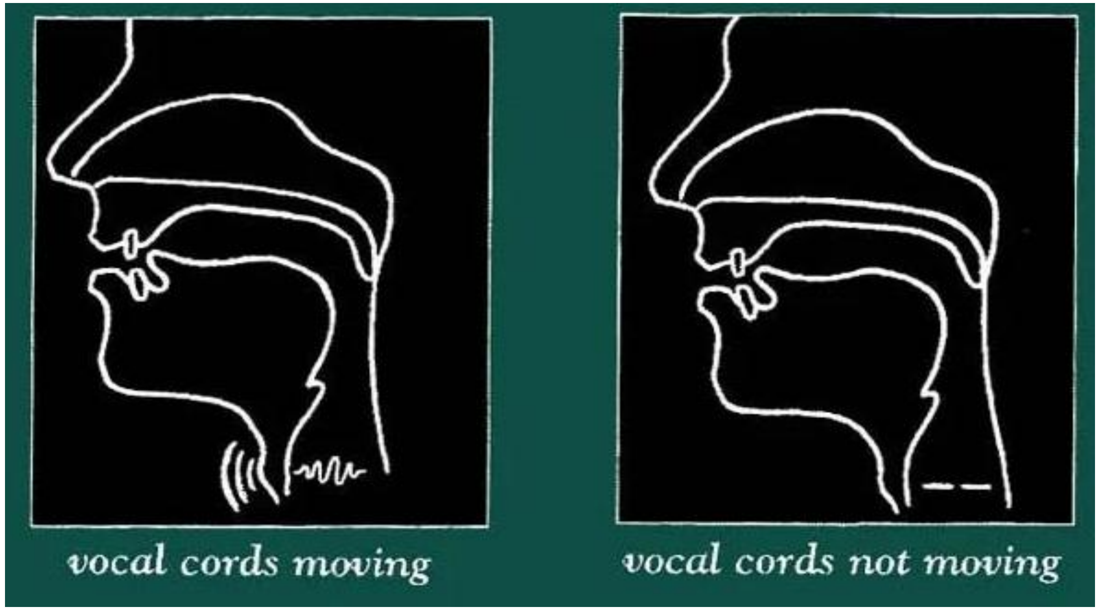
SOUND FREQUENCY SPECTRUM



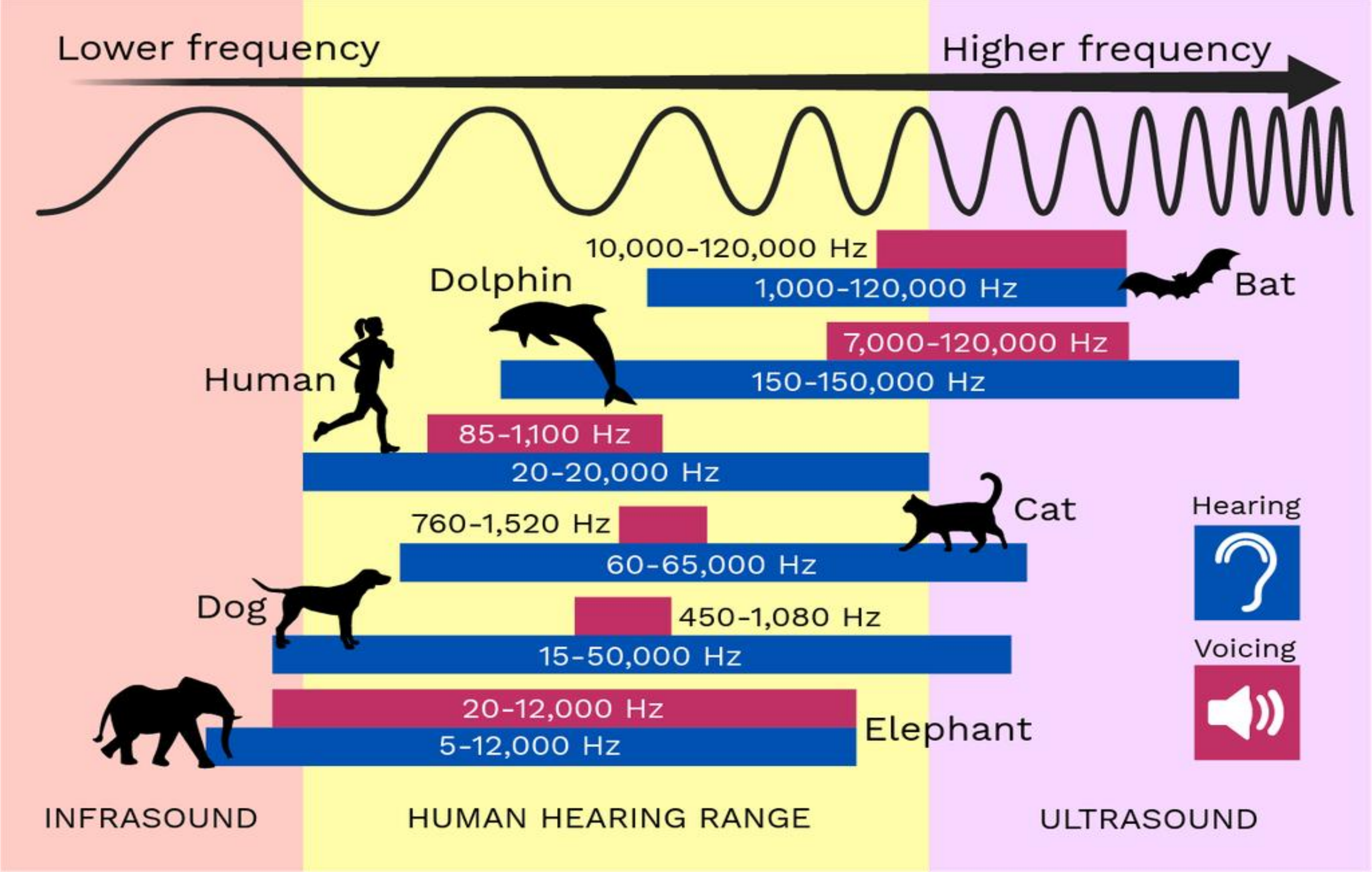
1. **Fundamental voicing frequency in humans** varies from 85 Hz to 1,100 Hz. Women's voices are generally higher in pitch than men's voices.
2. **Harmonic series of a voice** is the series of frequency components above the fundamental frequency.

VOICED Vs UNVOICED SOUNDS

1. **Voiced sounds** produced with the vocal cords vibrating.
   a) **Vowels:** Sounds **(a, e, i, o, u)**
      Frequency: 250 – 2,000 Hz
   a) **Consonants** e.g. **b, d, g, m, n, l, r, v, z**
      Frequency: 250 – 4,000 Hz
2. **Voiceless sounds**
   a) Consonants, e.g. **p, t, k, f, s, sh, ch**
      Frequency: 2,000 – 8,000 Hz

