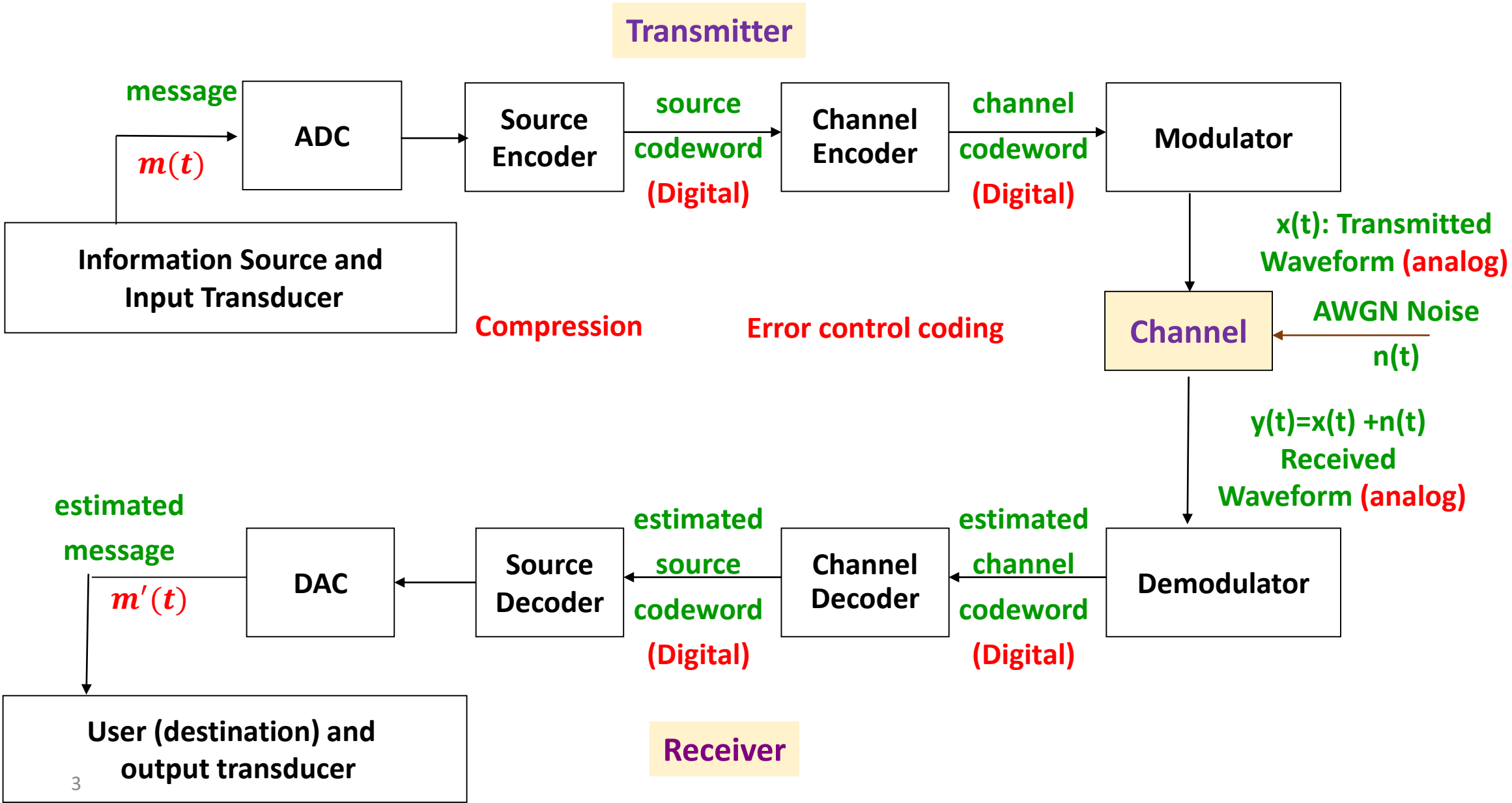# Introduction to Communication Systems

Course Description

- This course introduces computer and electrical engineering students to the principles of transmitting an information bearing signal from a source point to a destination point.

- To achieve this goal, the course presents models and techniques for representing the information signal and its characteristics both in the time and frequency domains.

- The course explains in detail the transmission of the message using analog modulation techniques, such as amplitude and frequency modulation. The focus is on the methods employed for the generation and demodulation of the information bearing signal, as well as bandwidth and power requirements.

- The course also considers the transmission of an analog signal via digital techniques. In this regard, the course highlights the steps involved in this process like sampling, quantization, binary encoding, baseband and bandpass modulation and demodulation schemes.

- Finally, the course analyses the performance of the various transmission schemes in terms of signal-to-noise ratio, bandwidth requirement, and error performance.

Course Outline

- **Signals and Systems Analysis:** Signal Classifications, Fourier Series, Parseval's Power Theorem, Fourier Transform and Properties, Rayleigh Energy Theorem, Linear time-invariant Systems and the Convolution Integral, Pulse Response and Rise-time, Bandwidth of a Signal and a System, Time-Bandwidth Product, Filters, Linear and non-linear Distortion, Hilbert Transform, Correlation and Spectral Density, Bandpass Signals and Systems

- **Amplitude Modulation (AM) Systems**: Normal AM, DSB-SC, Single Sideband, SSB, VSB, For each type we consider the Mathematical Representation, Spectrum, Bandwidth, Generation and Demodulation.

- **Frequency and Phase Modulation**: Basic Definitions, Single Tone FM, Spectral Analysis and Bandwidth, Carson's Rule, Generation and Demodulation.

- **Pulse Code Modulation**: Ideal, Natural, and Flat-Topped Sampling, Aliasing, Uniform Quantization, SQNR, Companding and non-uniform Quantization, DPCM, Delta Modulation,

- **Baseband and Bandpass Data Transmission**: The Matched Filter, PSD, Bandwidth and Probability of Error for Some Baseband and Bandpass Modulation Techniques, Like PNRZ, Manchester Coding, ASK, PSK, Coherent and Non-Coherent Binary FSK, QPSK.

- **Performance of Analog and Digital Communication Systems in Noise.**

A Basic Communication System Block Diagram
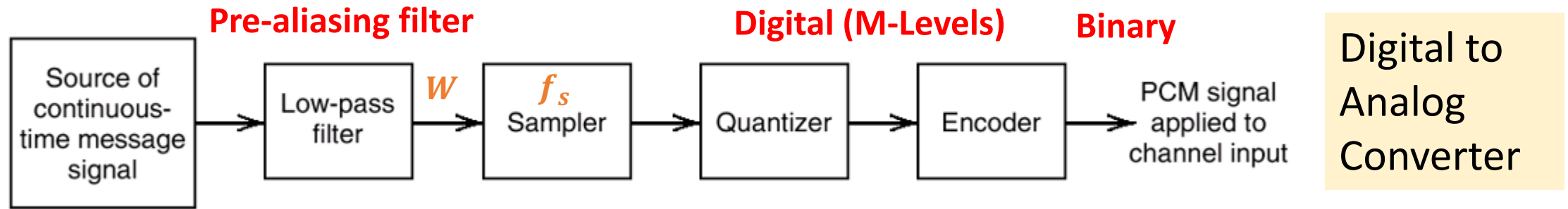
Communication System: Overall Purpose

- **Communication** is defined as "exchange of information".
- **Telecommunication** refers to communication over a distance greater than would normally be possible without artificial aids.
- **Telephony** is an example of **point-to-point** communication and normally involves a two –way flow of information.
- **Broadcast radio and television**: Information is transmitted from one location but is received at many locations using different receivers (point to multi-point communication)
- The purpose of a communication system is to transmit information from a source located at one end to a user (destination) located at another end.
- A distortion-less communication system is one for which **m'** is an exact replica of **m**. The task of the communication system is to minimize the difference between m and m', i.e., minimize

$$MSE = E(m - m')^2; \text{ Mean Squared Error}$$

- The communication system must be **efficient**, i.e., represent data with minimum number of binary digits.
- The communication system must be **reliable**, i.e., transmit data with minimum probability of error.

