Introduction to Telecommunications and Computer Engineering

Unit 3: Communications Systems & Signals

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Acknowledgements

These notes contain material from the following sources:

[1] *Data Communications and Networking*, by B.A.Forouzan, McGraw Hill, 2003.

[2] *Communication Systems*, by R.Palit, CSE Department, North South University, 2006.

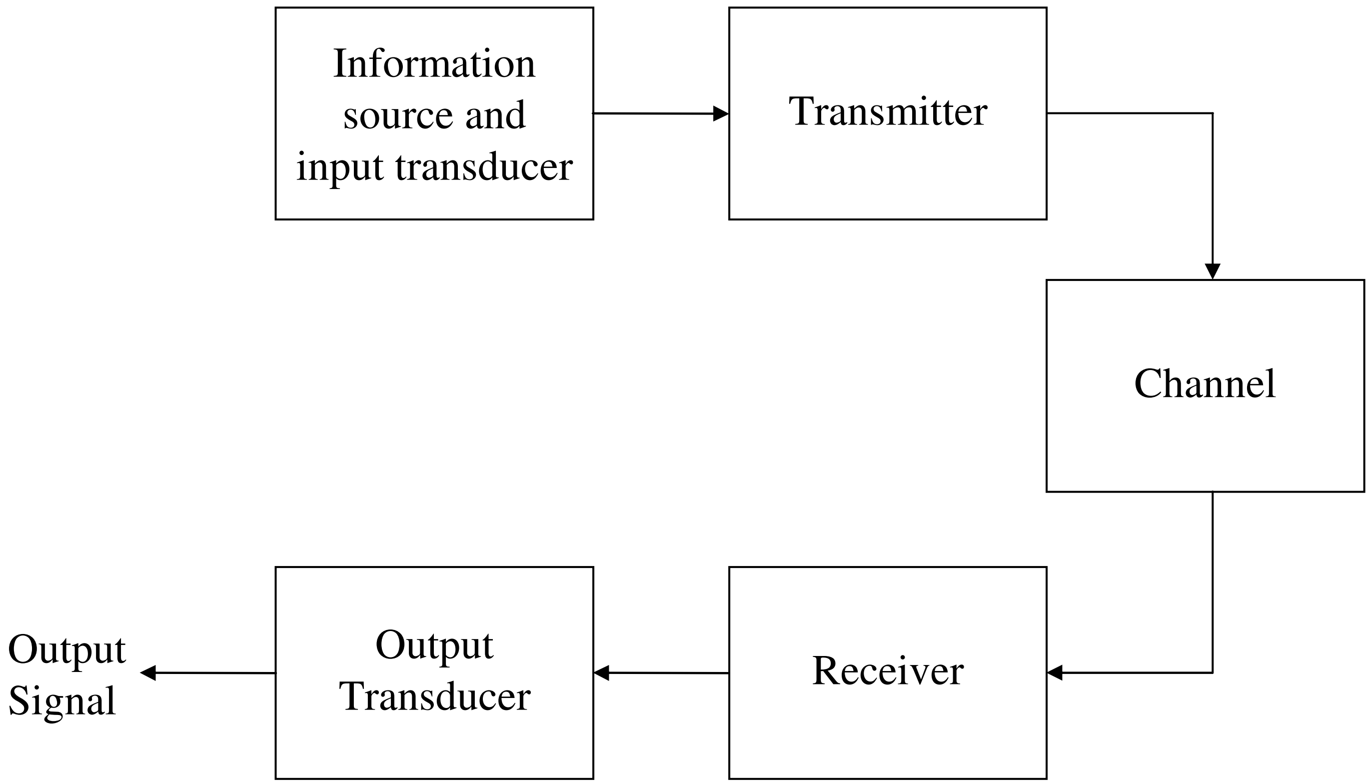
[3] *Electromagnetic Spectrum* and *Harmonics*, Wikipedia the Free Encyclopedia, www.wikipedia.org, 2007.

Basic Definitions

Communication can be defined as the successful transmission of information through a common system of symbols, signs, behavior, speech, writing, or signals.

Telecommunication refers to the communication of information at a distance. This covers many technologies including radio, telegraphy, television, telephone, data communication and computer networking. Telecommunication can be point-to-point, point-to-multipoint or broadcasting,

Model of Communication System



Functional Block Diagram of a Communication System

Fundamental Characteristics of a Communication System

1. **Delivery** - The system must deliver data to the correct destination.
2. **Accuracy** - The system must deliver the data accurately.
3. **Timeliness** - The system must deliver data in a timely manner.

   In the case of video and audio, timely delivery means delivering data as they are produced, in the same order that they are produced, and without significant delay. This kind of delivery is called **real-time transmission**.

Components of Communication systems

1. **Message** - the information (data) to be communicated. It can consist of text, numbers, pictures, sound, or video—or any combination of these.
2. **Sender** - the device that sends the data message. It can be a computer, workstation, telephone handset, video camera, and so on.
3. **Receiver -** the device that receives the message. It can be a computer, workstation, telephone handset, television, and so on.
4. **Medium** - the physical path by which a message travels from sender to receiver. It could be a twisted-pair wire, coaxial cable, fiber optic cable, or radio waves (terrestrial or satellite microwave).
5. **Protocol -** a set of rules that governs data communications. It represents an agreement between the communicating devices. Without a protocol, two devices may be connected but not communicating.

Modes of Communication

- ***Simplex transmission*** - signals are transmitted in only one direction; one station is transmitter and the other is receiver.
- ***Half-duplex transmission*** - both stations may transmit, but only one at a time
- ***Full-duplex transmission*** - both stations may transmit simultaneously

Types of Data

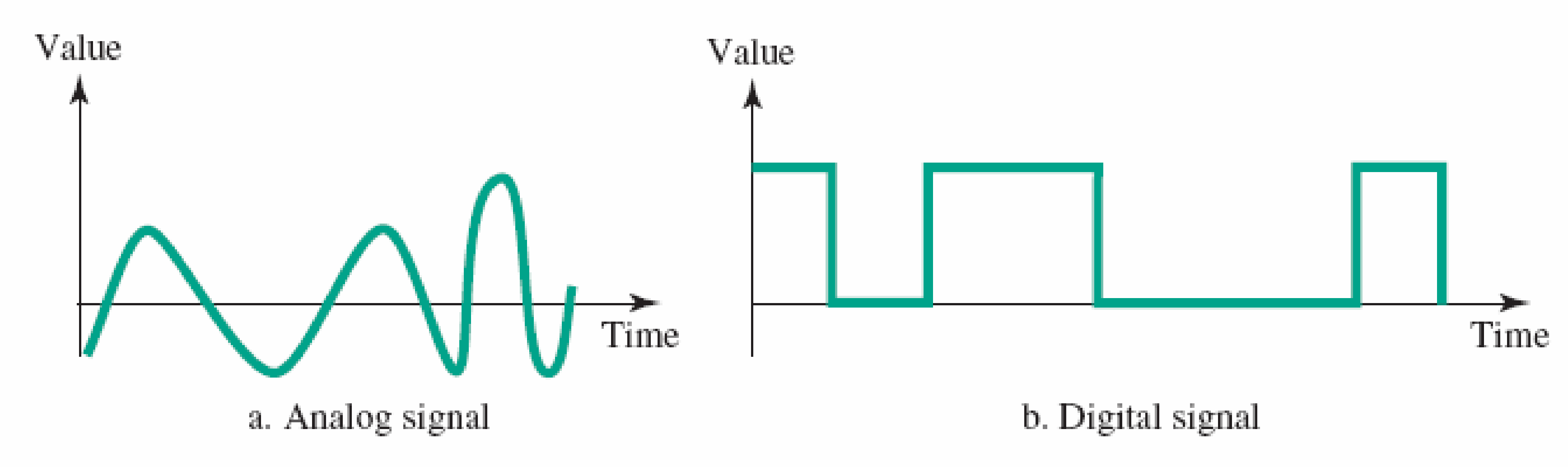
Data (pieces of information) can be analog or digital

- **Analog data** take on the continuous values in some interval. An example of analog data is the human voice.
- **Digital data** take on discrete values; examples are text and integers. Digital data is data stored in the memory of a computer in the form of 0s and 1s.

Signals

Signals are electric or electromagnetic representation of data.

Signaling is the physical propagation of the signal along a suitable medium. **Transmission** is the communication of data by the propagation and processing of signals.



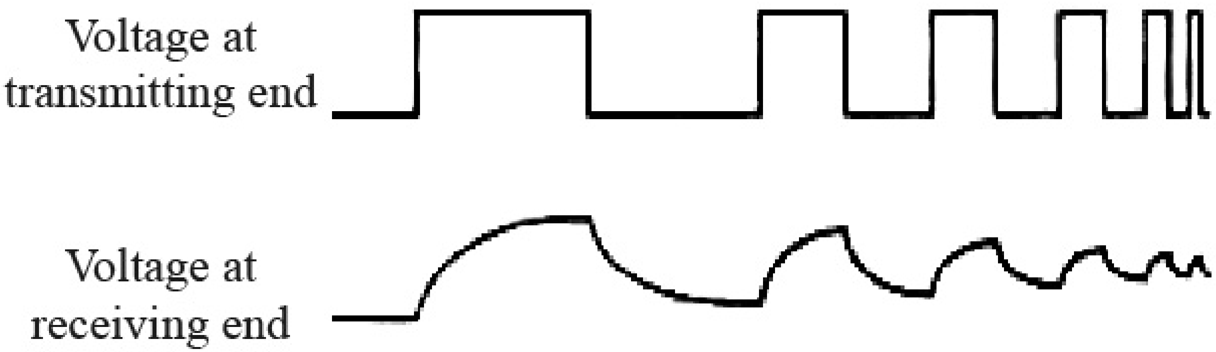
From [1]

Analog and Digital Signals

An analog signal is a continuously varying electromagnetic wave that may be propagated over a variety of media, depending on spectrum.