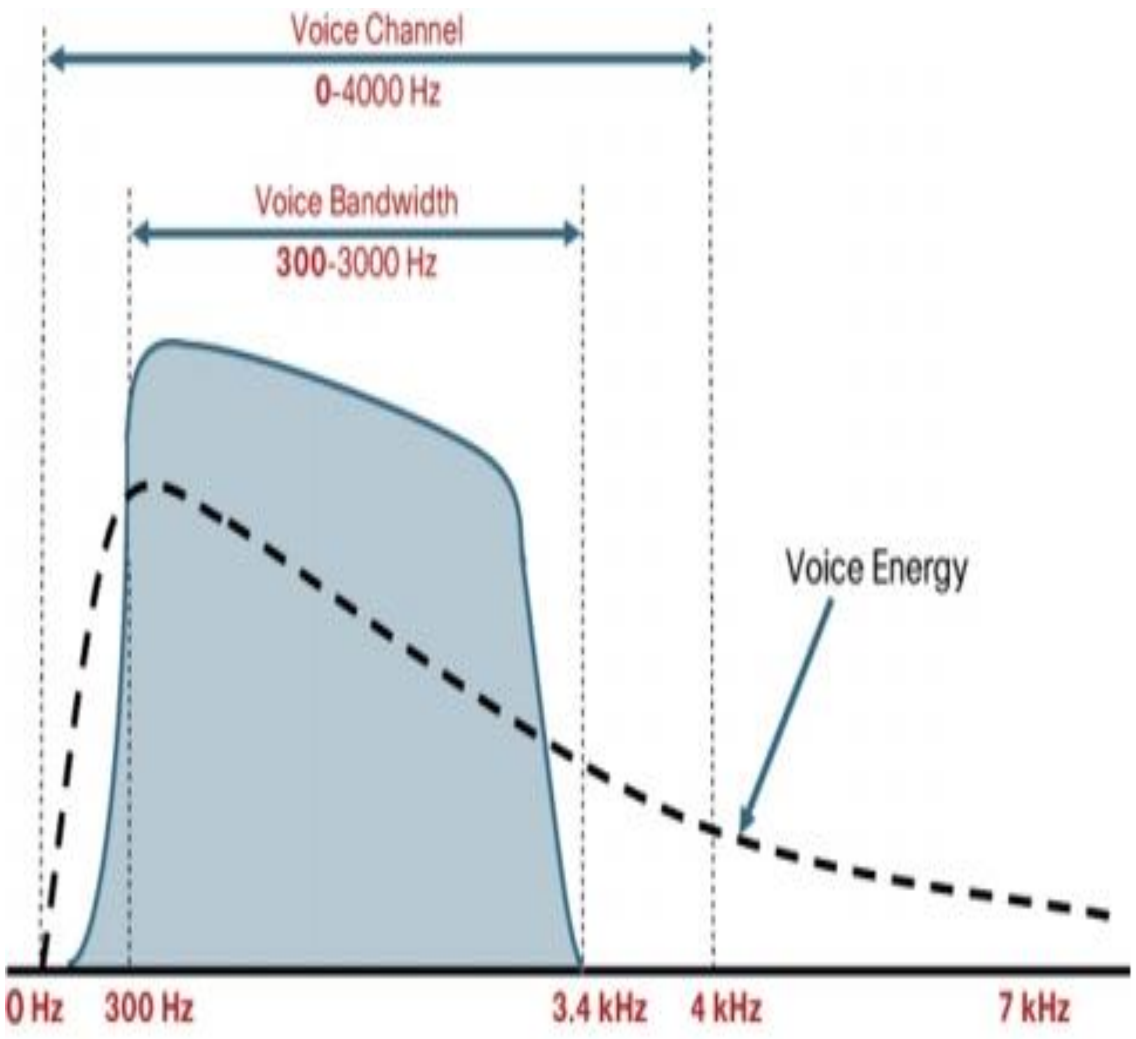


HEARING & VOICING AMONGST DIFFERENT ANIMALS



TELEPHONE FREQUENCY RANGE

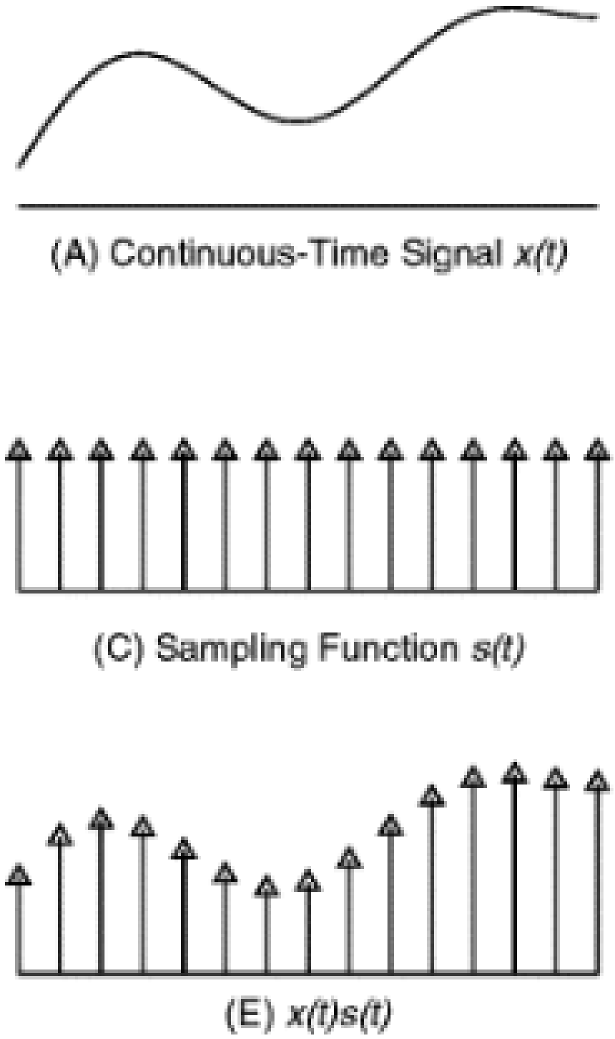
1. **The frequency range for a standard telephone** is 300–3,400 hertz (Hz). This range is known as the voiceband in telephony.
2. **The voiceband** was chosen because:
   1. **Acceptable level of intelligibility** is obtained by transmitting voice in range 0.3 -3.4 KHz
   2. **Most of the voice energy** is concentrated in this band.
   3. **Wideband audio (HD Voice)** has frequency range to 50 Hz to 7,000 Hz.



SAMPLING THEORY

Sampling theorem can be stated by any of the following:

1. **A signal that is band-limited of finite energy** at a frequency f_m can be completely described by samples taken at a uniform time intervals of no less than $\frac{1}{2f_m}$ apart.
2. **A band-limited signal of finite energy with no frequency above f_m** may be adequately recovered from samples taken at the rate $2f_m$ samples per second.

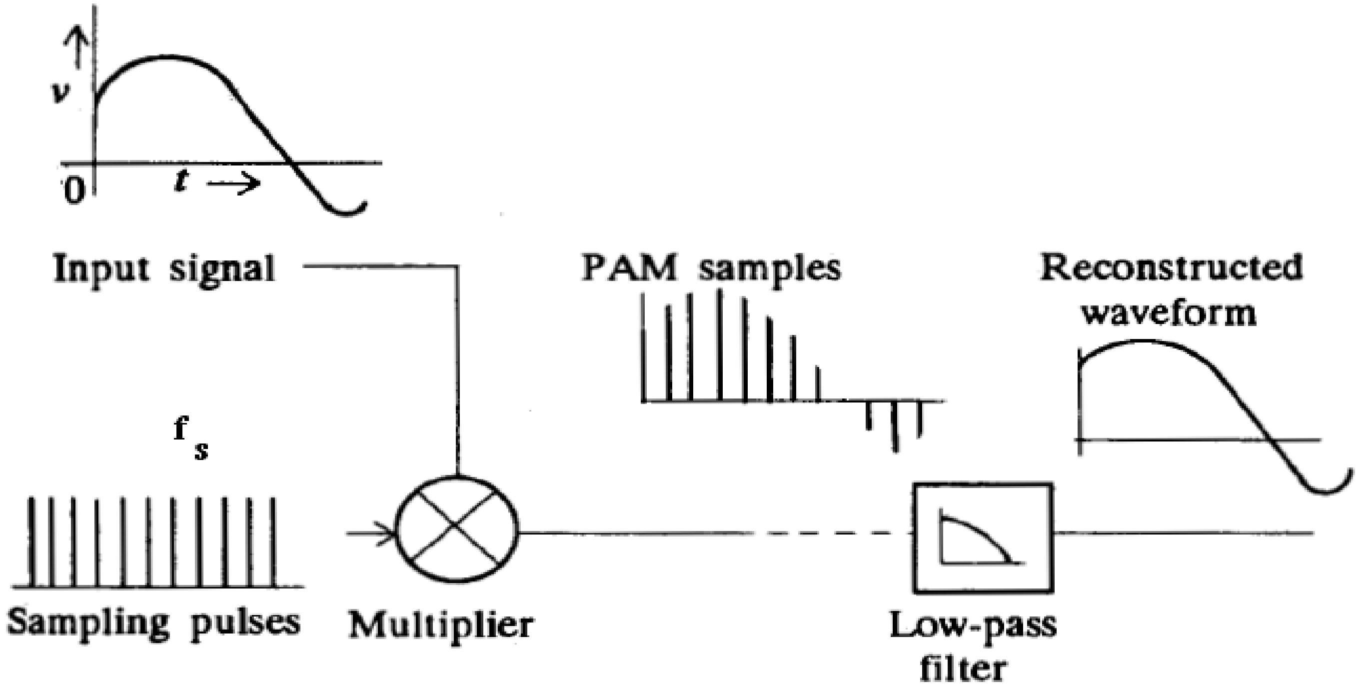


MATCHING THEORY TO SAMPLING REQUIREMENTS

REQUIREMENT	SAMPLING THEORY
1. There should be sufficient number of samples so that the original signal is adequately represented	A signal that is band-limited of finite energy at a frequency f_m can be completely described by samples taken at a uniform time intervals of no less than $\frac{1}{2f_m}$ apart.
2. It should be possible to reconstruct the original signal from its samples.	A band-limited signal of finite energy with no frequency above f_m may be adequately recovered from samples taken at the rate $2f_m$ samples per second.

PULSE-AMPLITUDE MODULATION

Pulse-amplitude modulation (PAM) is a form of signal modulation where the message information is encoded in the amplitude of a series of signal samples.