Communication System: Types of Sources

- Sources are of two types; **analog and digital**. For an analog source, the input transducer is used to convert the physical message generated by the source into a time-varying electrical signal called the *message signal* (like the human voice).

- The source could also be digital in nature (e.g., email text).

- An analog source can be converted into digital via sampling, quantization, and encoding.

Pre-aliasing filter   **Digital (M-Levels)**   **Binary**

Source of continuous-time message signal → Low-pass filter — W → f_s Sampler → Quantizer → Encoder → PCM signal applied to channel input | Digital to Analog Converter |

➤ **Sampler:** If W is the highest frequency component in a signal, then the sampling rate should follow the Nyquist Rate where $f_s > 2W$.

➤ The output of the sampler is a continuous amplitude discrete time signal.

➤ **Quantizer:** Converts the continuous amplitude samples into **discrete** level samples. Output belongs to a finite set of countable symbols,

➤ $S = \{m_1, m_2, \ldots, m_M\}$. This will be our discrete source of Size M.

➤ **Binary Encoder:** Each quantized level is represented by $r = log_2 M$ binary digits

Communication System: Source Encoding

- **Source encoder:** designed to represent the source information more compactly by removing redundancies of the source.

- The input to the source encoder is a string of symbols occurring at a rate of R_s symbols/sec. It converts the symbol sequence into a binary sequence of 0's and 1's by assigning *codewords* to the symbols in the input sequence.

- The source is characterized by the **entropy *H*** which represents the minimum number of bits/symbol that can represent the source.
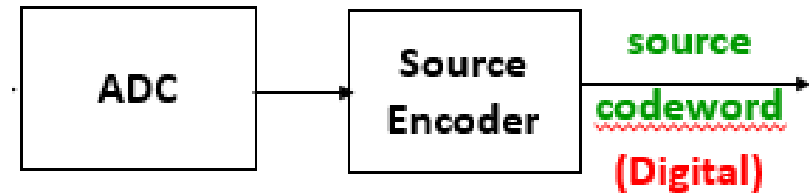

  Two possible ways of encoding a source:

- **a. fixed length encoding** : Here the encoder assigns a fixed length codeword to each input symbol regardless of the probability of occurrence of the symbol . This encoding scheme is *efficient* if the symbols occur with equal probability in a statistically independent sequence .

- **b. variable length encoding** : Here the encoder assigns codewords with variable lengths depending upon the probability of occurrence of the input symbol. Symbols with high probability of occurrence are assigned short codewords, while symbols with low probability of occurrence are assigned long codewords.

Communication System: Channel Encoding

- **Channel Encoder**: Here, extra bits are deliberately added to the input data sequence. These extra bits are used to correct or detect the errors in the received channel codeword. If all errors are corrected, the estimated codeword matches the original source codeword (High transmission reliability is achieved).
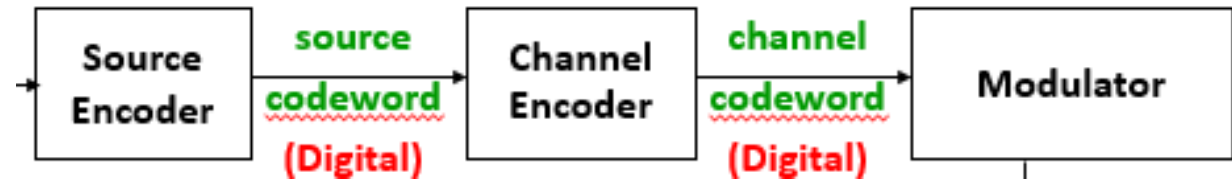
Types of Channel Encoders

- **Block codes** : the encoder takes a block of bits of size k bits and adds r error control bits (parity check bits) to produce a block of size n. The notation used is (n,k) block code

- **Convolution codes**: the information bearing stream is encoded in a continuous fashion by continuously interleaving information bits and error control bits.

- The channel is characterized by a **capacity C** which determines the maximum rate at which information Rs can be transmitted over the channel with an arbitrarily small probability of error, $R_s < C$.

Communication System: Modulation

- The **modulator at the transmitter side** changes some feature of a high frequency carrier in accordance with the message signal. In general, the carrier takes the form $\mathbf{c(t)} = \boldsymbol{Acos(2\pi f_c t + \varphi)}.$ The message m(t) may change the amplitude, frequency, or phase of the carrier resulting in amplitude, frequency, or phase modulation.

- The modulator can be used to:
  - Minimize the effect of channel noise.
  - Match the frequency spectrum of the transmitted signal with channel characteristics.
  - Provide the capability to multiplex many signals and to overcome some equipment limitations.

- **When the message signal is a digital one, the modulator converts the bits into one of two signals corresponding to digits 1 or 0:**
  - Digit 1 is mapped into a signal $s_1(t)$
  - Digit 0 is mapped into a signal $s_2(t)$

- $s_1(t)$ and $s_2(t)$ must be distinct so the receiver can differentiate between them and reduce the probability of error.

- The probability of error decreases as the distance (i.e., the difference) between the two signals increases.

Modulator

x(t): Transmitted
Waveform (analog)