A digital signal is sequence of voltage pulses that may be transmitted over a medium (e.g. a constant +ve voltage may represent binary 1 whereas a –ve may represent binary 0).

Although digital signals are ***cheaper*** and less susceptible to ***noise interference***, it suffers from more ***attenuation*** than analog signals



From [2]

Attenuation of Digital Signals

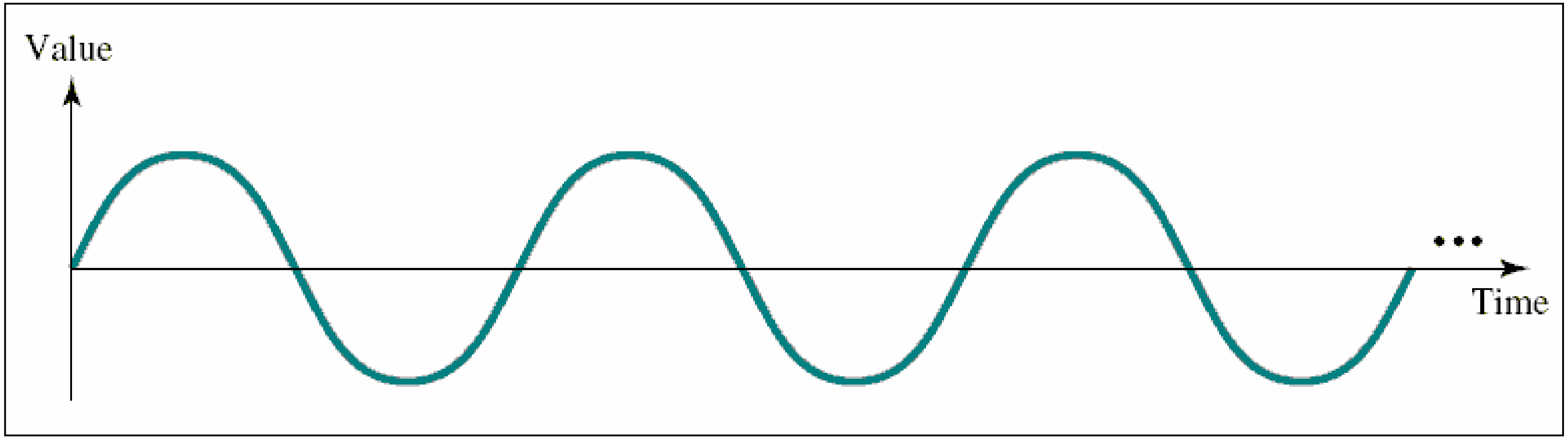
Periodic and Aperiodic Signals

A **periodic signal** completes a pattern within a measurable time frame, called a **period,** and repeats that pattern over subsequent identical periods. The completion of one full pattern is called a **cycle.**

An **aperiodic signal** changes without exhibiting a pattern or cycle that repeats over time.

Both analog and digital signals can be periodic or aperiodic. In data communication, however, we commonly use periodic analog signals and aperiodic digital signals to send data from one point to another.

Analog Signals – A sine wave



From [1]

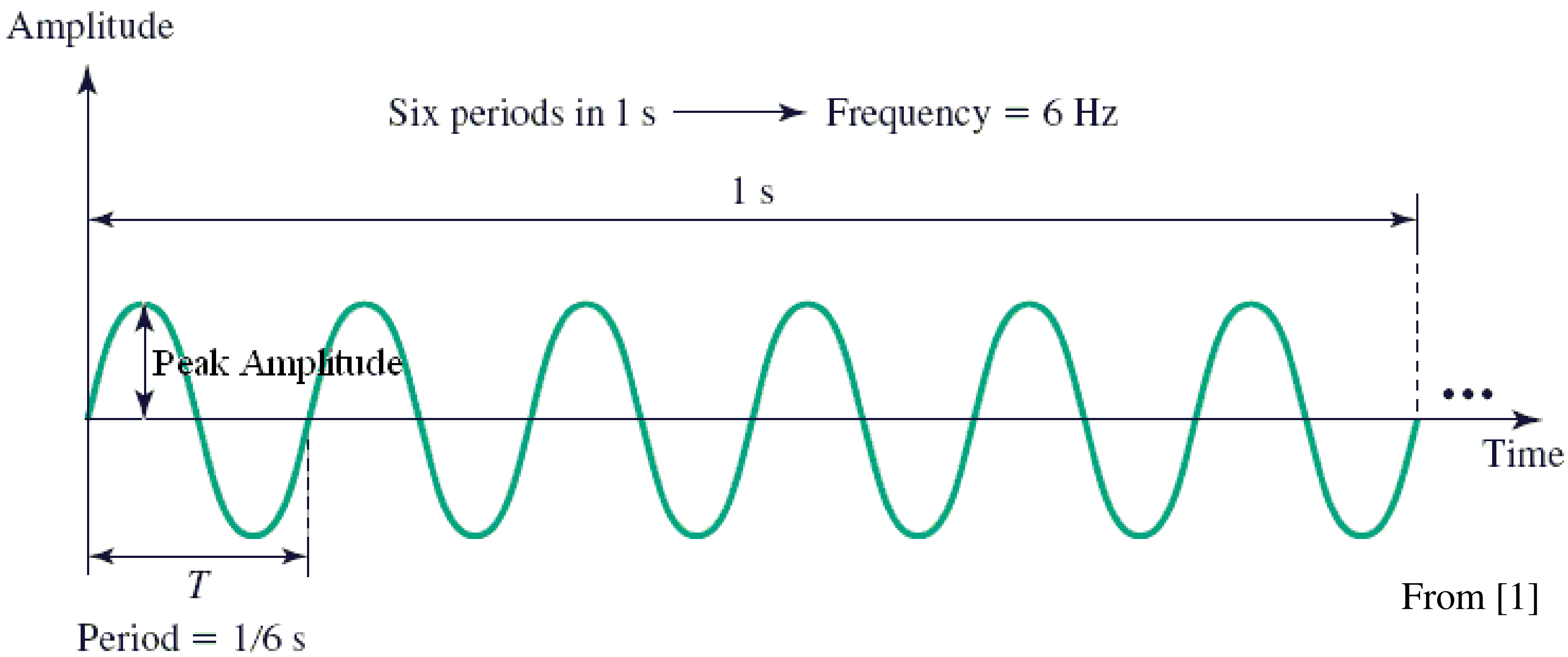
We can mathematically describe a sine wave as

$$s(t) = A \sin(2\pi f t + \theta)$$

Where *s* is the *instantaneous amplitude, A* the *peak amplitude*, *f* the *frequency,* and θ the *phase*

These last three characteristics fully describe a sine wave.

Characteristics of a Sine Wave



From [1]

Amplitude, Period and Frequency

The **amplitude** of a signal is a measure of its intensity.

The **peak amplitude** (*A* measured in volts) of a signal represents the absolute value of its highest intensity, proportional to the energy it carries.

Period (*T* measured in seconds) refers to the amount of time, in seconds, a signal needs to complete one cycle whereas **frequency** (*f* measured in Hertz) refers to the number of periods in one second,

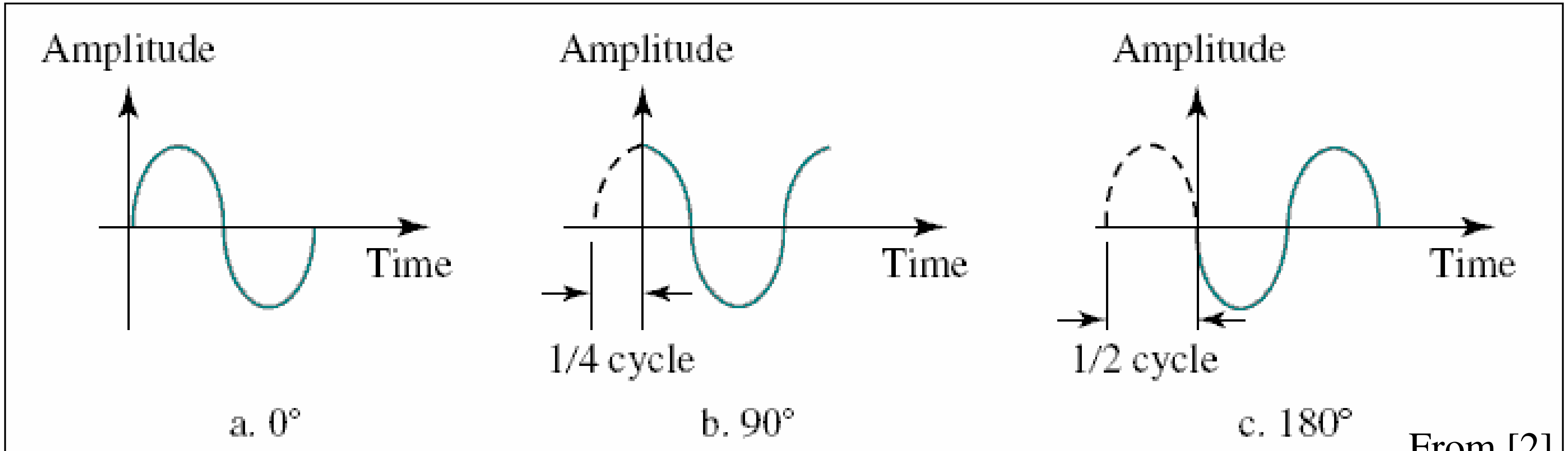
Therefore $f = 1 / T$ and $T = 1 / f$. Units of frequency and period:

Unit	*Equivalent*	*Unit*	*Equivalent*
Seconds (s)	1 s	hertz (Hz)	1 Hz
Milliseconds (ms)	10^{-3} s	kilohertz (KHz)	10^{3} Hz
Microseconds (μs)	10^{-6} s	megahertz (MHz)	10^{6} Hz
Nanoseconds (ns)	10^{-9} s	gigahertz (GHz)	10^{9} Hz
Picoseconds (ps)	10^{-12} s	terahertz (THz)	10^{12} Hz

From [2]

Phase (difference)

Phase describes the position of the waveform relative to time zero. Phase is measured in degrees or radians [360° is 2π rad; 1° is π/180 rad, and 1 rad is 180°].