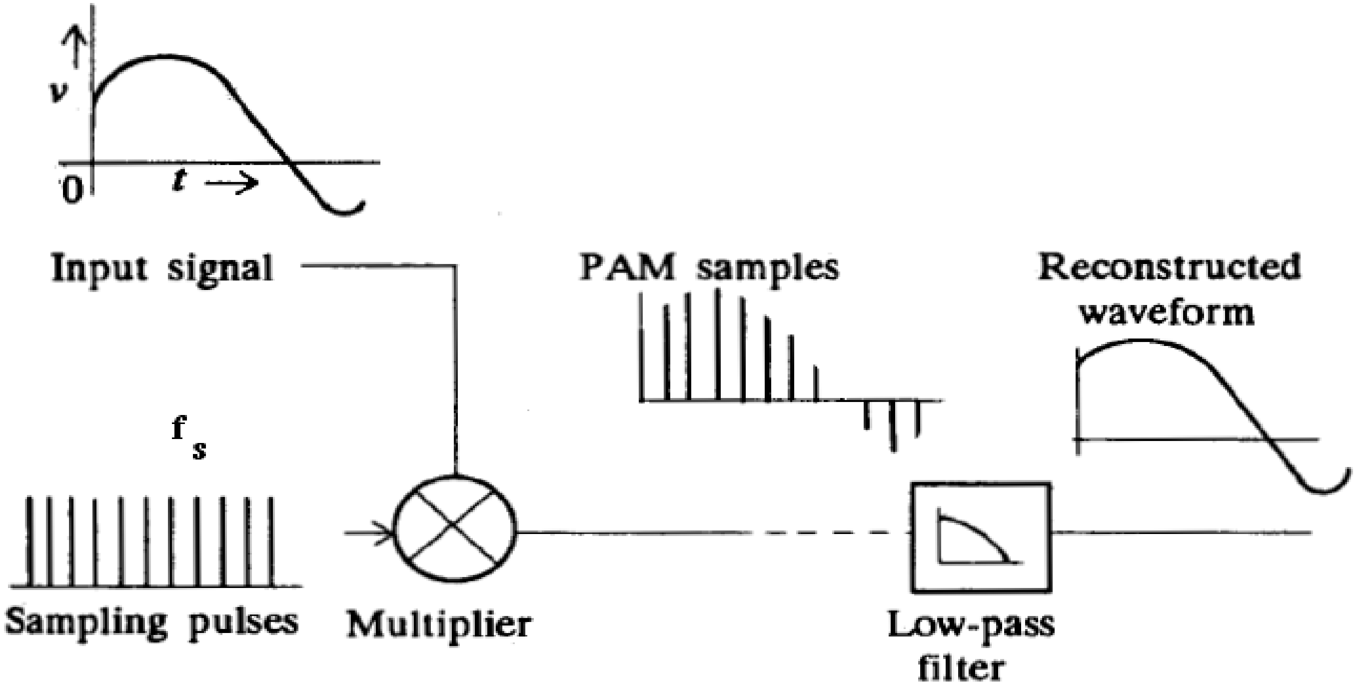


Nyquist Criterion/Theorem

- $f_s > 2f_{max}$ where f_{max} is the highest frequency in the analog input signal

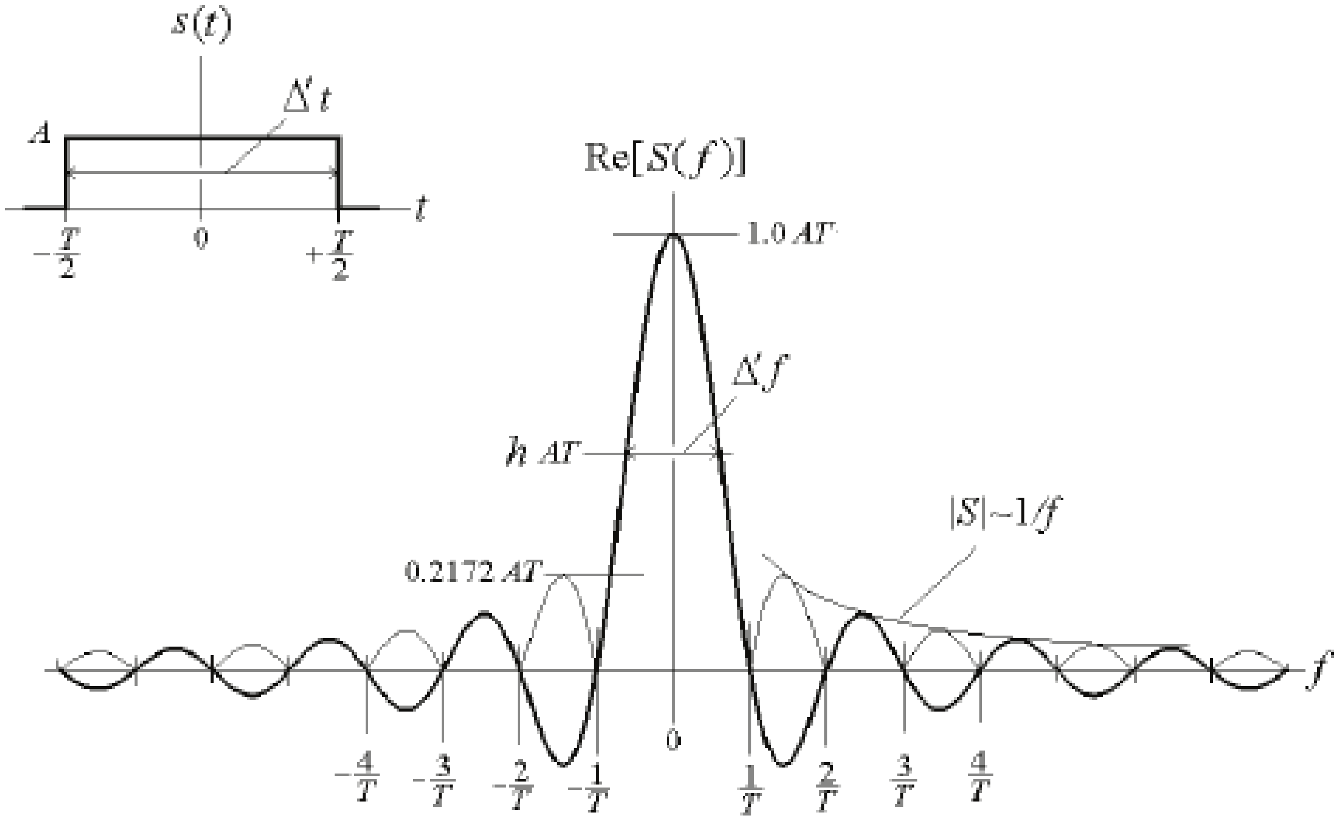
WHY ARE WE INTERESTED IN FOURIER TRANSFORM OF PAM SIGNALS?

1. **Fourier transform** of a signal represents the signal in the frequency domain.
2. **The Fourier transform of the PAM signals** assists us to design the parameters of the low-pass reconstruction filter.

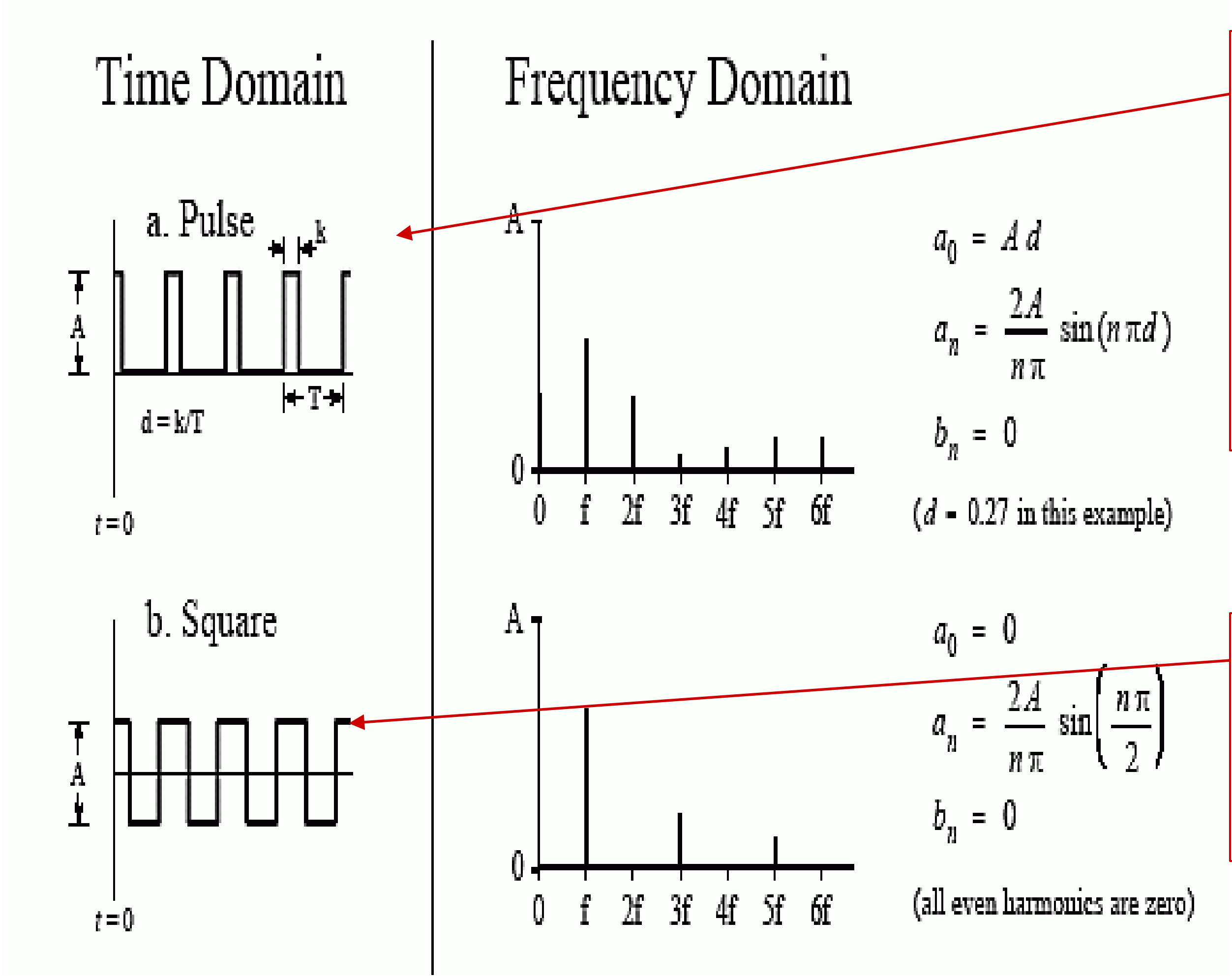


RECAP: FOURIER TRANSFORM OF A PULSE

1. **The spectrum of a single pulse** has a sin x/x shape and covers a band of frequencies.
2. **The width of the central lobe** of the spectrum of a single pulse is inversely proportional to the pulse width.