AWGN Noise

Channel

n(t)
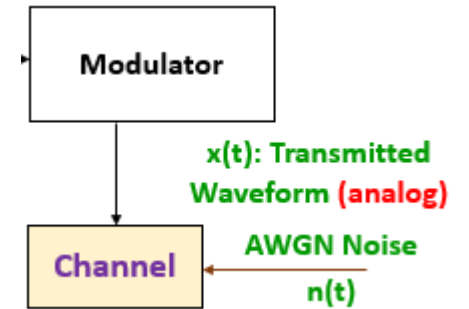
Digital Modulation Schemes

Examples on Binary Modulation: The binary digits modulate a high frequency carrier in amplitude, phase, or frequency (e.g., ASK, PSK, FSK, QPSK).
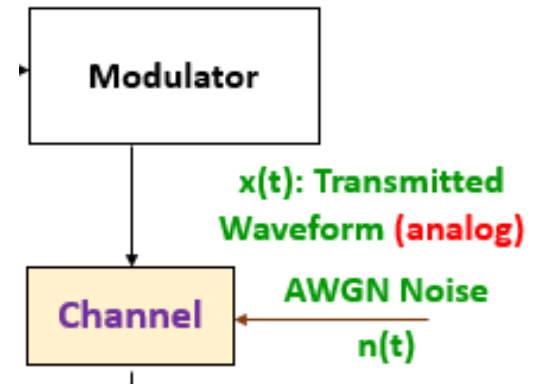
0 1 0 0 1 1 0 0 0 1 1
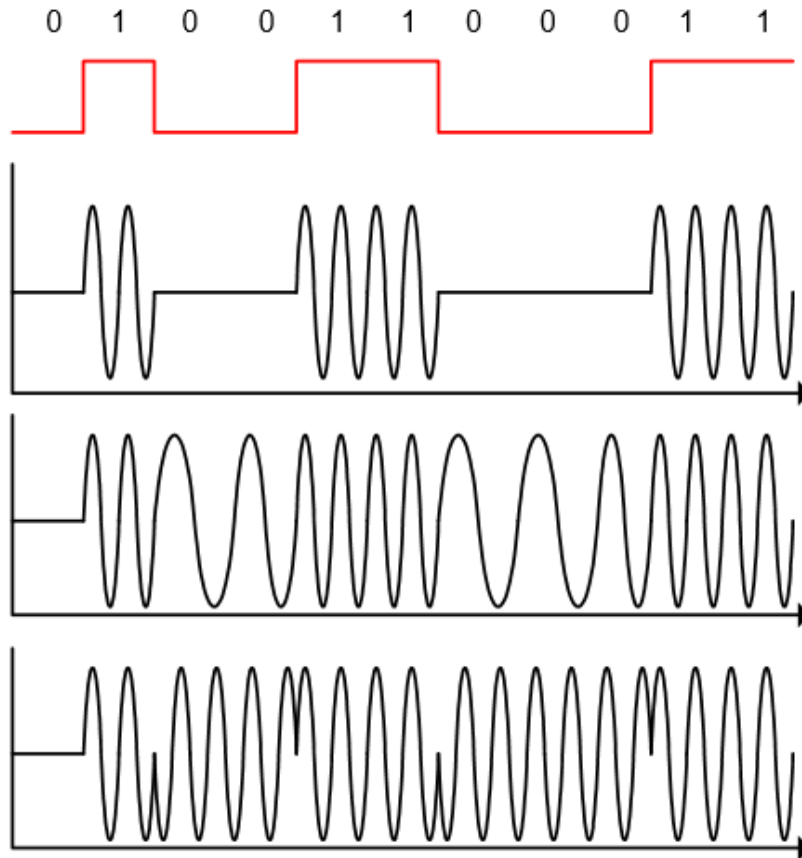
Original signal (in a pulse sequence)

In unipolar non-return to zero format (NRZ) or polar NRZ

Amplitude shift-keying (ASK)

$$s_1(t) = A\cos(2\pi f_c t) \qquad s_2(t) = 0$$

Frequency shift-keying (FSK)

$$s_1(t) = A\cos(2\pi(f_c + \Delta f)t)$$

$$s_2(t) = A\cos(2\pi(f_c - \Delta f)t)$$

Phase shift-keying (PSK)

$$s_1(t) = A\cos(2\pi f_c t) \quad s_2(t) = A\cos(2\pi f_c t + \pi)$$

Modulator

x(t): Transmitted Waveform (analog)

AWGN Noise

Channel

n(t)

Communication System: Channel

- The ***channel*** is the medium over which the signal is transmitted, (wired or wireless), like free space, an optical fiber, transmission lines, twisted pair of wire. The channel introduces impairments on the transmitted signal due to

- **Linear distortion**: A limited bandwidth which results in linear distortion (amplitude distortion and/or phase distortion). Channel does not obey the distortion-less condition for transmission:

$$y(t) = kx(t - t_d) ,\qquad \text{time-domain}$$

$$Y(f) = kX(f)e^{-j2\pi f t_d} \quad , \text{frequency-domain}$$
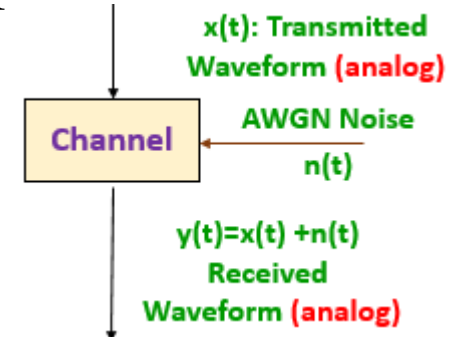
   k: constant amplification (or attenuation)

   t_d: constant time delay
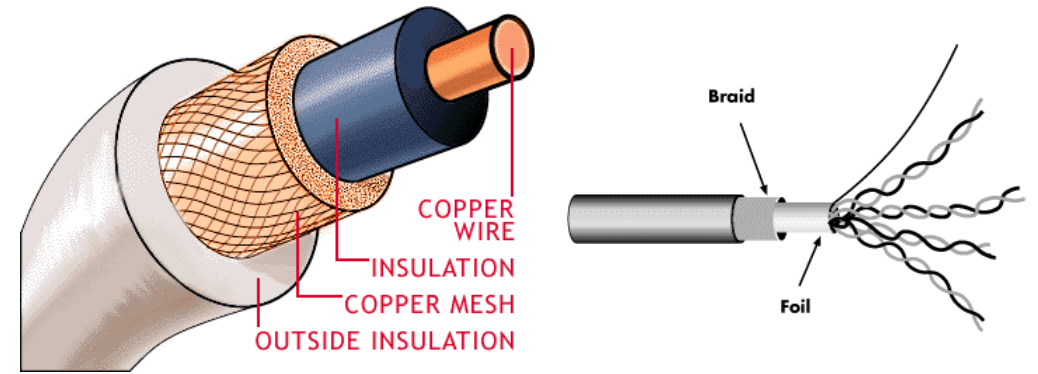
- The channel be also have some degree of non-linearity resulting in non-linear distortion. These **nonlinearities and/or imperfections** in the frequency response of the channel generate new frequencies not originally present in the signal.
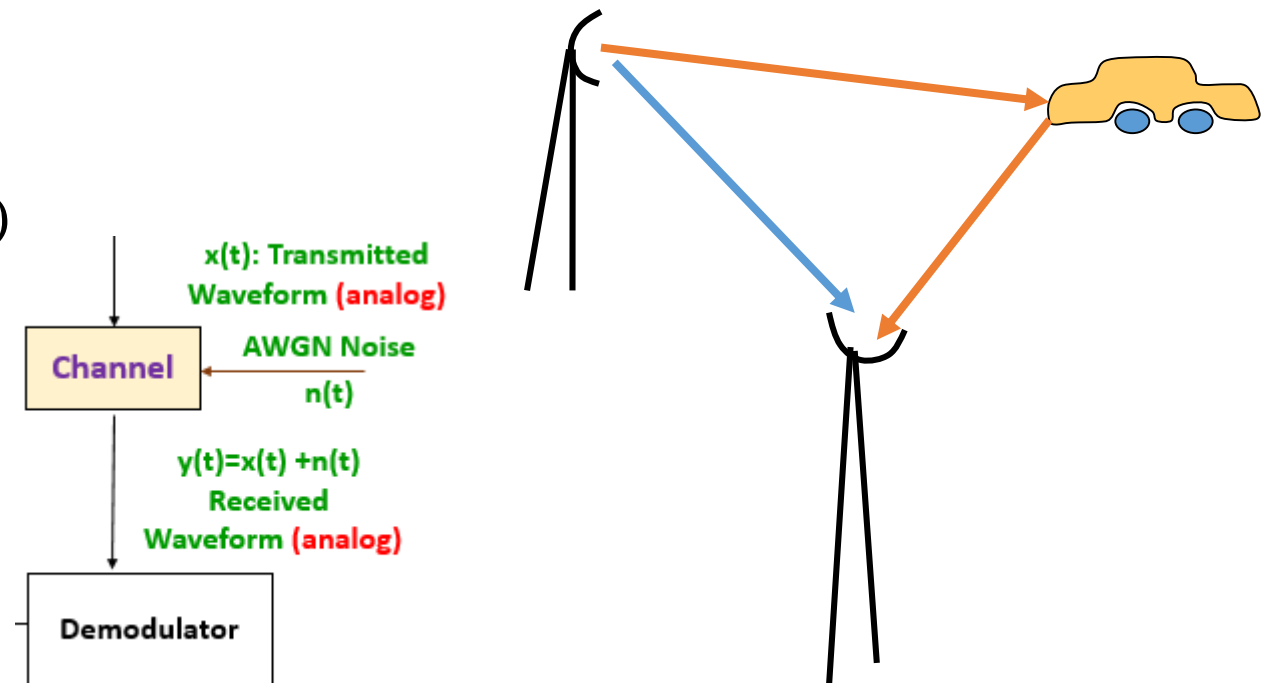
- Input: $x(t) = \cos(2\pi f_0 t)$;