From [2]

A phase shift of 360° corresponds to a shift of a complete period; a phase shift of 180° corresponds to a shift of one-half of a period; and a phase shift of 90° corresponds to a shift of one-quarter of a period

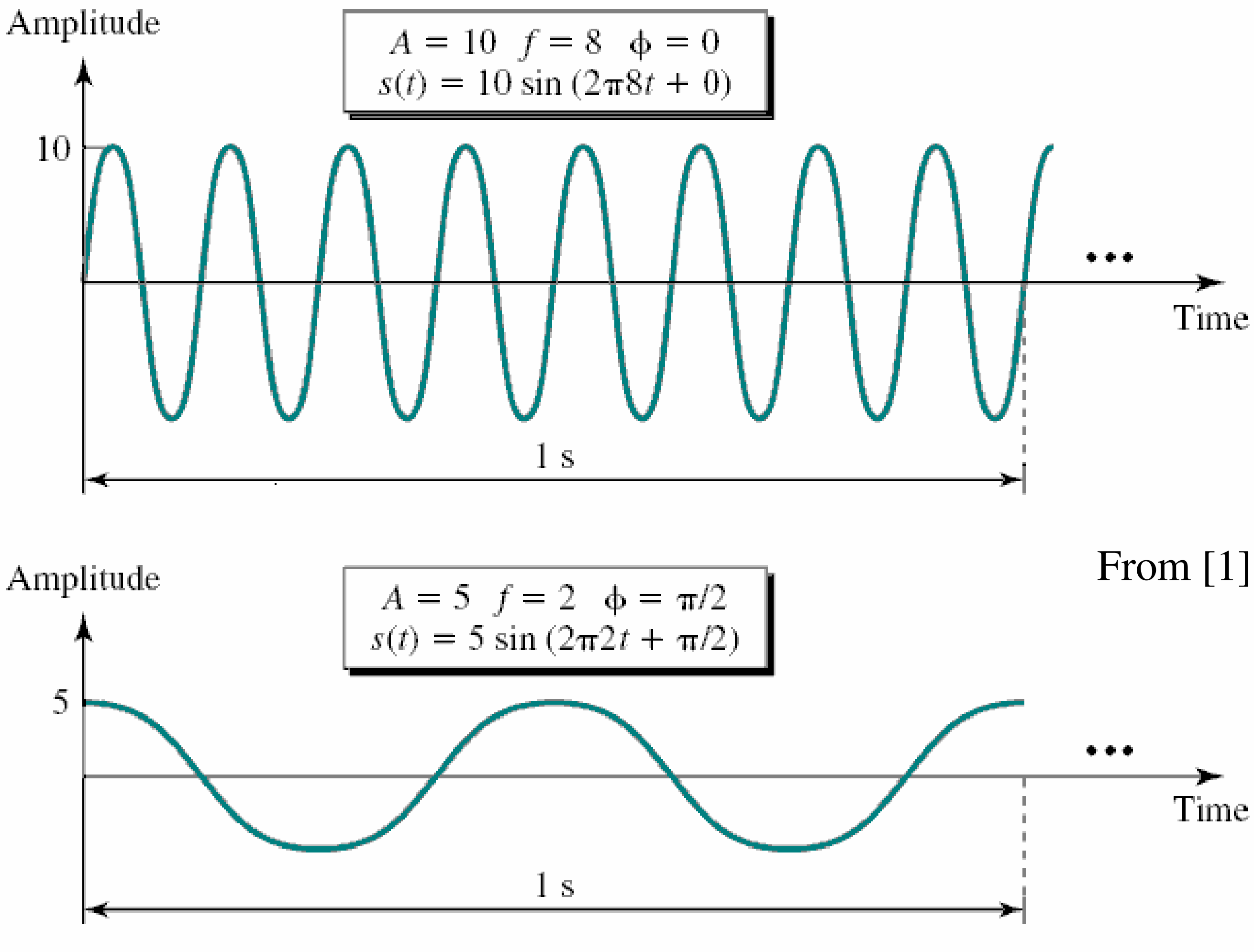
Time Domain Plots

These plot instantaneous amplitude with respect to frequency

Remember:

$$s(t) =$$

$$A \sin(2\pi f t + \theta)$$



From [1]

Composite Signals & Fourier Analysis

A single-frequency sine wave is not useful in data communications; to make it useful we need to change one or more of its characteristics, thereby making it a **composite signal**.

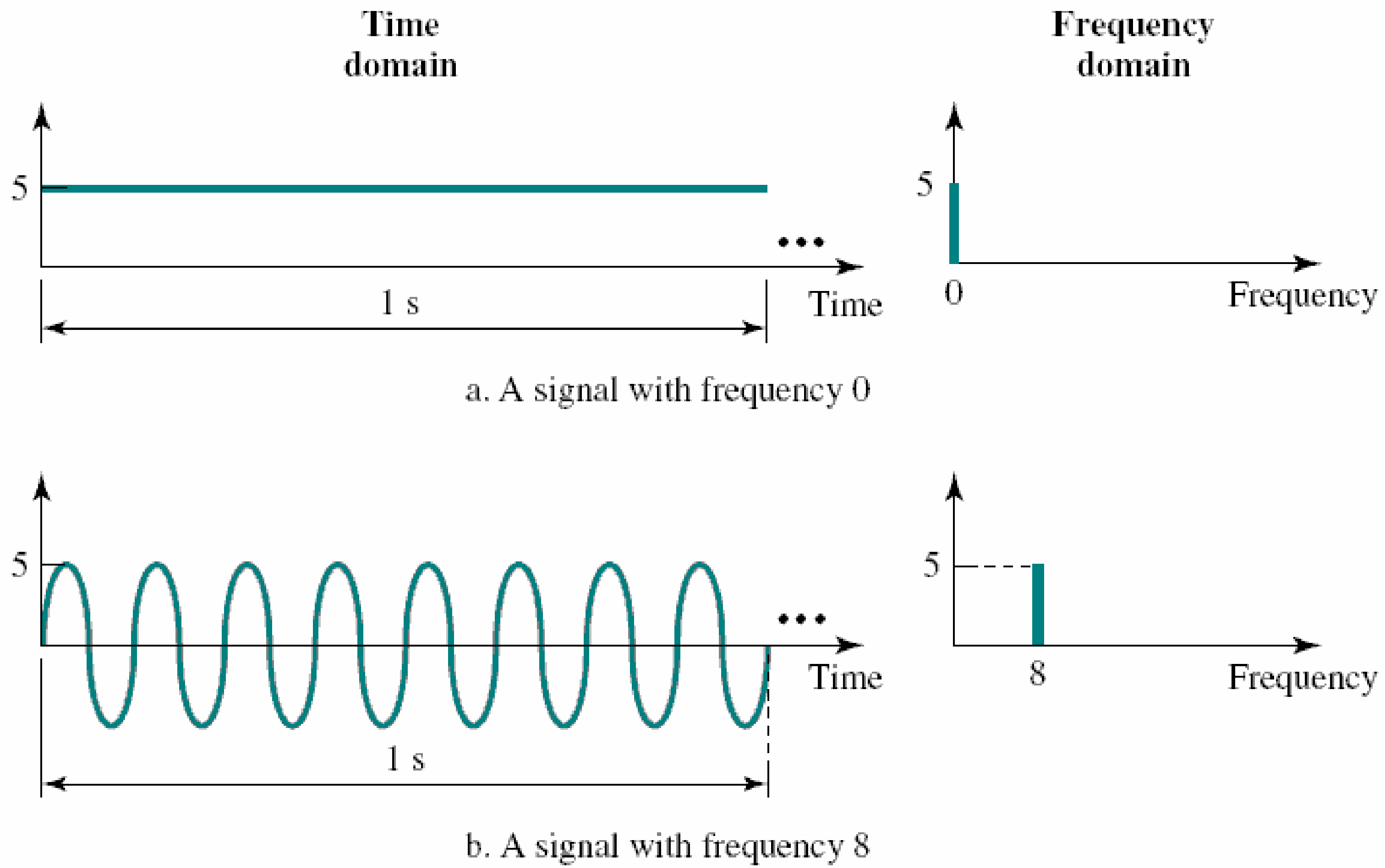
According to **Fourier analysis**, any composite signal can be represented as a combination of simple sine waves with different frequencies, phases, and amplitudes. i.e. we can write a sine-wave as

$$s(t) = A_1 \sin(2\pi f_1 t + \phi_1) + A_2 \sin(2\pi f_2 t + \phi_2) + A_3 \sin(2\pi f_3 t + \phi_3) + \cdots$$

From [1]

Frequency Domain Plots

A Frequency Domain Plot or Frequency Spectrum plots peak amplitude with respect to frequency. Analog signals are best represented by these.