RECAP: FOURIER TRANSFORM OF A PULSE TRAIN



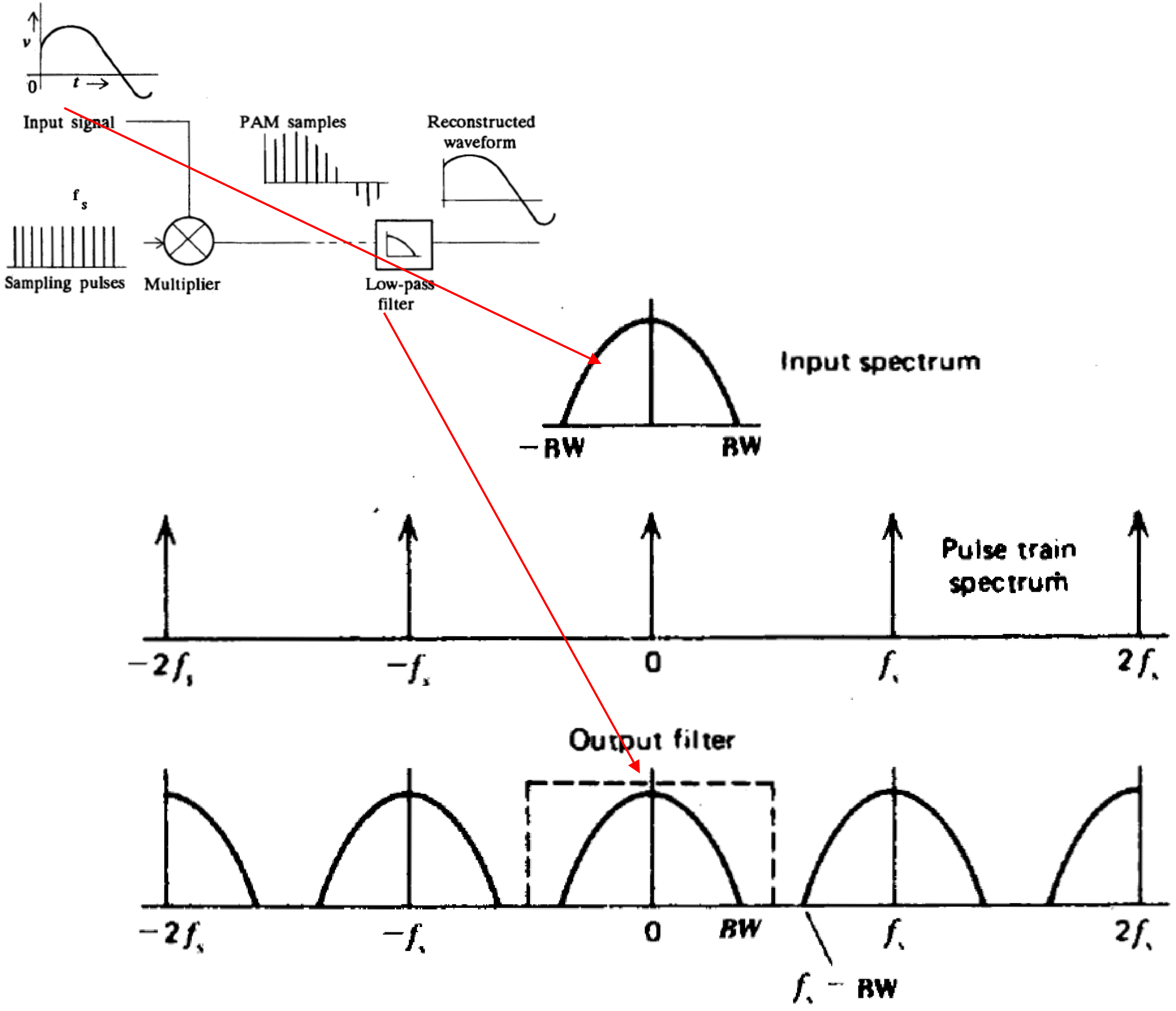
Duty cycle of 0.27: The spectral content at closest to 3f is quite small.

- At a duty-cycle of exactly one-third, the spectral content at 3f would be zero.

Duty cycle is 0.5: The spectral content at 2f (and 4f and 6f etc..) is always zero.

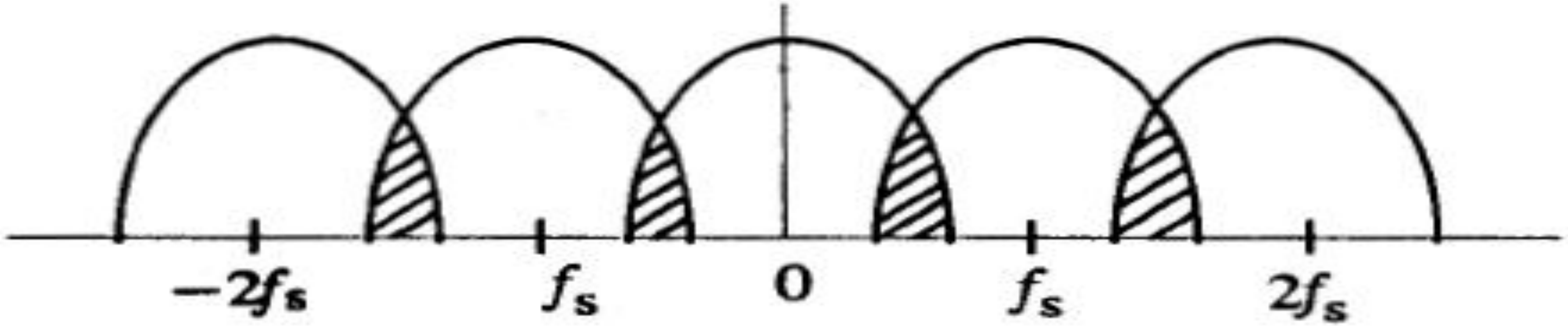
PULSE AMPLITUDE MODULATION (PAM) SPECTRUM

1. **Low-pass filter** is used to demodulate PAM signals by passing the baseband and removing higher frequencies.
2. **Cutoff frequency** of the low-pass filter should be large enough to accommodate the highest frequency component of the message signal.
3. **Reconstructed signal** may have some amplitude distortion due to the aperture effect.