- Output: $y(t) = 0.9cos(2\pi f_0 t) + 0.1\cos(6\pi f_0 t)$

Braid

COPPER WIRE

INSULATION

COPPER MESH

OUTSIDE INSULATION

Foil

x(t): Transmitted Waveform (analog)

Channel

AWGN Noise n(t)

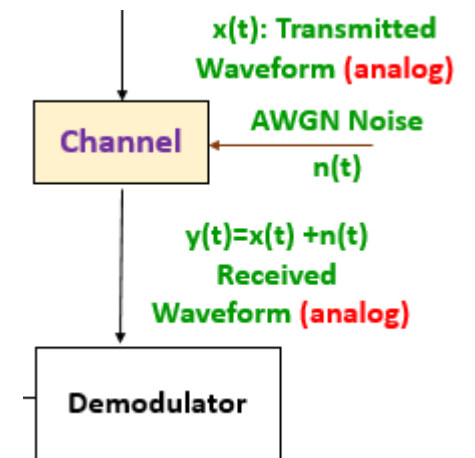y(t)=x(t) +n(t)
Received Waveform (analog)

Communication System: Interference

- **Interference**: Multiple copies of the transmitted signal are received at the destination. They arrive at different times due to multipath propagation of the radio waves (waves bouncing on buildings, trees, cars, etc)

- Input: $x(t) = \cos(2\pi f_0 t)$;

- Output: $y(t) = 0.8 cos(2\pi f_0(t - t_0) + 0.4 cos(2\pi f_0(t - 2t_0)$

- The received components may be:
  - In-phase (constructive interference)
  - Out of phase (destructive interference)

x(t): Transmitted Waveform (analog)

AWGN Noise n(t)

Channel

y(t)=x(t) +n(t) Received Waveform (analog)

Demodulator

- **Additive White Gaussian Noise** is a basic noise model used in communication systems to mimic the effect of many random processes that occur in nature.

- This noise comes from many natural noise sources, such as the thermal vibrations of atoms in conductors (referred to as thermal noise), shot noise, black-body radiation from the earth and other warm objects, and from celestial sources such as the Sun.

- The central limit theorem of probability theory indicates that the summation of many random processes will tend to have distribution called Gaussian or Normal.

  - Transmitted signal: $x(t)$;
  - Channel Output: $y(t) = x(t) + n(t)$;
  - The pdf of $n(t)$ follows the Gaussian distribution
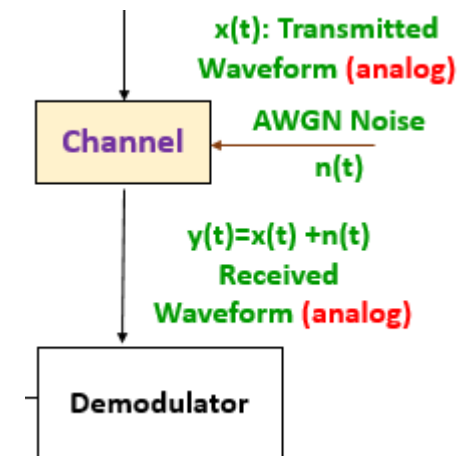 - The power spectral density is a constant over a wide range of the frequency spectrum

$x(t)$: Transmitted Waveform (analog)

AWGN Noise
$n(t)$

Channel

$y(t) = x(t) + n(t)$
Received Waveform (analog)

Demodulator

Communication System: Optimum Binary Receiver Performance

- The purpose of the *receiver* is to recreate an estimate of original signal x(t) from the degraded version x(t) + n(t) of the transmitted signal over the noisy channel. *Here, demodulation takes place*

- In a digital data transmission, the receiver has to decide which symbol was transmitted such that the probability of making errors in minimized. The receiver which satisfies this criterion is called an **optimum receiver.**

- Bit Error Probability (in the binary case): $P_b^* = Q\left(\sqrt{\dfrac{\int_0^\tau (s_1(t) - s_2(t))^2 \, dt}{2N_0}}\right)$

- τ: **binary symbol duration**

- N_0: **AWGN power**

x(t): Transmitted Waveform (analog)

AWGN Noise n(t)

Channel

y(t)=x(t) +n(t) Received Waveform (analog)

Demodulator

Analog vs Digital: Advantages and Disadvantages

An analog signal can take an infinite variety of shapes. The distortion caused by noise cannot be removed by amplification or filtering.

Advantages of Digital Transmission

- Digital signals are more immune to channel noise by using channel coding techniques where error correction can be implemented (ideally, perfect decoding is possible by virtue of Shannon channel coding theorem).

- Digital signals belong to a finite set of possible waveforms. Repeaters along the transmission path can identify the transmitted digital waveform logically (0 or 1) and regenerates a noise free pulse sequence.

- Digital signals derived from all types of analog sources can be represented using a uniform format.

- Digital signals are easier to process by using microprocessors and VLSI

- Digital systems are more flexible to implement and allow for implementation of sophisticated functions and control.

Analog vs Digital: Advantages and Disadvantages

- Digital signals make use of digital signal processing techniques (encryption, error control coding,…). This will enhance security of the transmitted signal.

- Digital circuits are less subject to distortion and interference

- Digital circuits are more reliable and less expensive than analog circuits

- Easier data multiplexing

Few Disadvantages of Digital Transmission

- Heavy signal processing

- Synchronization is crucial

- Transmission bandwidth is large

- When the S/N ratio drops below a certain value, the quality of service can change suddenly from very good to very bad

Milestones in Communications

- 1837: Samuel Morse developed the Morse code used in telegraph.
- 1843 – Samuel Morse builds the first long distance electric telegraph line (telegraph remained in service in the US until 2006)
- 1864: Maxwell formulated the electromagnetic theory 1864
- 1875: Bell invented the telephone
- 1887: Hertz demonstrated physical evidence of EM waves, meaning that EM waves can be transmitted and received.
- 1890's-1900's: **Marconi** & Popov established (independently) the long-distance radio
- 1906: Radio broadcast began
- 1918: Armstrong invented super-heterodyne radio receiver (and FM in 1933)
- 1925 – John Logie Baird transmits the first television signal.
- 1928: Nyquist proposed the sampling theorem.