From [1]

Example Frequency Domain Plot

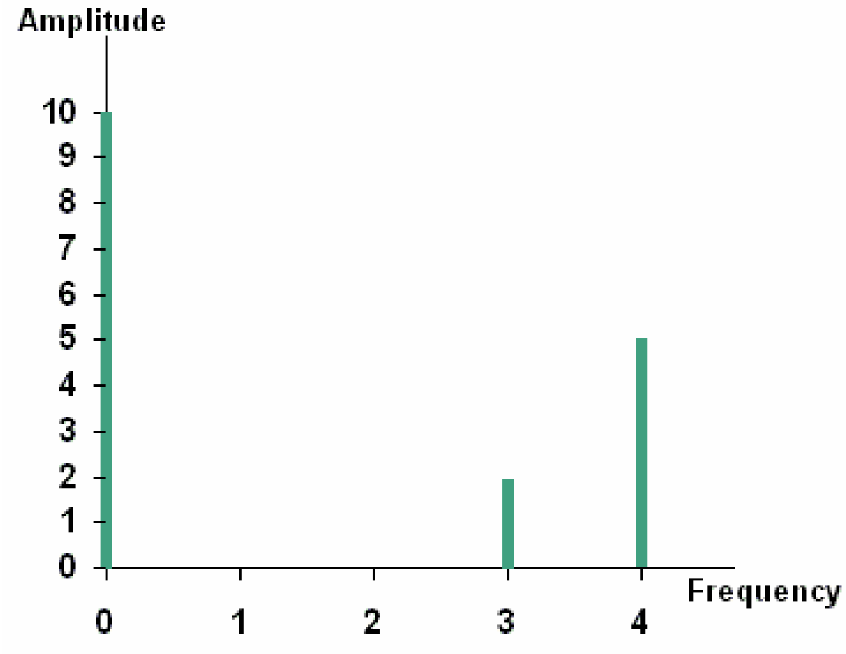
Draw the frequency spectrum for ***s(t)*** **= 10 + 5x*Sin*8*πt* + 2x*Sin*6*πt***

Answer: Remember *s(t)* of a composite signal can be represented as
$A_1 \sin(2\pi f_1 t + \theta_1) + A_2 \sin(2\pi f_2 t + \theta_2) + A_3 \sin(2\pi f_3 t + \theta_3)$....
Compare with $10 + 5\,Sin8\pi t + 2\,Sin6\pi t$.
This should give you values for (f_1, A_1), (f_2, A_2) and (f_3, A_3), which can now be plotted on the spectrum.

$A_1 \sin(2\pi f_1 t + \theta_1) = 10$
i.e. at $f_1 = 0$, $A_1 = 10$

$A_2 \sin(2\pi f_2 t + \theta_2) = 5\,Sin8\pi t$
i.e. at $f_2 = 4$, $A_2 = 5$

$A_3 \sin(2\pi f_3 t + \theta_3) = 2\,Sin6\pi t$
i.e. at $f_3 = 3$, $A_3 = 2$



Fundamental Frequency and Harmonics

When you have a composite wave with the equation $\boldsymbol{s(t)} =$ $\boldsymbol{A_1 \sin(2\pi f\ t + \theta_1) + A_2 \sin(2\pi 2f\ t + \theta_2) + A_3 \sin(2\pi 3f\ t + \theta_3)....}$ the first basic wave with frequency f is called the ***first harmonic*** or the ***fundamental frequency***. The one with frequency $2f$ is the ***second harmonic*** or the ***first overtone*** and so on. Overtones whose frequency is not an integer multiple of the fundamental are called ***inharmonic***.

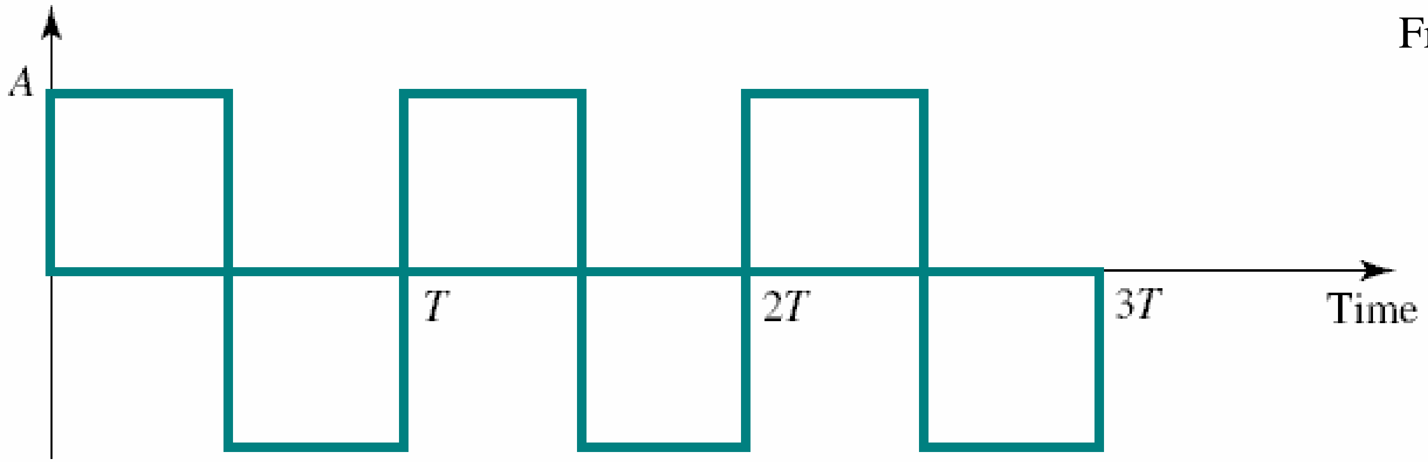
An example involving musical instruments

1f	440 Hz	fundamental frequency	first harmonic
2f	880 Hz	first overtone	second harmonic
3f	1320 Hz	second overtone	third harmonic
4f	1760 Hz	third overtone	fourth harmonic

Fourier Analysis of a Square Wave

According to **Fourier analysis,** we can prove that this signal can be decomposed into a series of sine waves as shown below.

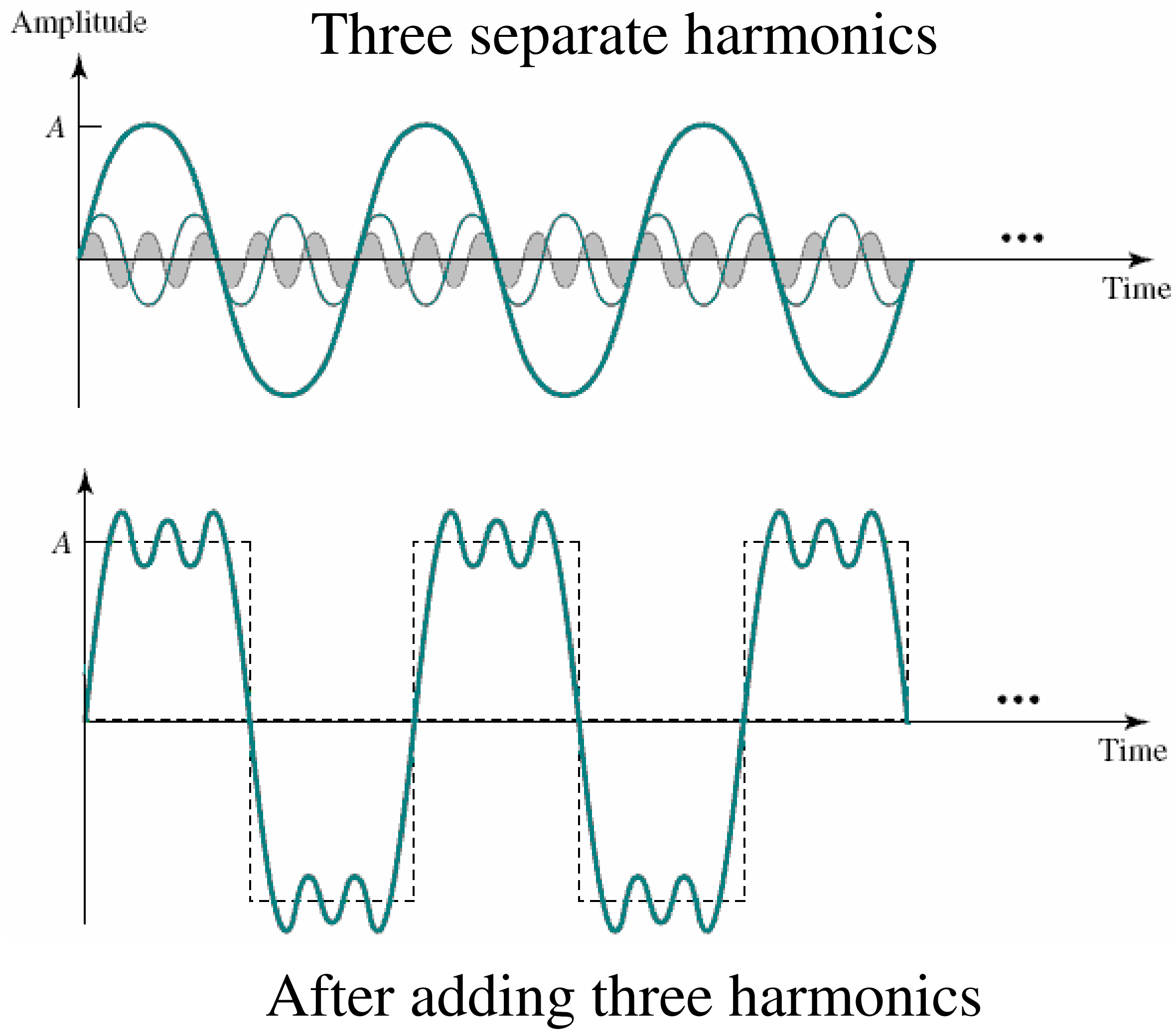
From [1]



$$s(t) = \frac{4A}{\pi} \sin 2\pi ft + \frac{4A}{3\pi} \sin [2\pi(3f)t] + \frac{4A}{5\pi} \sin [2\pi(5f)t] + \cdots$$

We have a series of sine waves with frequencies f, $3f$, $5f$, $7f$, . . . and amplitudes $4A/\pi$, $4A/3\pi$, $4A/5\pi$, $4A/7\pi$, and so on. The term with frequency f is dominant and is called the **fundamental frequency.** The term with frequency $3f$ is called the third harmonic, the term with frequency $5f$ is the fifth harmonic, and so on.