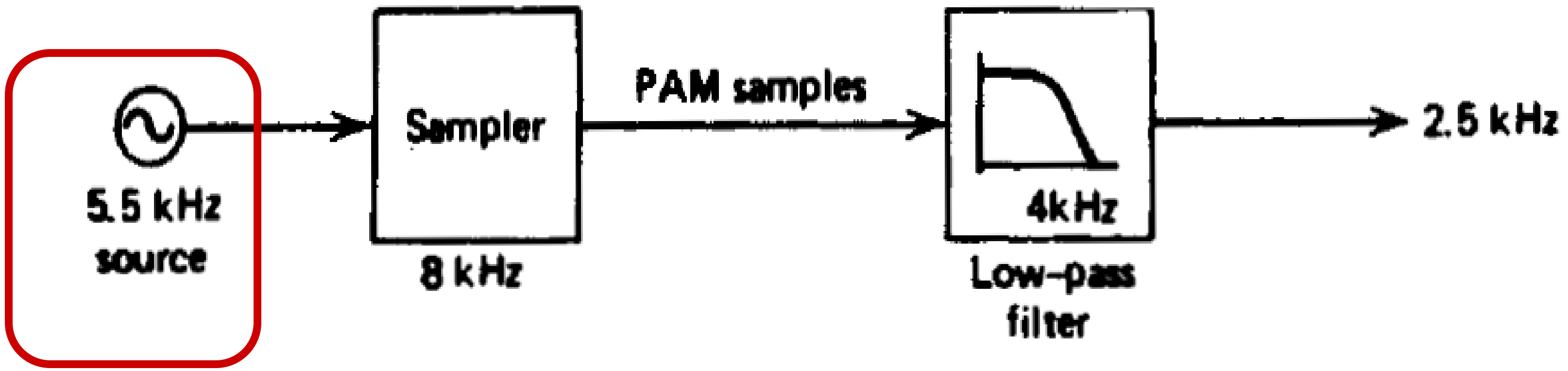
ALIASING/FOLD-OVER DISTORTION

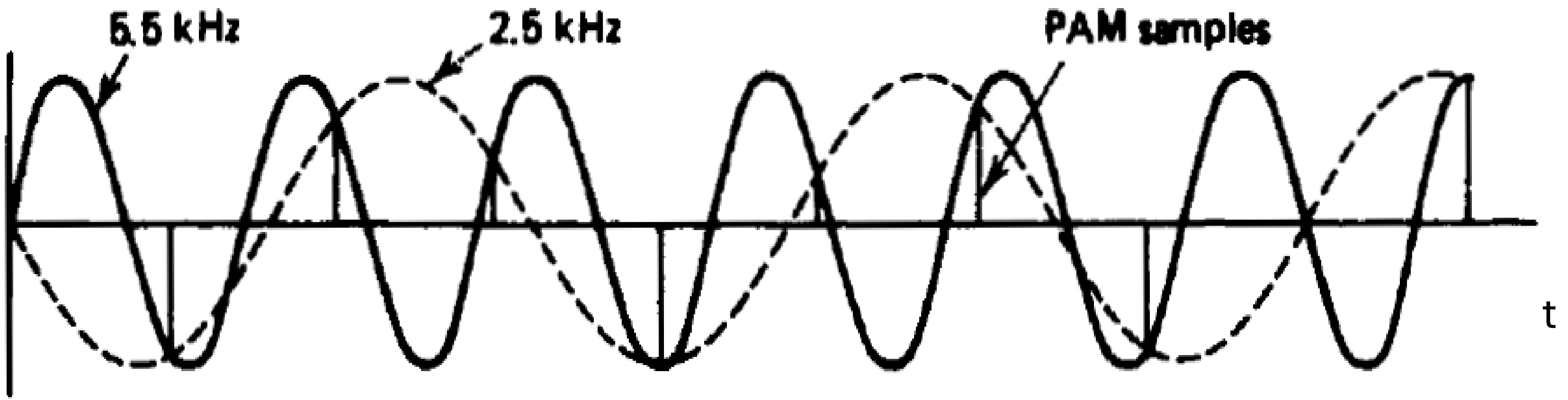
1. **Aliasing/fold-over** occurs when the sampling frequency f_s is less than the Nyquist frequency, $2f_{max}$ resulting in an overlap of the spectrum.



2. Aliasing/fold-over distortion is avoided in traditional telephony by:
   a) Band-limiting the signal to the range 0.3-3.4 KHz.
   b) over-sampling at 8KHz instead of 6.8 KHz which is the Nyquist rate.

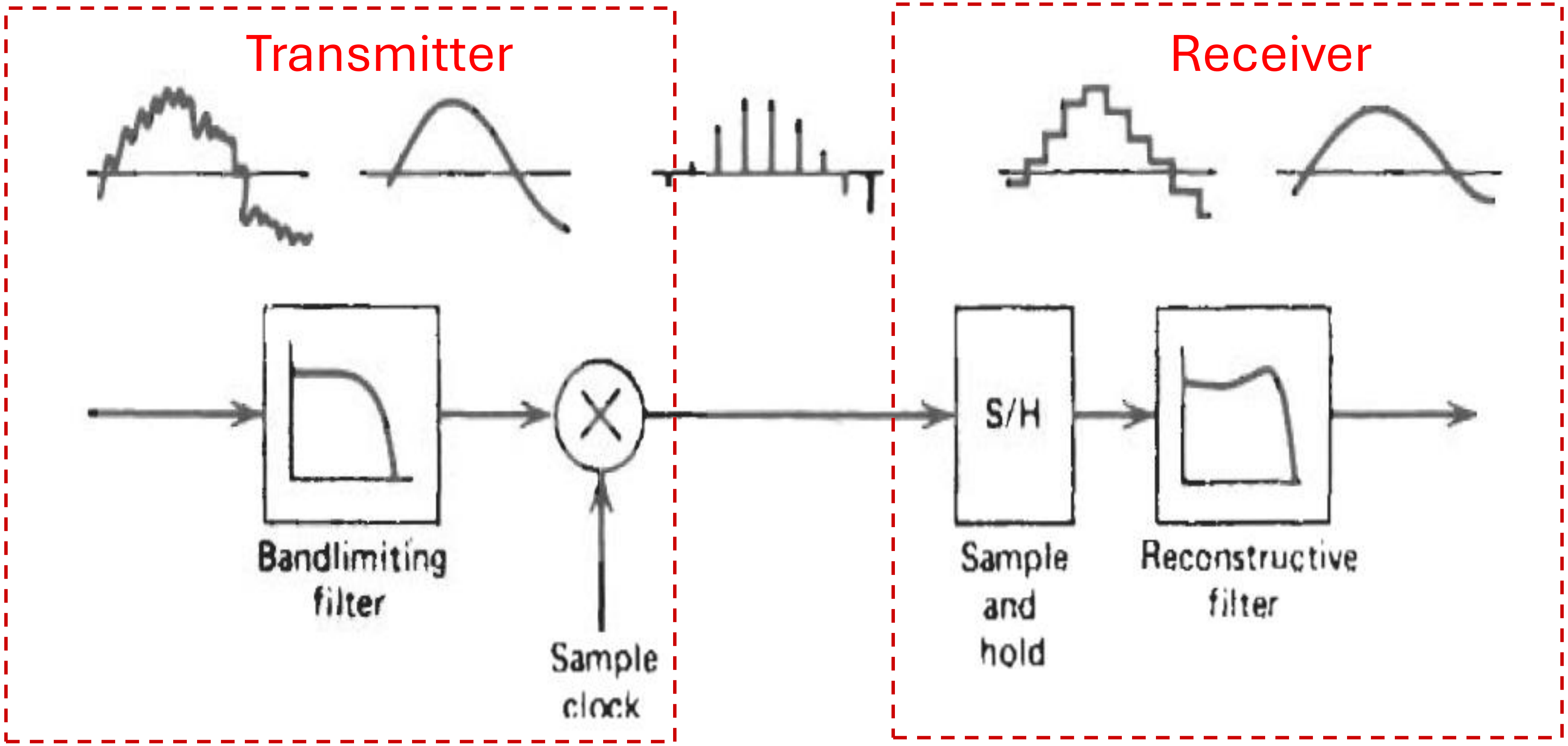
EXAMPLE OF ALIASING DISTORTION





As shown above, a signal of 5.5 KHz sampled at 8 KHz appears to be a 2.5KHz signal (dotted).

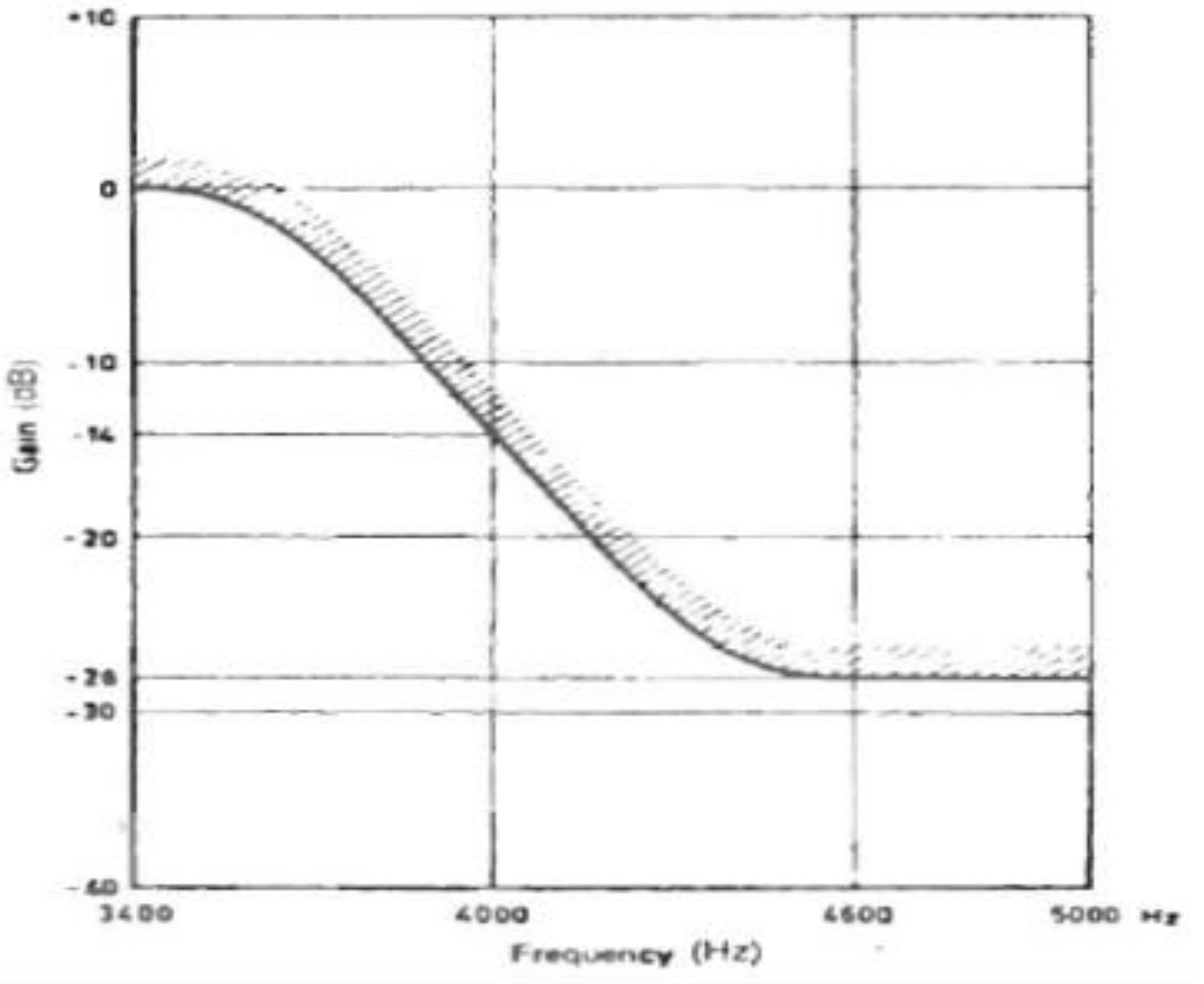
END-TO-END PAM COMMUNICATION SYSTEM



The response of the reconstructive filter is usually modified to account for the spectrum of the wider staircase samples.