- 1947: Era of Microwave relay systems
- 1947 – Full-scale commercial television is first broadcast.
- 1948: Information theory formulated by Shannon.
- 1957: Era of satellite communication began.
- 1965 – First email sent (at MIT)
- 1966: **Kuen Kao** pioneered fiber-optical communications (Nobel Prize Winner).
- 1970's: Era of computer networks began.
- 1981: Analog cellular system (1-G AMPS System).
- 1988: Digital cellular system debuted in Europe (2-G Mobile System)
- 1990's: Era of internet
- 2000: 3G Mobile network.

The Basic Human Communication Model
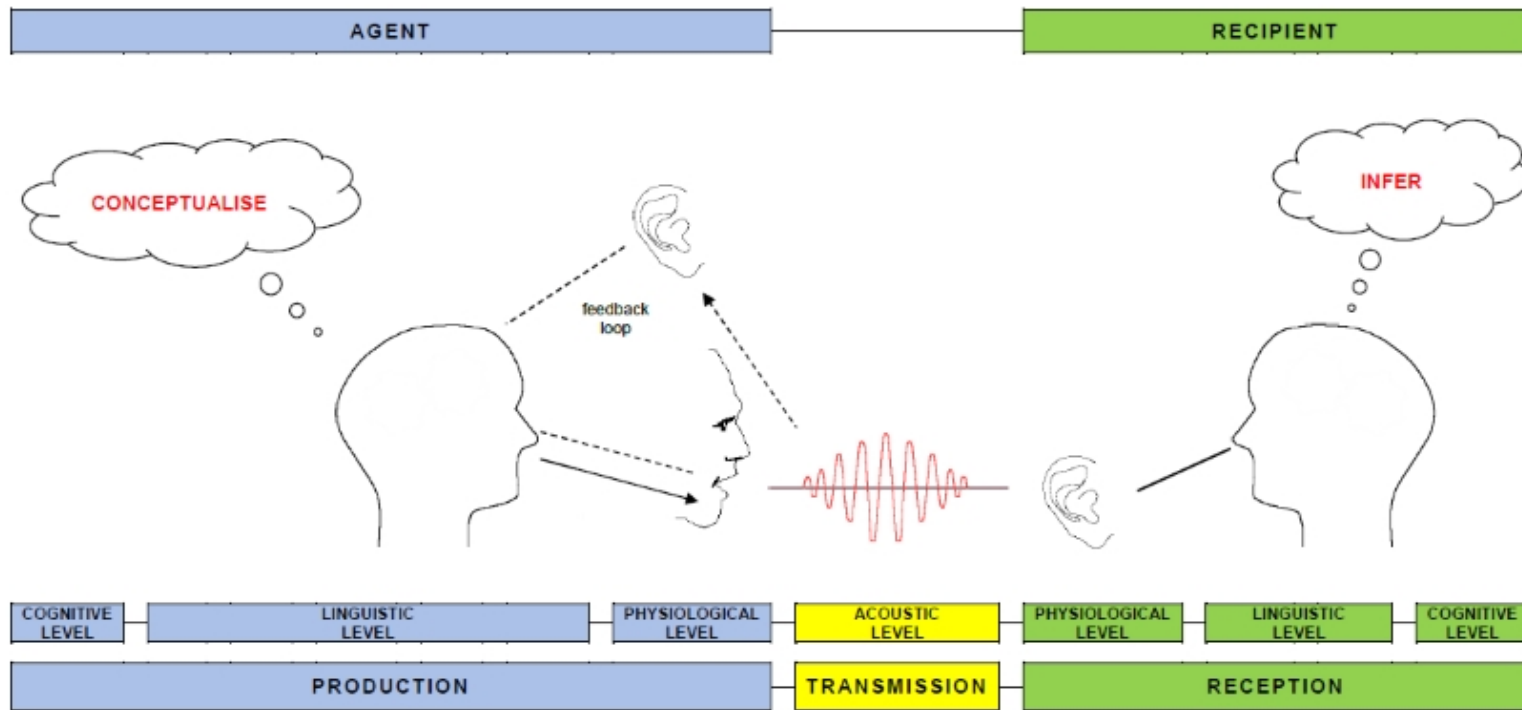
This model consists of three main parts:
Production: is the process by which a human agent expresses himself through first deciding what message s/he wishes to communicate (cognitive level). He then plans and **encodes** the appropriate linguistic utterance to represent the concept (linguistic level) and, finally, produce this utterance through the suitable co-ordination of the vocal apparatus (physiological level).

Transmission: is the sending of the linguistic utterance through some medium to the recipient. Here, the spoken utterance travels through the medium of air to the recipient's ear (acoustical waves).

Reception: is the process by which the **recipient** detects the verbal utterance through the sense of hearing (physiological level) and then **decodes** the linguistic expression (linguistic level). Finally, s/he then infers what is meant by the linguistic expression (cognitive level).

https://www.sltinfo.com/communication-model/

Morse Telegraph

- Before the invention of the telegraph, most messages that had to be sent over long distances were carried by messengers
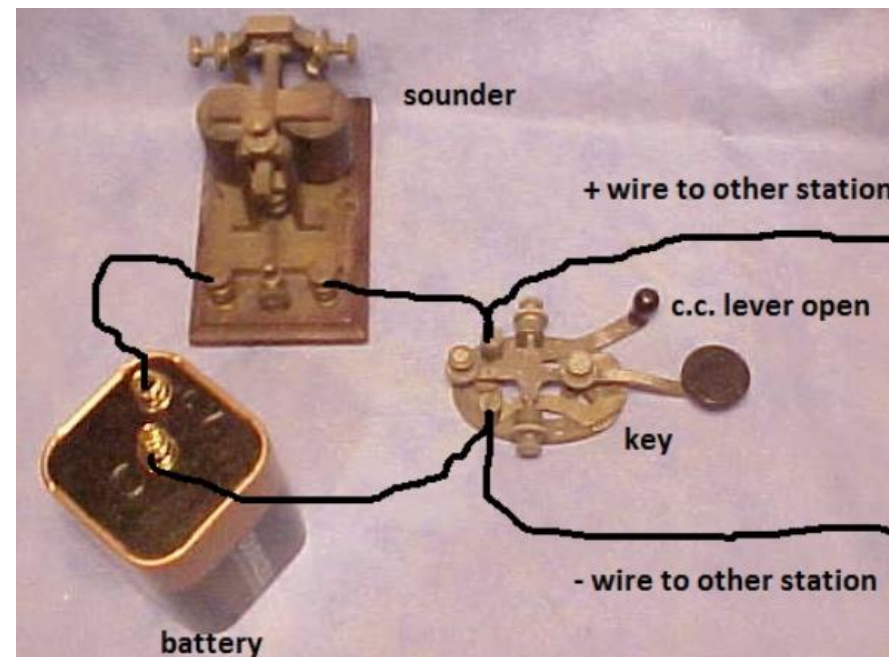
Electric Telegraph: human to human transmission of coded messages