Adding Harmonics

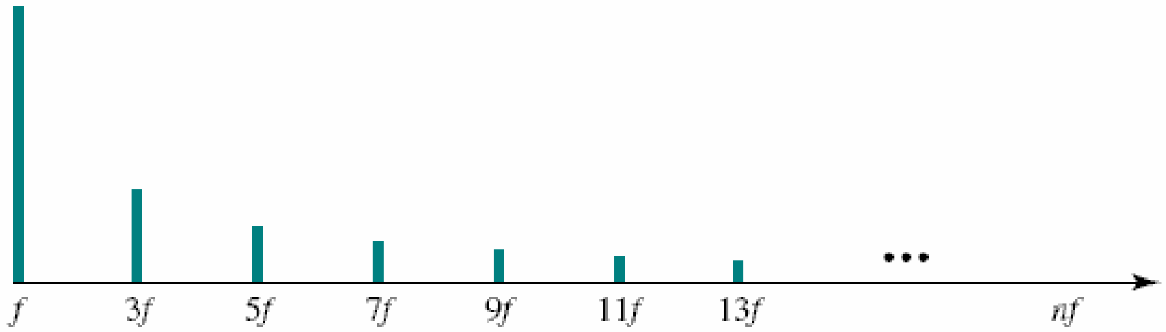


If we add these three harmonics, we do not get a square wave—we get something which is close, but not exact. For something more exact we need to add more harmonics

From [1]

Frequency Spectrum

The description of a signal using the frequency domain and containing all its components is called the **frequency spectrum** (or frequency domain plot) of that signal.



a. Frequency spectrum of a square wave

b. Frequency spectrum of an approximation with only three harmonics

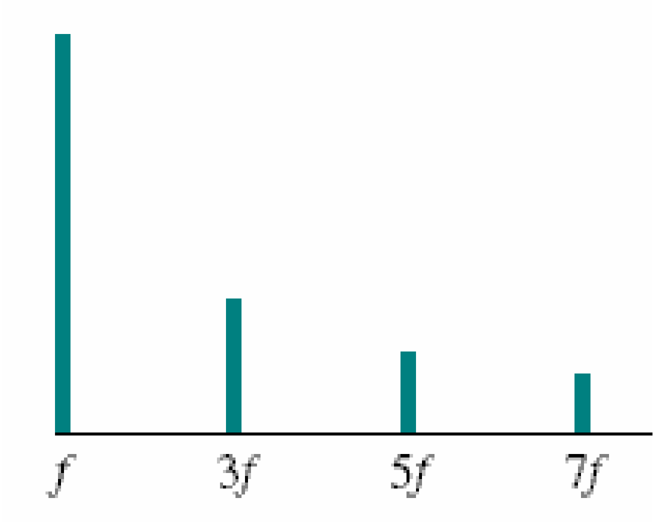
From [1]

Fourier Analysis of a Square Wave

To produce a square wave with Amplitude A and Frequency f, the required composite wave has the following components:

$4A/\pi \ \mathrm{Sin}\ [2\pi f t]$	f = Fundamental Freq =	1st Harmonic
$4A/3\pi \ \mathrm{Sin}\ [2\pi (3f)\, t]$	$3f$ = 2nd Overtone =	3rd Harmonic
$4A/5\pi \ \mathrm{Sin}\ [2\pi (5f)\, t]$	$5f$ = 4th Overtone =	5th Harmonic
$4A/7\pi \ \mathrm{Sin}\ [2\pi (7f)\, t]$	$7f$ = 6th Overtone =	7th Harmonic

....



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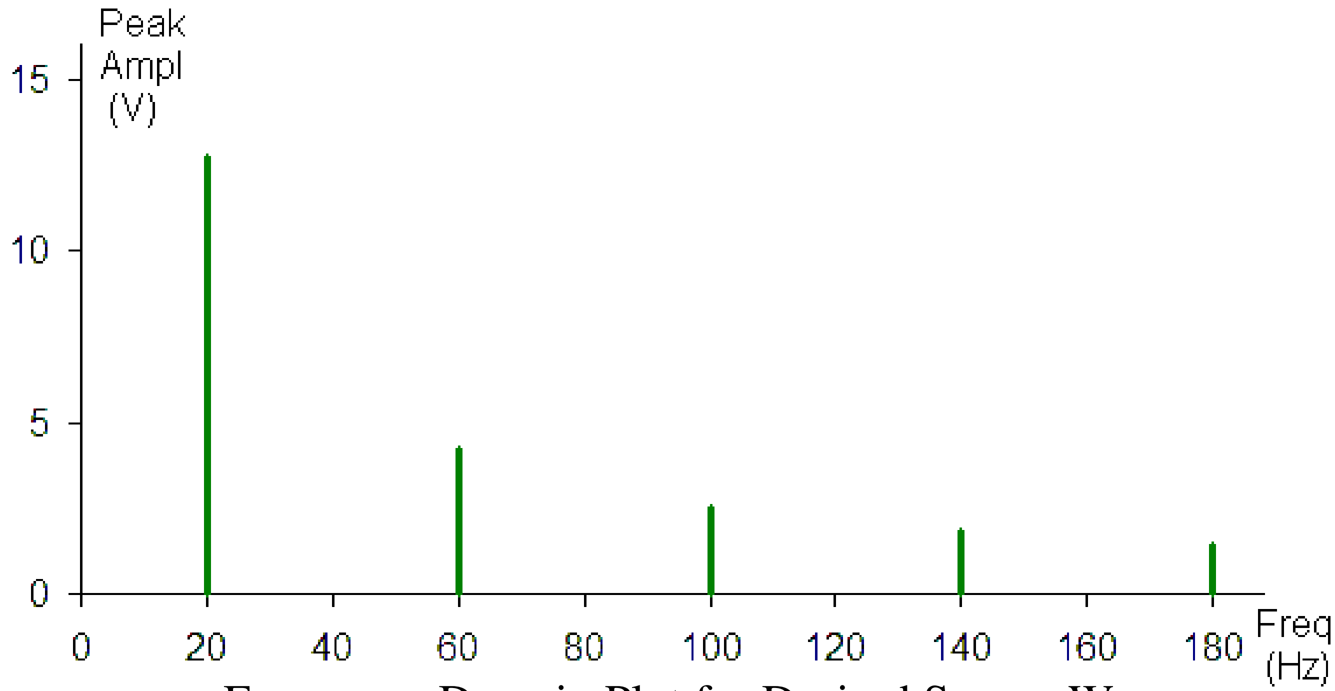
An Example - A square wave with amplitude 10V and frequency 20Hz produced using 5 harmonics has the following components:

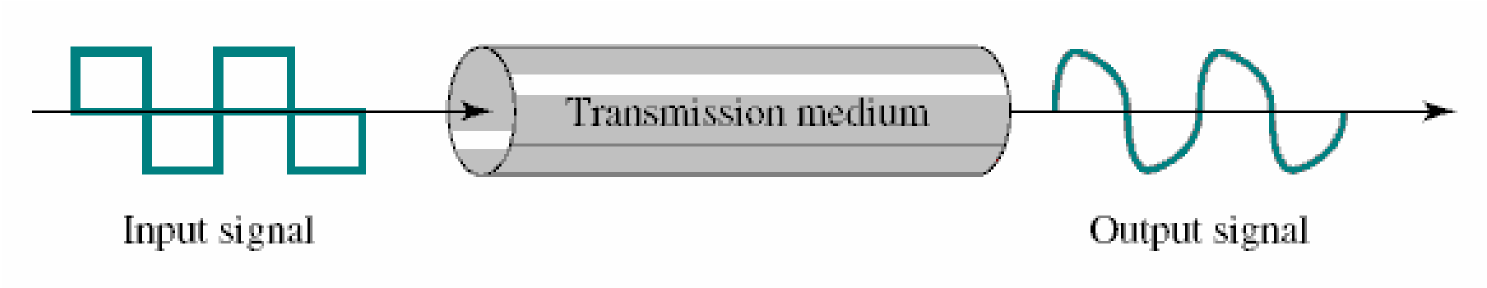
$4A/\pi \; \text{Sin}\,[2\,\pi f\, t]$	$A_1 = 4x10/\pi = 12.70$	$f_1 = 20Hz$
$4A/3\pi \; \text{Sin}\,[2\,\pi (3f)\, t]$	$A_2 = 4x10/3\pi = 4.24$	$f_2 = 60Hz$
$4A/5\pi \; \text{Sin}\,[2\,\pi (5f)\, t]$	$A_3 = 4x10/5\pi = 2.55$	$f_3 = 100Hz$
$4A/7\pi \; \text{Sin}\,[2\,\pi (7f)\, t]$	$A_4 = 4x10/7\pi = 1.82$	$f_4 = 140Hz$
$4A/9\pi \; \text{Sin}\,[2\,\pi (9f)\, t]$	$A_5 = 4x10/9\pi = 1.41$	$f_5 = 180Hz$