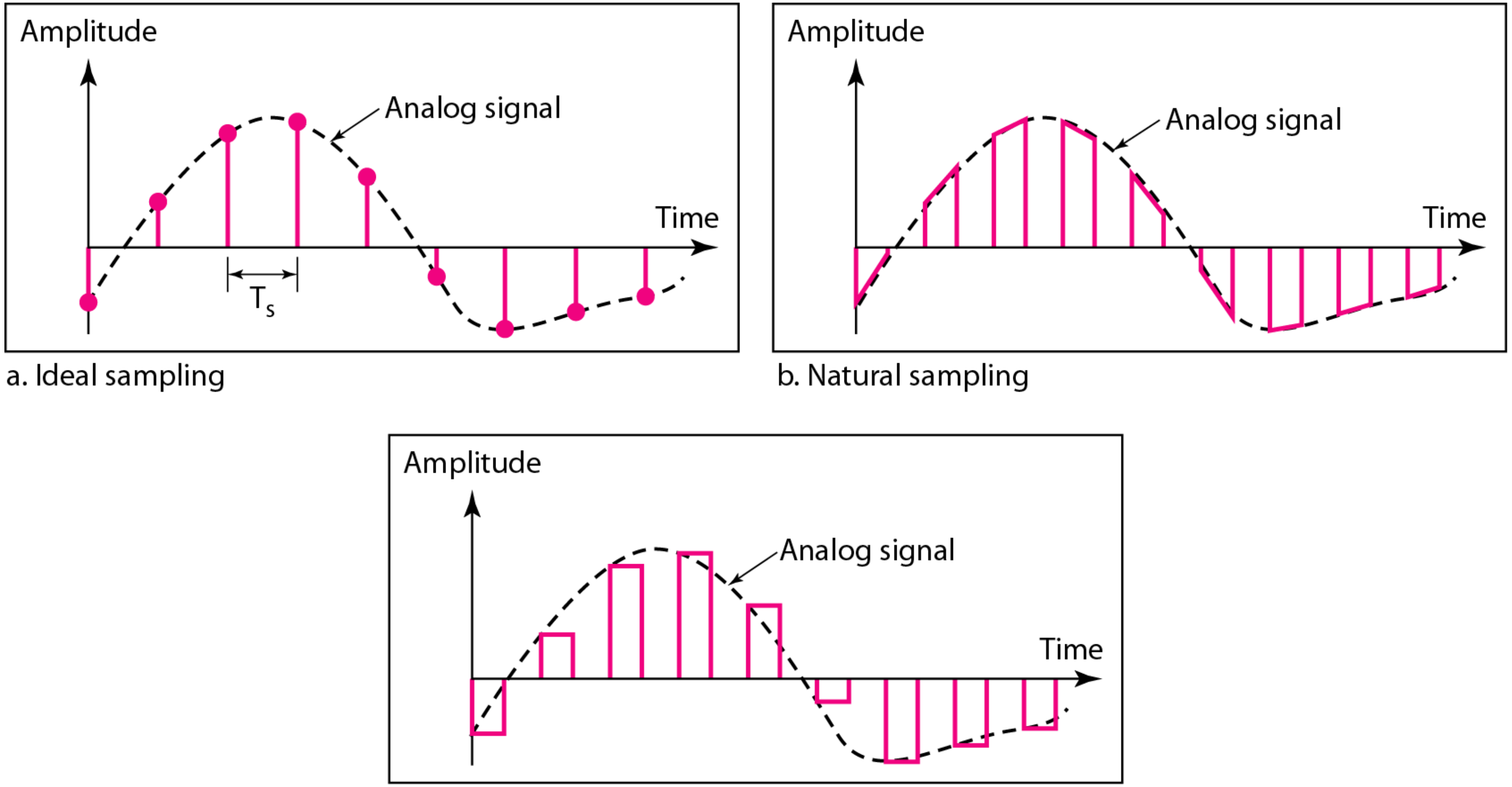
BAND-LIMITING FILTER DESIGNED TO MEET INTERNATIONAL TELECOMMUNICATION UNION (ITU) RECOMMENDATION FOR PCM VOICE CODERS

1. **ITU Standard for telephone speech sampling** requires14 dB attenuation is provided at 4 KHz.
2. **This can be restated** as 'the signal strength in watts at 4 KHz is 4% of that at 3.4 KHz.



SAMPLING TECHNIQUES

There are three sampling techniques, i.e Ideal, **Natural** and **Flat-top** as shown below.