- Wires are stretched from one point to another; electric current is either allowed to flow through the wires or is broken by a switch called telegraph key
- Electric current is used to activate a sounder which makes clicking sounds – short / long times between clicks are decoded into letters from the alphabet

Morse Telegraph: text message is encoded into a sequence of **dots and dashes** [1837]

- Dots and dashes are converted into short and long pulses of electric current
- Digital transmission system – relies only on 2 signal levels (data rate 20 bits/sec) **"What hath God wrought!"**

	Morse Code		Morse Code		Morse Code		Morse Code
A	· —	J	· — — —	S	· · ·	2	· · — — —
B	— · · ·	K	— · —	T	—	3	· · · — —
C	— · — ·	L	· — · ·	U	· · —	4	· · · · —
D	— · ·	M	— —	V	· · · —	5	· · · · ·
E	·	N	— ·	W	· — —	6	— · · · ·
F	· · — ·	O	— — —	X	— · · —	7	— — · · ·
G	— — ·	P	· — — ·	Y	— · — —	8	— — — · ·
H	· · · ·	Q	— — · —	Z	— — · ·	9	— — — — ·
I	· ·	R	· — ·	1	· — — — —	0	— — — — —

# The Classical Telephone System

- Bell discovered that the human voice can be transmitted over wires, which led to the invention of the telephone.

The components of a telephone system include

- **Microphone:** Converts the vibrations in the air into an electrical signal
- **Receiver**: Converts the received electrical signal into sound waves (the reverse action of a microphone) e.g.: loudspeaker
- **Transmission system**: Conveys the information representing the audio signal from the microphone to the receiver
- **Signaling and switching system**: Determines and makes appropriate connections among the pieces of the transmission system to create a path from the transmitter to the receiver
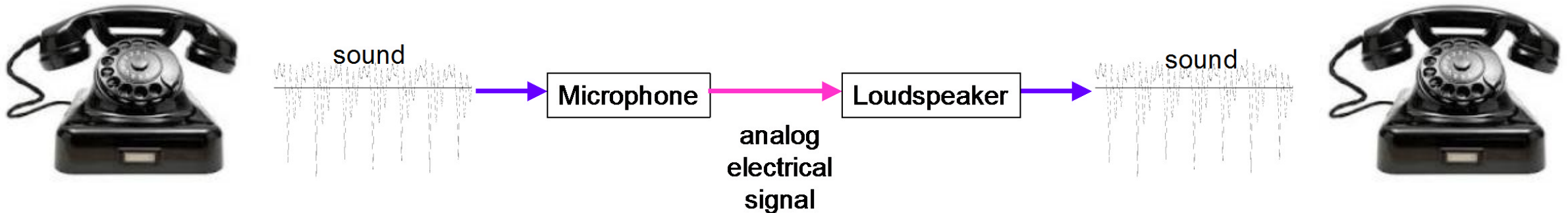
Information within human speech signals: 100 to 12,000 Hz, but Telephones filter speech to a predetermined (standard) frequency range set at 200 to 3400 Hz.

Satellite Communications

Satellite

5.9-6.4 GHz

3.7-4.2 GHz

Uplink

Downlink

Ground stations
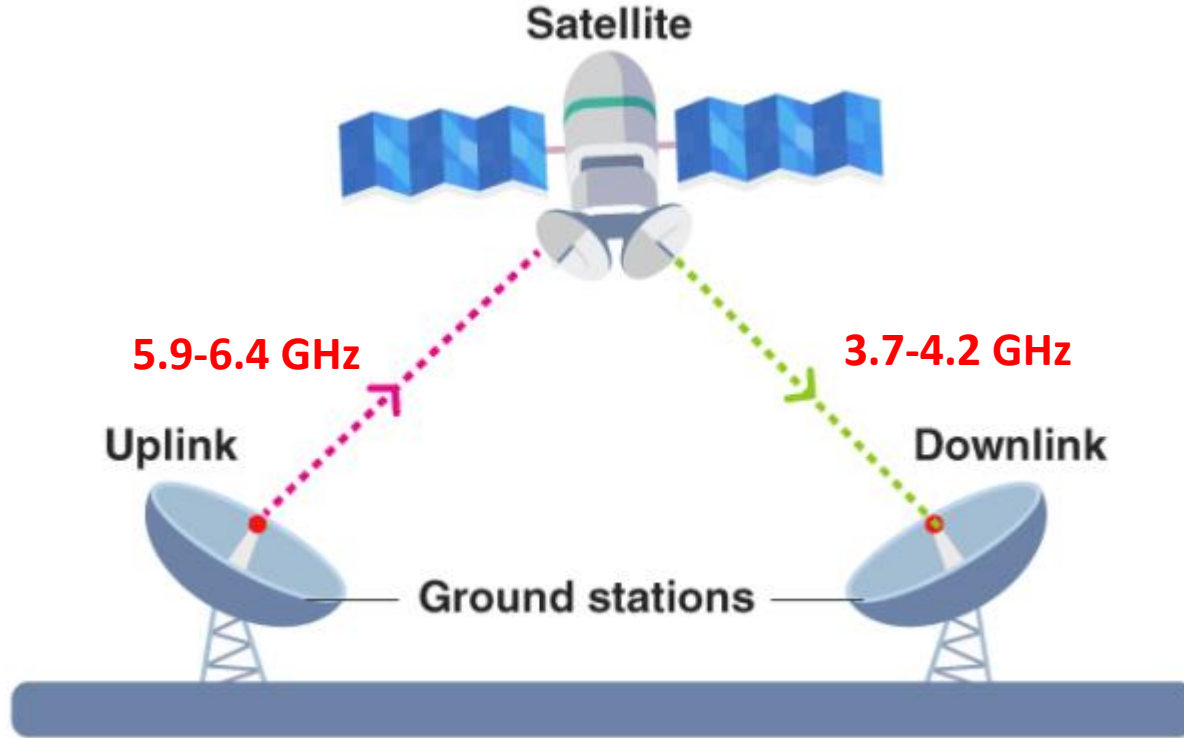
- Satellite communication is the method of transporting information from one place to another using a communication satellite in orbit around the Earth.
- A communication satellite is an artificial satellite that transmits the signal via a transponder by creating a channel between the transmitter and the receiver located at different locations on the Earth.
- Telephone, radio, television, digital cinema, and internet use satellite communications.

There are three stages that are involved which explain the working of satellite communications. These are:

Uplink: The process by which the signal on the other side of the earth is first beamed up to the satellite from the ground station on the earth.

Transponders: The second stage involves transponders such as radio receivers, amplifiers, and transmitters. These transponders are used for boosting the incoming signal and to change their frequency so that the outgoing signals are not altered.