Frequency Domain Plot for Desired Square Wave

Signal Corruption

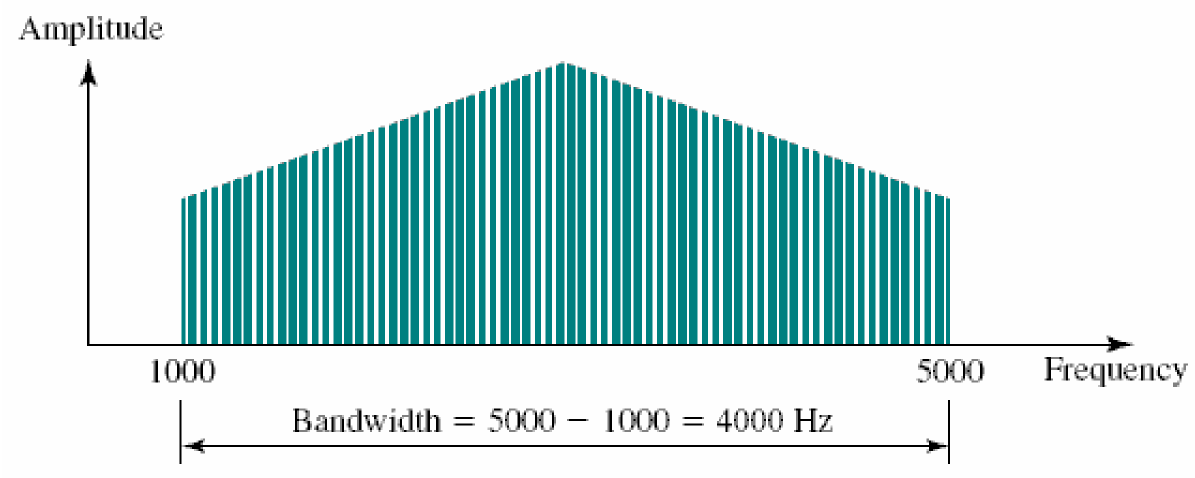
A transmission medium may pass some frequencies and may block or weaken others. This means that when we send a composite signal, containing many frequencies, at one end of a transmission medium, we may not receive the same signal at the other end. To maintain the integrity of the signal, the medium needs to pass every frequency.



From [1]

Bandwidth

The range of frequencies that a medium can pass is called **bandwidth**. The bandwidth is a range and is normally referred to as the difference between the highest and the lowest frequencies that the medium can satisfactorily pass.



From [1]

If the bandwidth of a medium does not match the spectrum of a signal, some of the frequencies are lost. Square wave signals have a spectrum that expands to infinity. No transmission medium has such a bandwidth. This means that passing a square wave through any medium will always deform the signal.

Digital Signals

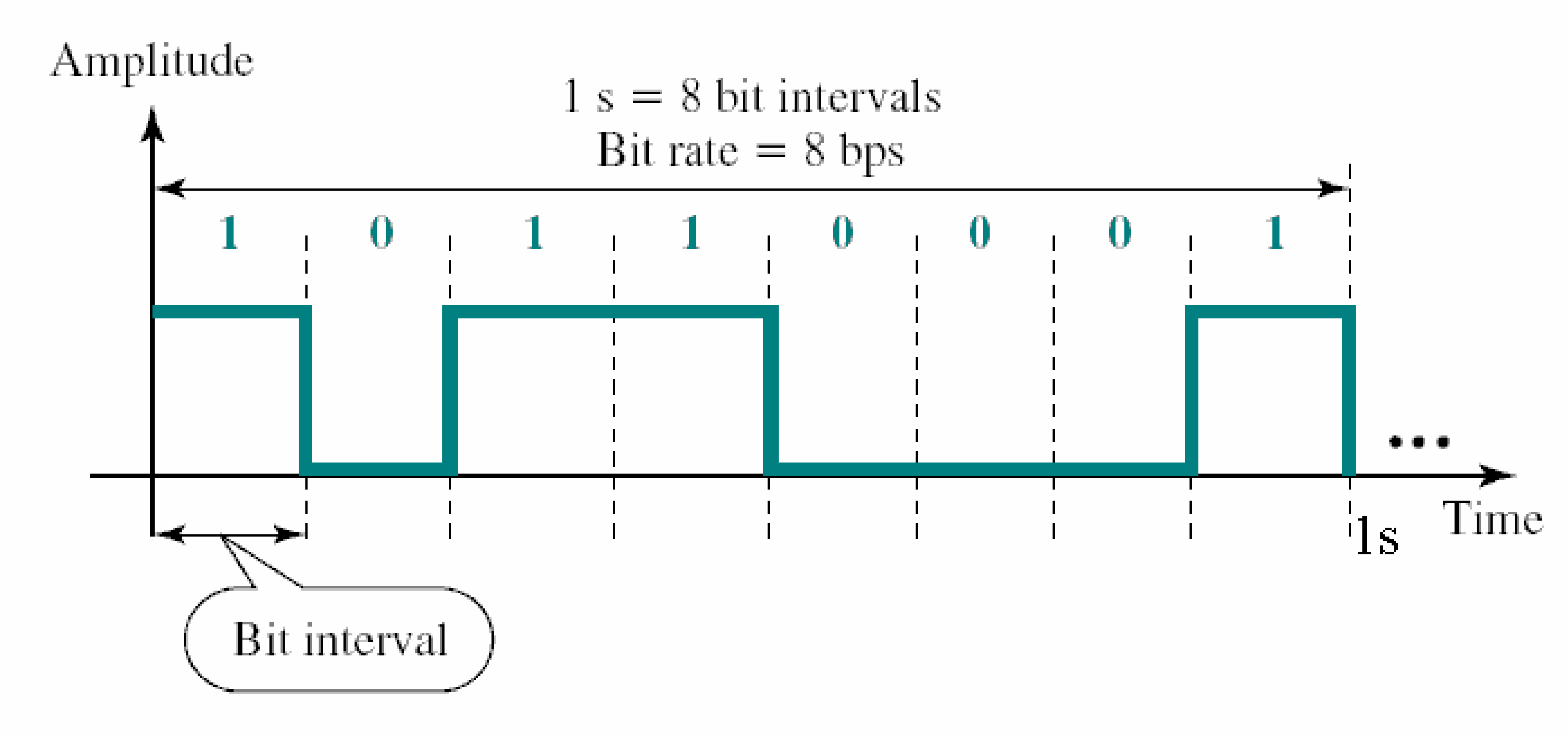
In addition to being represented by an analog signal, data can be represented by a digital signal. For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage. Remember that a digital signal is a composite signal with an infinite bandwidth.

Most digital signals are aperiodic, and thus period or frequency is not appropriate. Two new terms—*bit interval* (instead of *period*) and *bit rate* (instead of *frequency*)—are used to describe digital signals.

The **bit interval** is the time required to send one single bit.

The **bit rate** is the number of bit intervals per second. This means that the bit rate is the number of bits sent in 1 s, usually expressed in **bits per second (bps).**

Digital Signals



From [1]

Digital Signals through different Bandwidth

Digital Signal Through a Wide-Bandwidth Medium

We can send a digital signal through them (e.g. coaxial cables for a LAN). Some of the frequencies are blocked by the medium, but still enough frequencies are passed to preserve a decent signal shape.