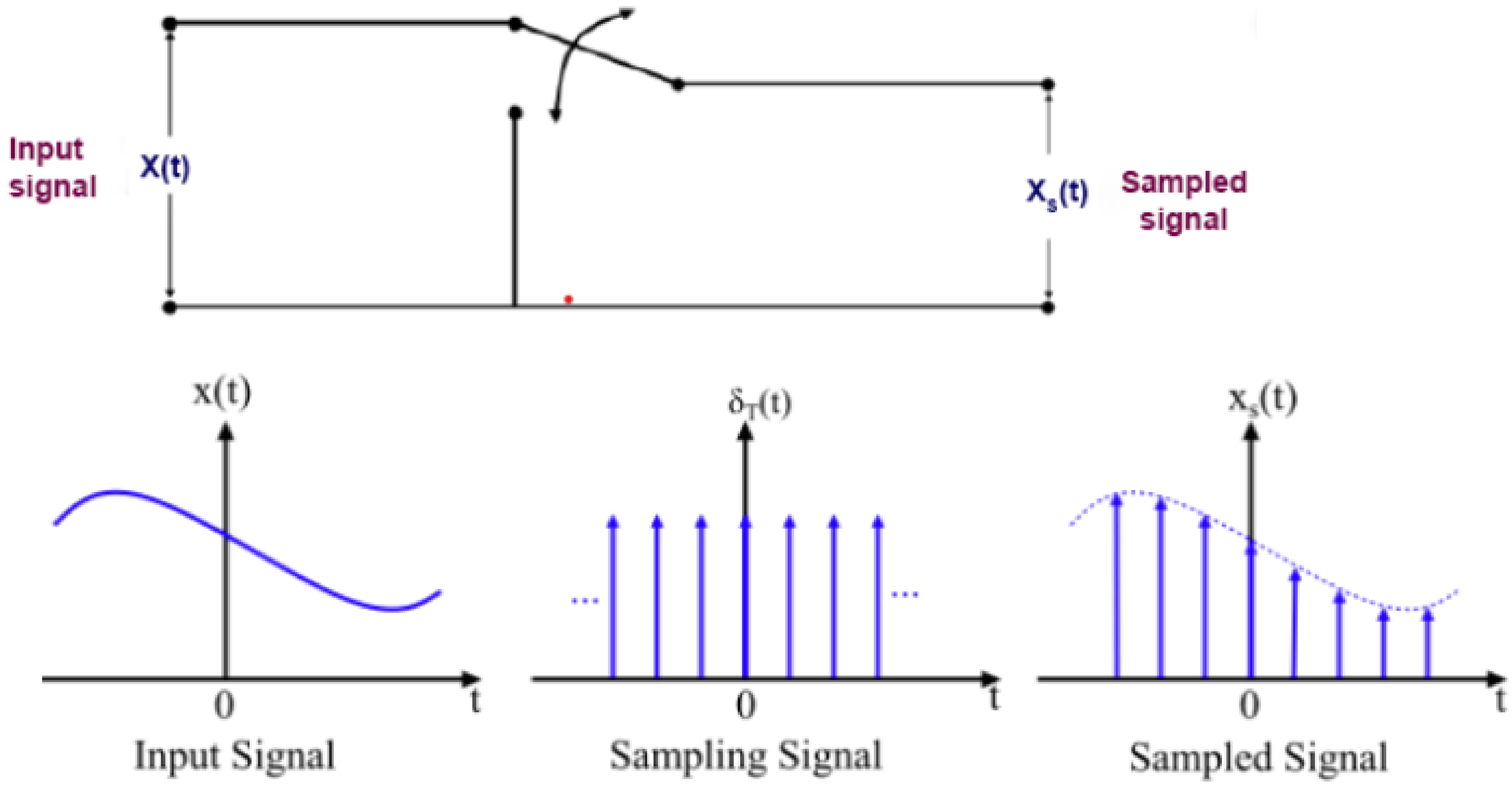
a. Ideal sampling

b. Natural sampling

c. Flat-top sampling

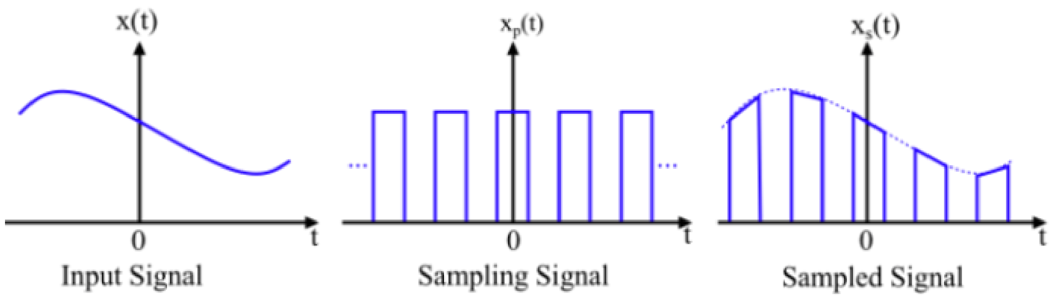
IDEAL SAMPLING

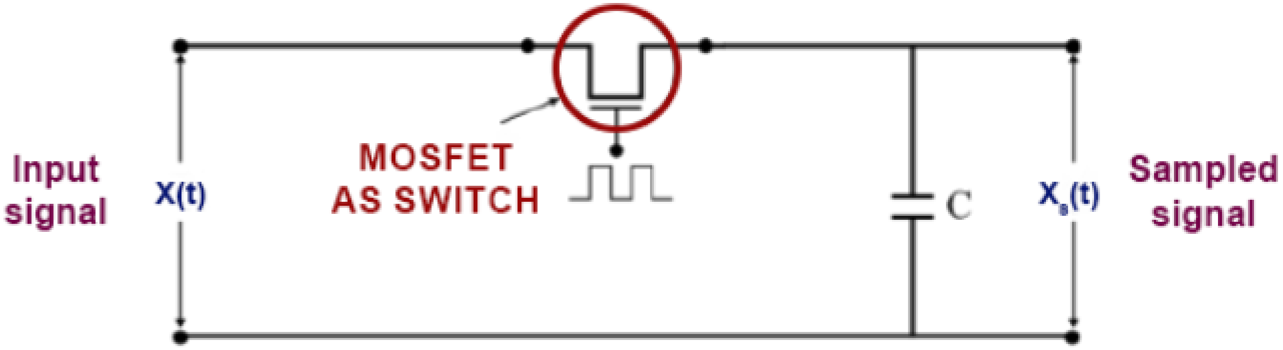
- **Ideal Sampling (Instantaneous sampling or Impulse Sampling)** uses a train of impulses and the principle used is known as multiplication principle.



NATURAL SAMPLING

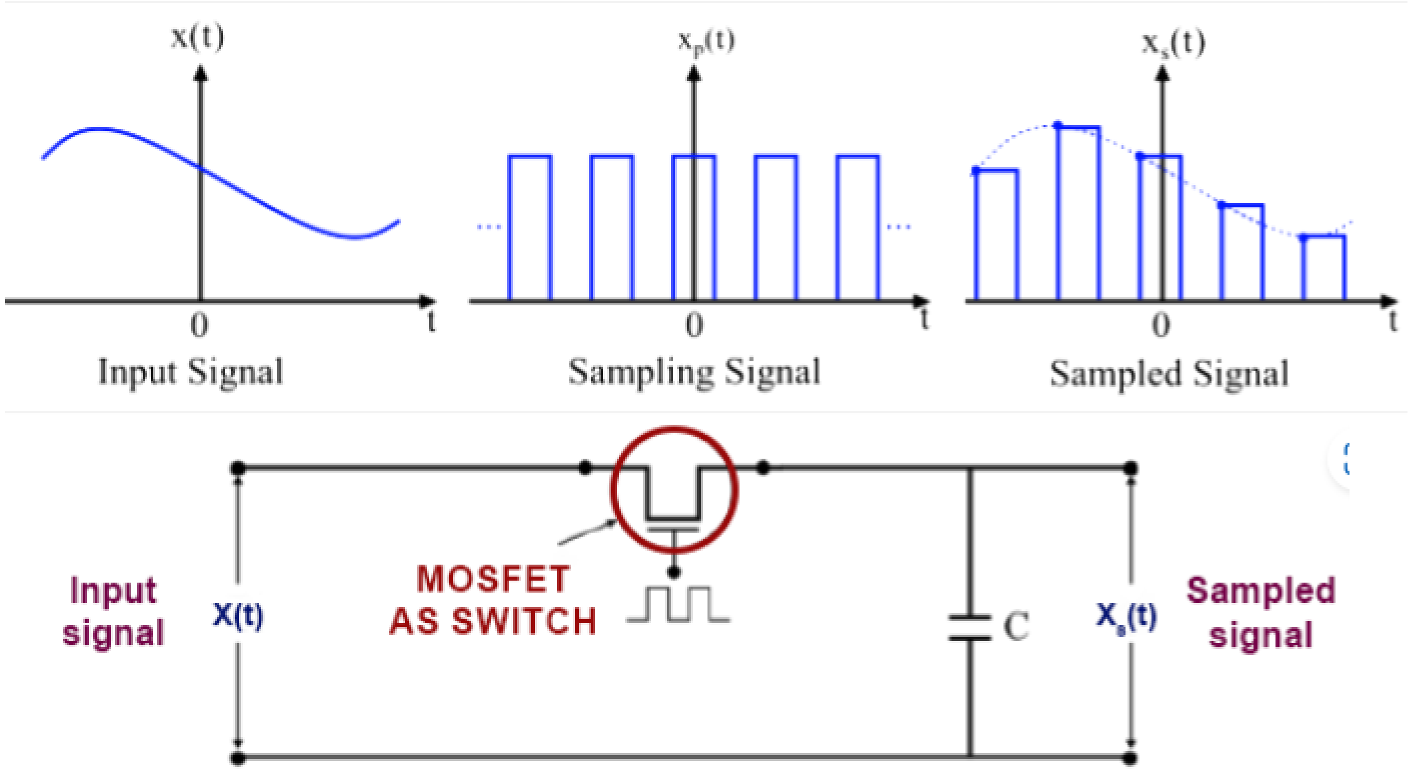
1. **Natural sampling** multiplies the original signal by a train of rectangular sampling pulses with unit amplitude
2. **The output signal** follows the original waveform's contours.
3. **Natural sampling** is usually realized using a MOSFET switch as shown.





FLAT-TOP SAMPLING

- **Flat-top sampling** is a type of natural sampling in which each sample is obtained by maintaining the value of the continuous signal constant for a set period of time, resulting in a flat-top waveform.



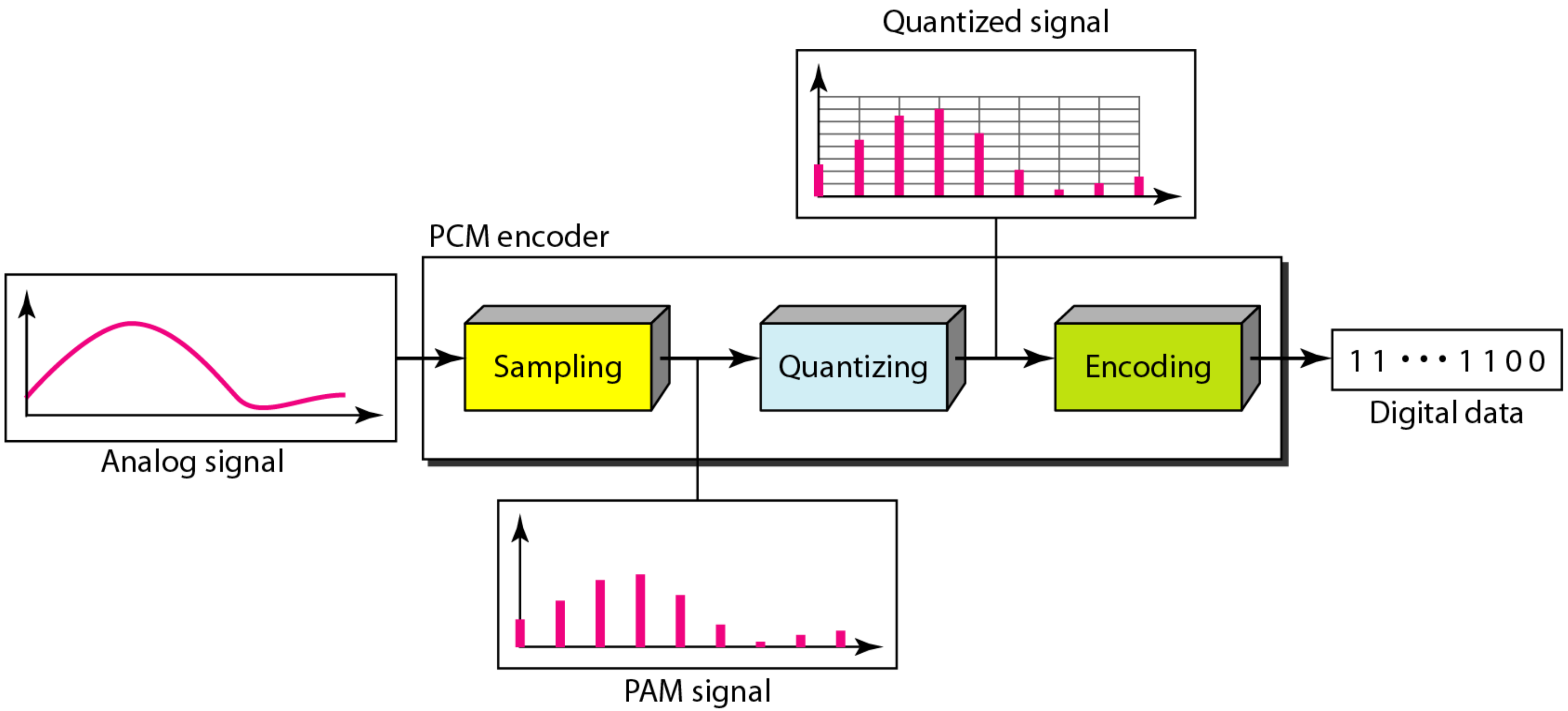
PULSE CODE MODULATION (PCM)