Downlink: Here the data is sent to the other end of the receiver on the earth.

https://www.sltinfo.com/communication-model/
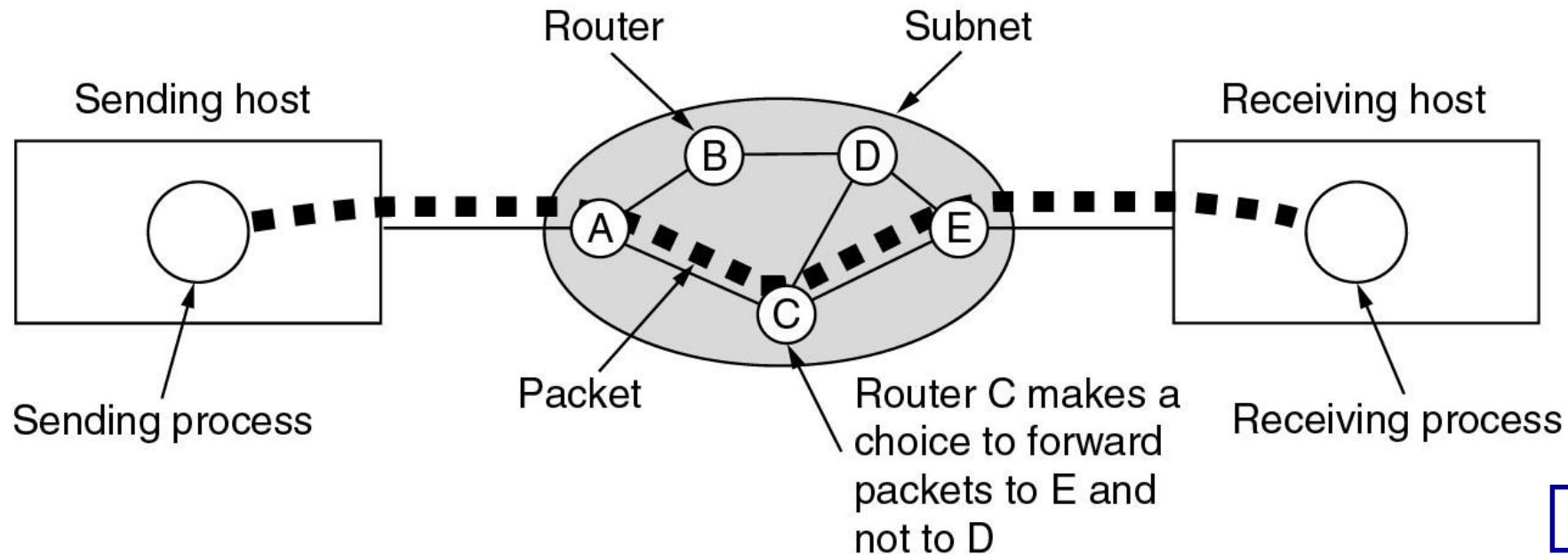
Cellular Mobile Phone Network
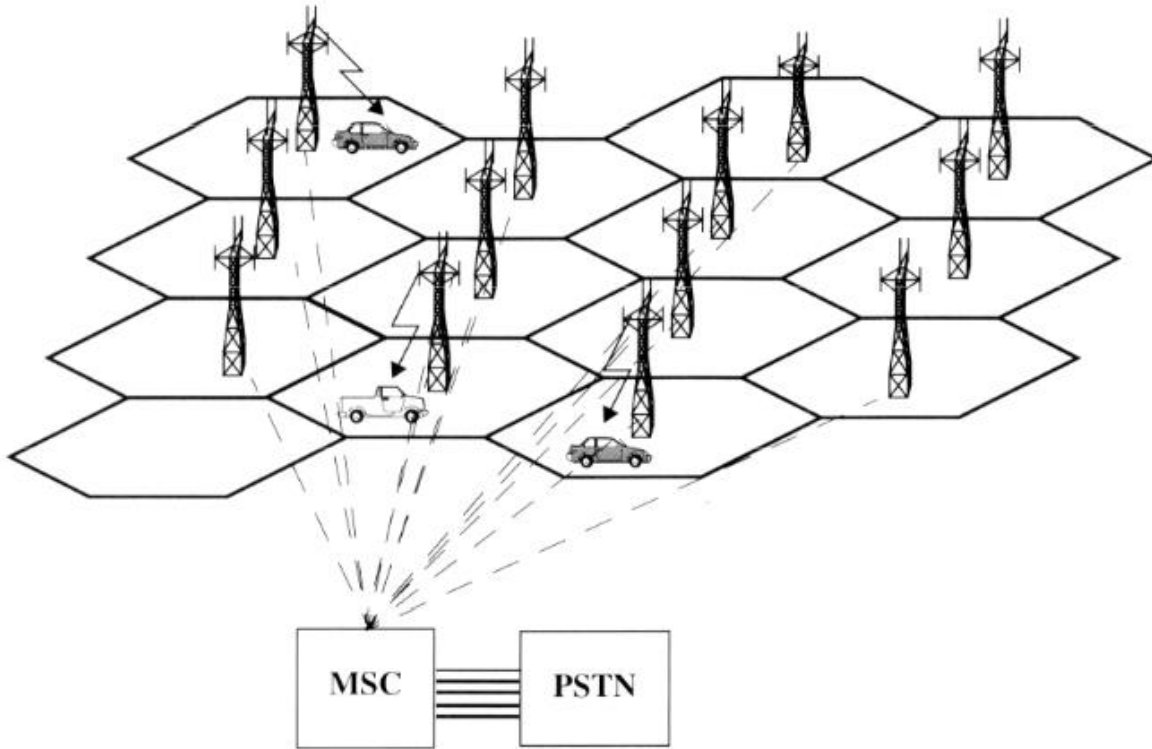
- the signal from a mobile unit (cell phone) to a *base station* is transmitted by *radio waves* through the air.
- The area is divided into a number of cells (typically 2 to 10 miles in size, but can be smaller for more crowded areas) and a base station is positioned within each cell.
- Frequency reuse is employed to maximize capacity
- If a user (mobile phone) is within a particular cell, the call is handled by the corresponding base station within that cell
- The base station transmits the signal to the mobile switching center (MTS) which switches the signal to another base station, or to a Public Switched Telephone Network (PSTN), depending on the destination of the call: whether another mobile unit or a regular telephone.

- As a user moves from one cell to another, the call is "handed over" to the base station of the other cell-This is called *hand-off*
- The handover is (hopefully) transparent to the user.

Wide Area Networks (WANs)

Router   Subnet

Sending host   Receiving host

Sending process   Packet   Router C makes a   Receiving process
choice to forward
packets to E and
not to D

Tanenbaum

- **Host**: Collection of machines intended for running user programs.

- **Subnet**: Hosts are connected by communication subnet. Its job is to carry messages from host to host.

- In a WAN, the subnet consists of two distinct components:

  - **Transmission Lines**: (Also called channels, trunks) Move bits between machines.