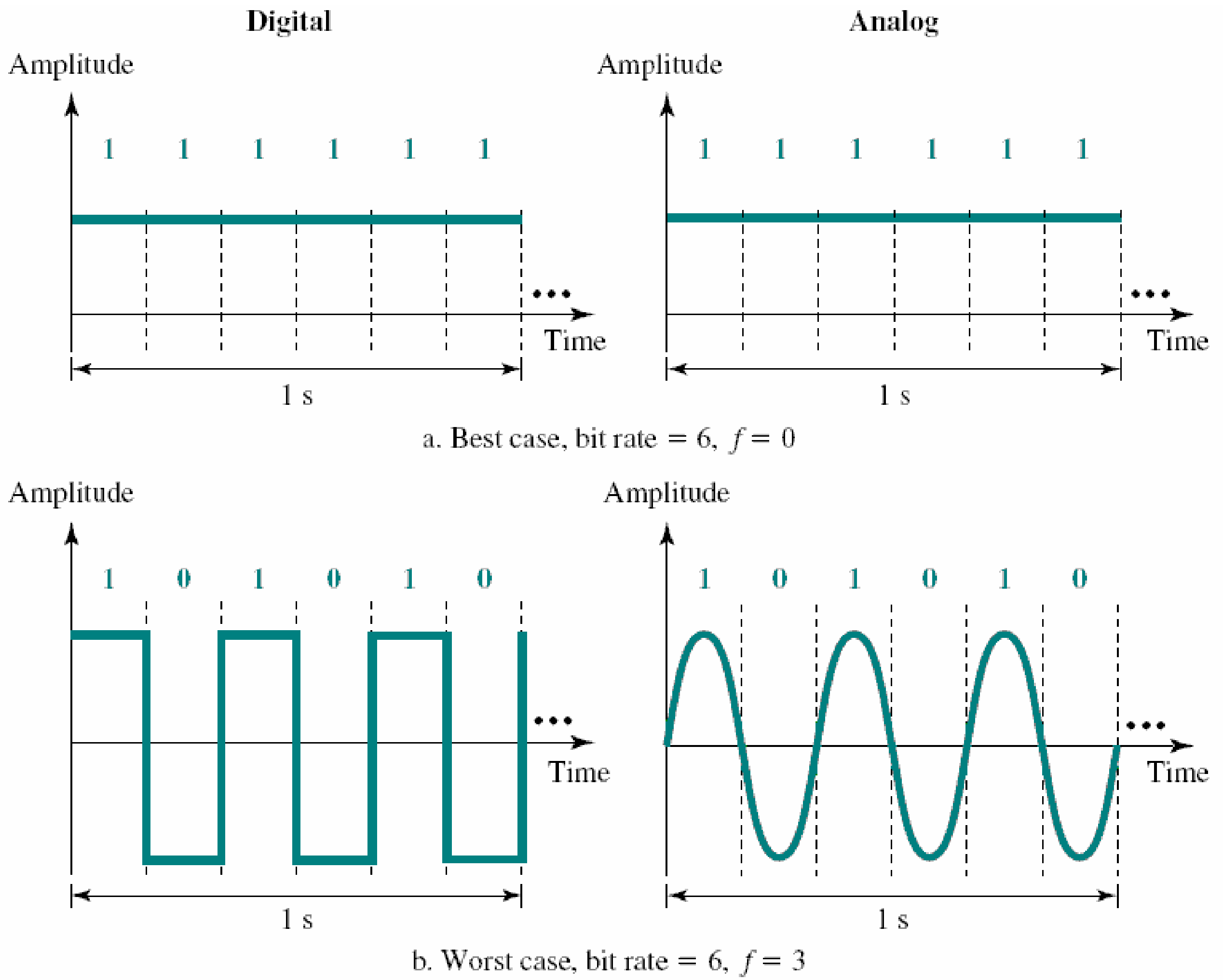
Digital Signal Through a Band-Limited Medium

We can send digital signal through them (e.g. telephone lines for the internet). There is a relationship between minimum required bandwidth B in hertz if we want to send n bps.

Using One Harmonic

If we need to simulate this digital signal of data rate 6 bps, sometimes we need to send a signal of frequency 0, sometimes 1, sometimes 2, and sometimes 3 Hz

To send *n* bps through an analog channel using an approximation, we need a bandwidth *B* such that $B = n / 2$.



From [1]

Using More Harmonics

A one frequency signal may not be adequate, since the analog and digitals signals may look different and the receiver may not recognise it correctly. To improve the shape of the signal for better communication, particularly for high data rates, we need to add some harmonics.

We need to add some odd harmonics. If we add the third harmonic to each case, we need $B = n/2 + 3n/2 = 4n/2$ Hz; if we add third and fifth harmonics, we need $B = n/2 + 3n/2 + 5n/2 = 9n/2$ Hz; and so on. In other words, we have

$$B >= n/2 \qquad \text{or} \qquad n <= 2B$$

Analog and Digital Bandwidth

The bit rate and the bandwidth are proportional to each other.

Bandwidth Requirements:

From [1]

Bit Rate	*Harmonic 1*	*Harmonics 1, 3*	*Harmonics 1, 3, 5*	*Harmonics 1, 3, 5, 7*
n = 1 Kbps	B = 500 Hz	B = 2 KHz	B = 4.5 KHz	B = 8 KHz
n = 10 Kbps	B = 5 KHz	B = 20 KHz	B = 45 KHz	B = 80 KHz
n = 100 Kbps	B = 50 KHz	B = 200 KHz	B = 450 KHz	B = 800 KHz

The analog bandwidth of a medium is expressed in hertz; the digital bandwidth, in bits per second.

Telephone lines have a bandwidth of 3 to 4 KHz for the regular user i.e. 6000 to 8000bps; but we know that sometimes we send more than 30,000 bps. This is achieved by the modem with modulation techniques that allow the representation of multiple bits in one single period of an analog signal.

Low-pass and Band-pass Channels

A channel or a link is either low-pass or band-pass.

A **low-pass channel** has a bandwidth with frequencies between 0 and f. The lower limit is 0, the upper limit can be any frequency (including infinity).

A **band-pass channel** has a bandwidth with frequencies between f_1 and f_2.

A ***digital signal*** theoretically needs a bandwidth between 0 and infinity (i.e. a low-pass channel). The lower limit (0) is fixed; the upper limit (infinity) can be relaxed if we lower our standards by accepting a limited number of harmonics.

An ***analog signal*** requires a band-pass channel since normally it has a narrower bandwidth than a digital signal. we can always shift a signal with a bandwidth from f_1 to f_2 to a signal with a bandwidth from f_3 to f_4 as long as the width of the bandwidth remains the same.

Data Rate Limits and Nyquist Bit Rate

Data rate depends on three factors:

1. The bandwidth available
2. The levels of signals we can use
3. The quality of the channel (the level of the noise).

For ***a noiseless channel***, the **Nyquist bit rate** formula defines the theoretical maximum bit rate:

$$\textbf{BitRate} = \textbf{2} \times \textbf{Bandwidth} \times \textbf{log}_2\, L$$

(L is the number of signal levels used to represent data)

Nyquist BitRate = 2 x *Bandwidth* x log_2 L

Example

Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. The maximum bit rate can be calculated as:

$$\text{BitRate} = 2 \times 3000 \times \log_2 2 = 6000 \text{ bps}$$

Example

Consider the same noiseless channel, transmitting a signal with four signal levels (for each level, we send two bits). The maximum bit rate can be calculated as:

$$\text{BitRate} = 2 \times 3000 \times \log_2 4 = 12{,}000 \text{ bps}$$

From [1]

Data Rate Limits and Shannon Capacity

In reality a channel is always noisy. The **Shannon capacity** formula is used to determine the theoretical highest data rate for a ***noisy channel***:

$$\textbf{Capacity} = \textbf{Bandwidth} \times \log_2 (1 + \textbf{SNR})$$

$$= \textbf{Bandwidth} \times 3.32 \log_{10} (1 + \textbf{SNR})$$

SNR is the ***signal-to-noise ratio*****,** and Capacity is the capacity of the channel in bits per second. The signal-to-noise ratio is the statistical ratio of the power of the signal to the power of the noise. In practice, the SNR is expressed in dB. $(SNR)_{dB} = 10 \log_{10}(P_{signal}/P_{noise})$, but it must be made unitless before applying Shannon's formula

Shannon Capacity = Bandwidth × $\log_2$ (1 + SNR)

Example

We can calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000 Hz (300 Hz to 3300 Hz). The signal-to-noise ratio is usually 3162. For this channel the capacity is calculated as

$$C = B \log_2 (1 + SNR) = 3000 \log_2 (1 + 3162) = 3000 \log_2 (3163)$$
$$C = 3000 \times 11.62 = 34{,}860 \text{ bps}$$

Example

Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. In other words, the noise is so strong that the signal is faint. For this channel the capacity is calculated as

$$C = B \log_2 (1 + SNR) = B \log_2 (1 + 0) = B \log_2 (1) = B \times 0 = 0$$

This means that the capacity of this channel is zero regardless of the bandwidth. In other words, we cannot receive any data through this channel.

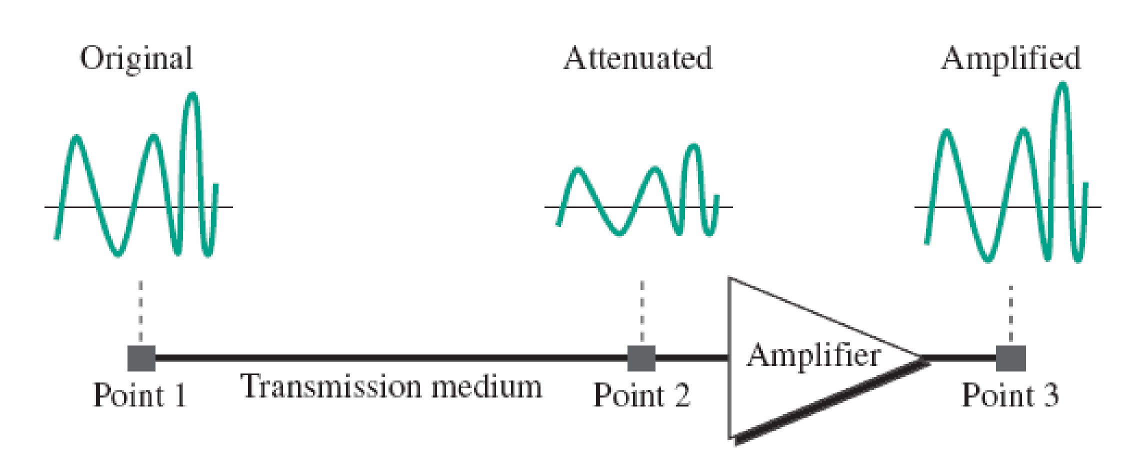
From [1]

Transmission Impairment - Attenuation

Signals travel through transmission media, which are not perfect. The imperfections cause **impairment** in the signal i.e. the signal at the beginning and end of the medium are not the same.