EEEN 464 – DIGITAL COMMUNICATION

Friday, February 6, 2026

PULSE CODE MODULATION (1)

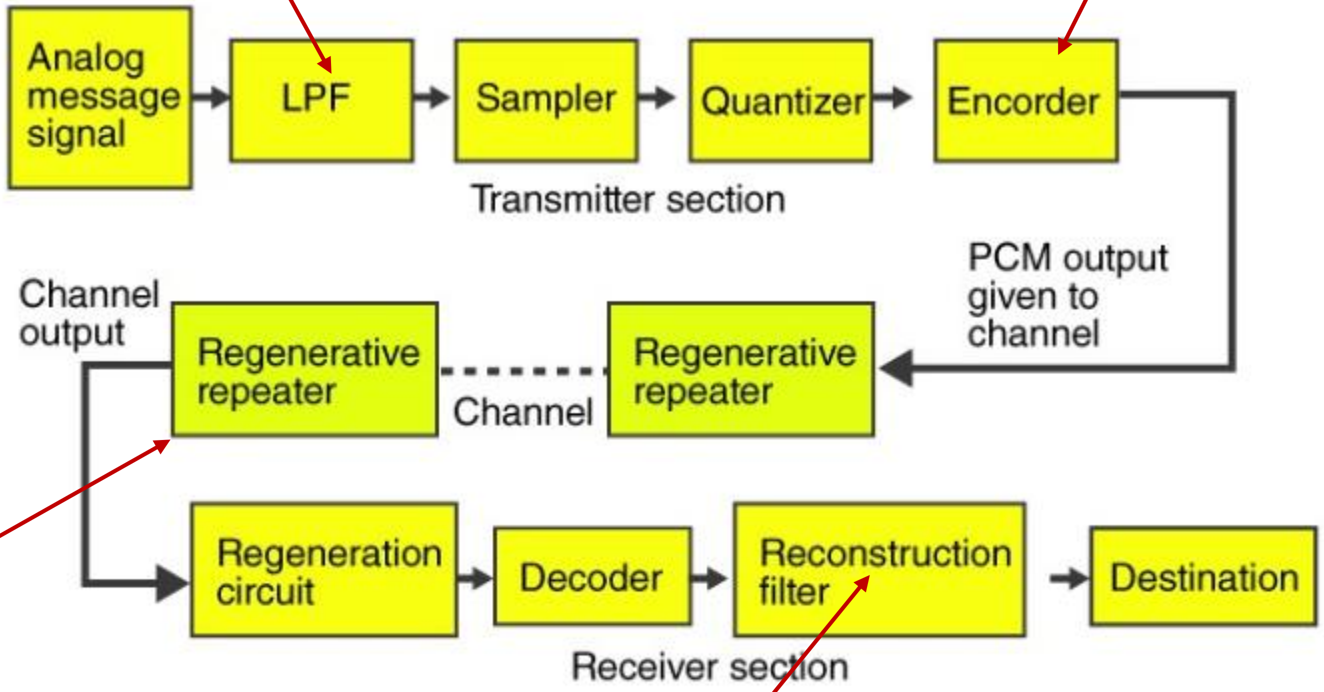
1. **Pulse-code modulation (PCM)** is a method used to digitally represent analog signals.
2. **PCM** is used in digital audio in computers, compact discs, digital telephony and other digital audio applications.



BLOCK DIAGRAM OF PCM COMMUNICATION SYSTEM

Low-pas Filter(LPF)
Anti-aliasing filter which removing the high-frequency components.

Encoder converts quantized values into binary codes



Regenerative repeater is used to compensate for the signal loss and also resynchronize the signal.

Reconstruction filter is a low-pass filter helps to obtain the original signal.

WORKED EXAMPLE - 01

Calculate the Nyquist rate for a signal given by:

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

Solution:

$$x(t) = 3\cos(2\pi f_1 t) + 10\sin(2\pi f_2 t) - \cos(2\pi f_3 t)$$

$$x(t) = 3\cos(2\pi 25t) + 10\sin(2\pi 150t) - \cos(2\pi 50t)$$

Maximum frequency is therefore 150Hz and the maximum Nyquist rate is f_s = 2x150 = 300 Samples/Sec

WORKED EXAMPLE - 02

Find the Nyquist frequency/rate and the Nyquist interval for the following signal:

$$x(t) = \frac{1}{2\pi}\cos(4000\pi t)\cos(1000\pi t)$$

WORKED EXAMPLE - 02 (SOLUTION)

$$2\cos(A)\cos(B) = \cos(A+B) + \cos(A-B)$$

$$x(t) = \frac{1}{2\pi}\cos(4000\pi t)\cos(1000\pi t)$$

$$x(t) = \frac{1}{4\pi}\left(\cos(4000\pi t + 1000\pi t) + \cos(4000\pi t - 1000\pi t)\right)$$

or

$$x(t) = \frac{1}{4\pi}(\cos(5000\pi t) + \cos(3000\pi t))$$

The maximum frequency is therefore 2,500Hz which implies that the Nyquist rate is 5,000 samples/sec and Nyquist interval is 1/5000 = 0.2msec

WORKED EXAMPLE /03

A PCM telephone system has a band-limiting filter of 4 KHz. If each sample is sampled, quantized at 256 levels, calculate the bit-rate of the PCM system.

ANS:

1. According to the Nyquist criterion, we must sample at twice the maximum baseband frequency = 2 x 4 KHz or 8,000 samples/sec.
2. To represent 256 quantized level, the system requires n bits to represent each sample, where $2^n = 256$ or n = 8.
3. The PCM bit rate is therefore 8 x 8,000 = 64 Kbits/s

LINEAR PULSE CODE MODULATION (LPCM)

EEEN 464 – DIGITAL COMMUNICATION

Friday, February 6, 2026

WHAT IS LPCM?

1. **Linear Pulse Code Modulation (LPCM)** is a digital audio format that records sound without compression.
2. **Linear Pulse Code Modulation (LPCM)** that uses linear quantization.
3. **LPCM Works as follows**:
   a) **Analog signal is sampled** at regular intervals
   b) **Amplitude is quantized at uniform levels** and encoded into a series of digital symbols, usually binary
   c) **The resulting digital representation is stored** as a series of numbers
4. **LPCM is used in audio CDs and WAV files**

LPCM AUDIO FORMATS /01

LPCM encoding is used in the following audio formats.

1. **WAV (Waveform Audio File)** developed by IBM and Microsoft in 1991 enables storing audio bitstreams. WAV organizes data in chunks and is the primary format for LPCM audio on Microsoft Windows platforms.
2. **AC3 known commercially as Dolby Digital** supports up to 5 channels, including left, right, center, left surround, right surround, and a low-frequency effects channel.
3. **AES3** exchanges digital audio signals between audio devices. AES3 can transmit two channels of PCM signals or audio over diverse media, such as unbalanced lines, optical fiber, and more.
4. **AES3** the **A**udio **E**ngineering **S**ociety (AES) and the European Broadcasting Union (EBU).

LPCM AUDIO FORMATS /02

4. **MPEG audio (M**oving **P**icture **E**xperts **G**roup**)**, is an organization that creates standards for encoding digital audio and video. It fully supports LPCM audio signals and is one of the most commonly used audio formats today.
5. **Audio Interchange File Format (AIFF)** was developed by Apple in 1988, is a versatile audio file format that stores sound data for electronic audio devices and personal computers.

CLASS EXERCISE

Download the same music file from https://pixabay.com/ in WAV and MP3 and compare the two formats in terms of file size and quality of the music.