  - **Switching Elements**: These are specialized computers used to connect two or more transmission lines. When data arrives on an incoming line, the switching element must choose an outgoing line to forward them on.
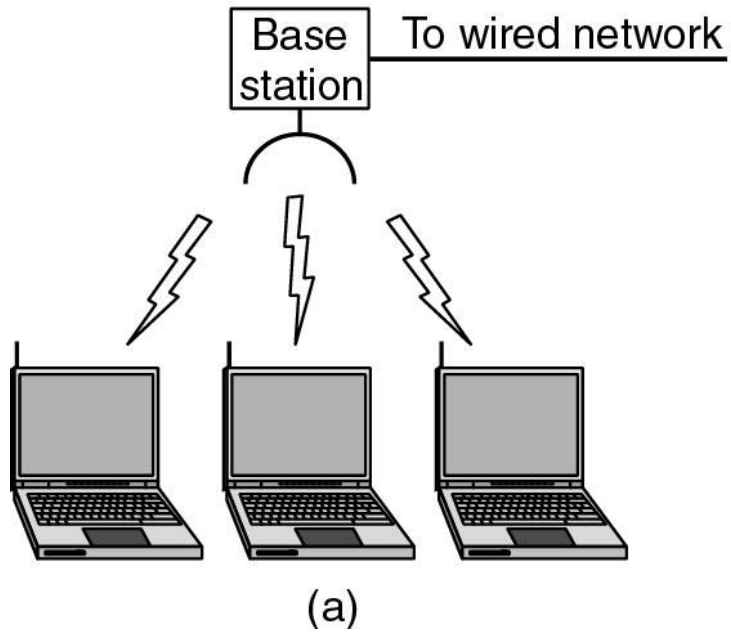
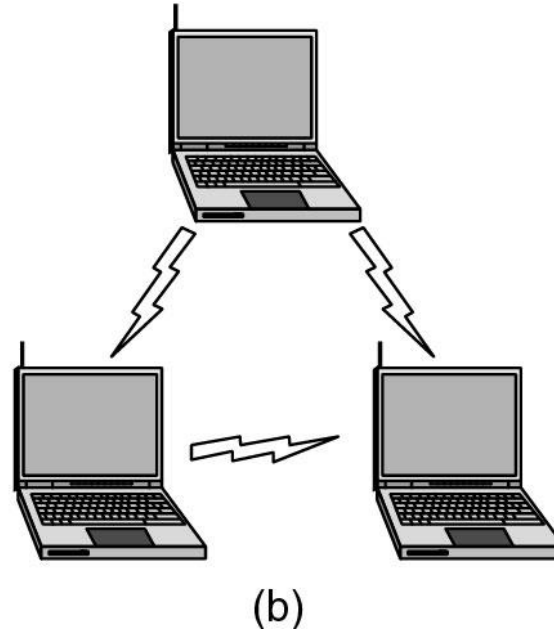# Internet

A network of networks

G = gateway
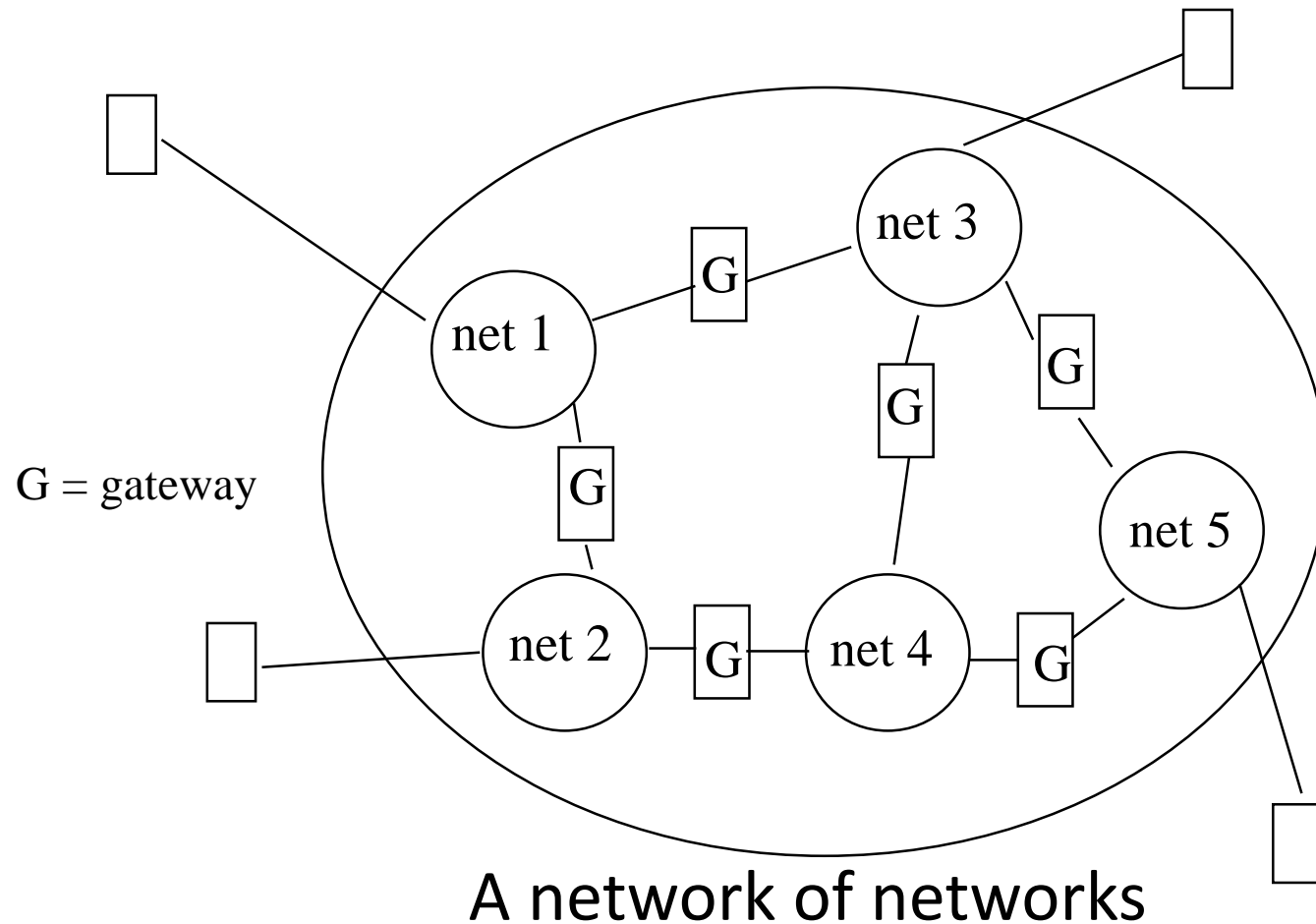
- The **Internet** is the global system of interconnected computer networks that uses the Internet protocol suite (TCP/IP) to communicate between networks and devices.
- It is a *network of networks* that consists of private, public, academic, business, and government networks of local to global scope, linked by a broad array of electronic, wireless, and optical networking technologies.
- The Internet carries a vast range of information resources and services, such as the inter linked hypertext documents and applications of the World Wide Web (WWW), electronic mail, telephony, and file sharing (Wikipedia)

Wireless Local Area Networks (WLANs)

(a) Wireless networking with a base station.

(b) Ad hoc networking.

Tanenbaum

- A **wireless LAN** (**WLAN**) is a wireless computer network that links two or more devices using wireless communication to form a local area network (LAN) within a limited area such as a home, school, computer laboratory, campus, or office building.
- This gives users the ability to move around within the area and remain connected to the network.
- Through a gateway, a WLAN can also provide a connection to the wider Internet (Wikipedia)

Electromagnetic Spectrum and its use in Communication