3 types of impairment usually occur: attenuation, distortion & noise

Attenuation means loss of energy. When a signal, simple or composite, travels through a medium, it loses some of its energy so that it can overcome the resistance of the medium. To compensate for this loss amplifiers are used to ***amplify*** (strengthen) the signal:



From [1]

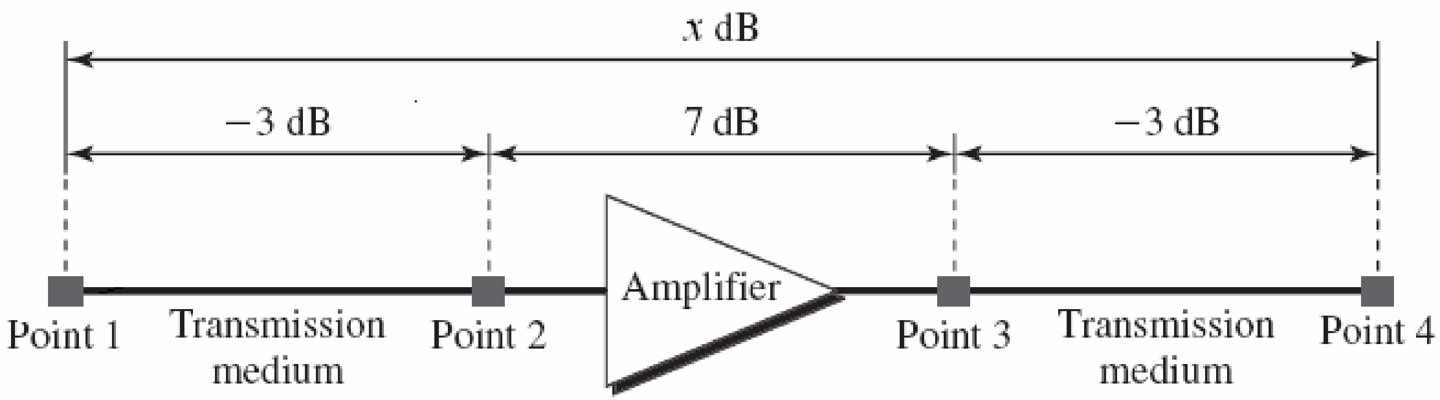
Attenuation, Amplification and the Decibel

Amplification simply refers to the strengthening of a signal whereas **attenuation** refers to its weakening.

The **decibel (dB)** measures the relative strengths of two signals or a signal at two different points. Note that the decibel is negative if a signal is attenuated and positive if a signal is amplified.

$$A_{dB} = 10 \log_{10} (P_2 / P_1)$$

where P_1 and P_2 are the powers of a signal at points 1 and 2, respectively

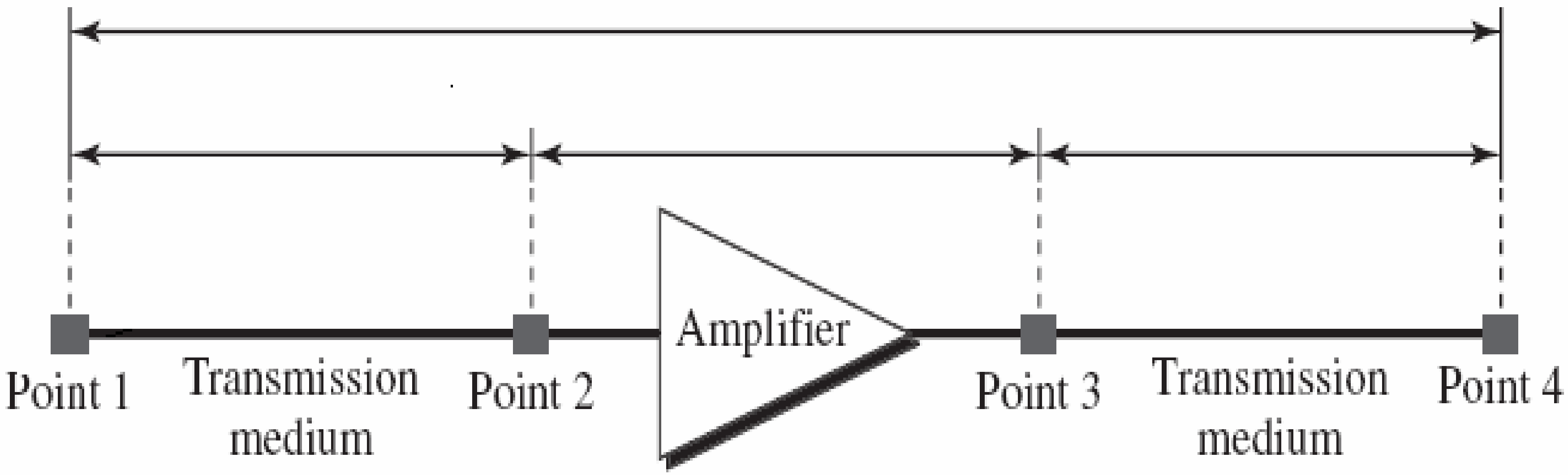


In the diagram above $x = -3 + 7 - 3 = 1$

Calculating Amplification/Attenuation

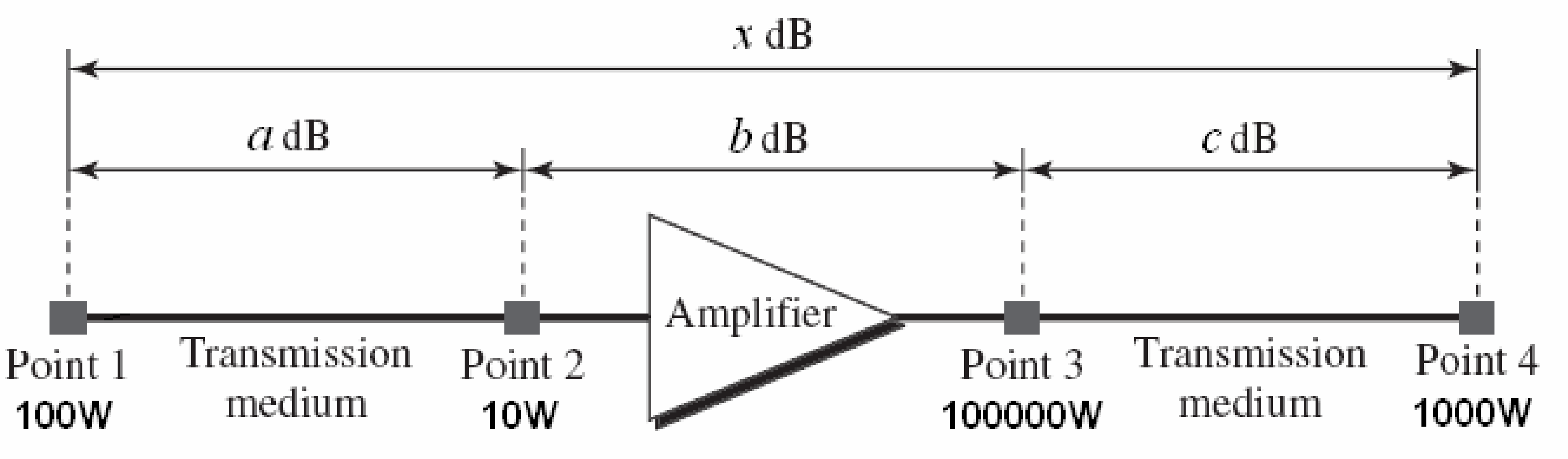
$$A_{P1toP2} = 10 \log_{10} (P_2 / P_1)$$

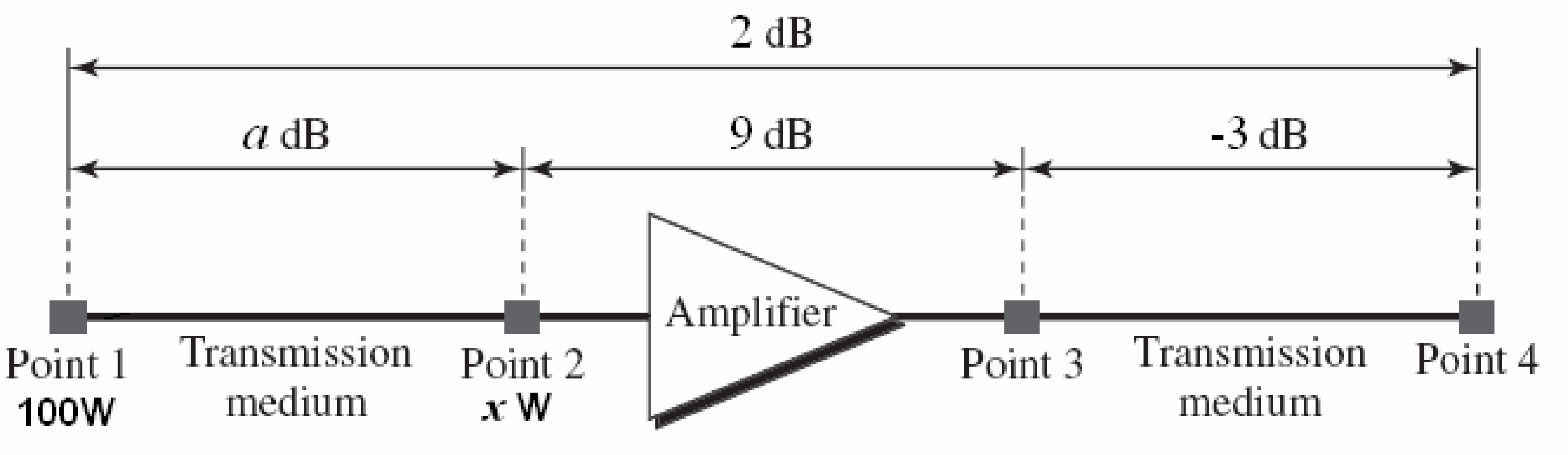
A is measured in decibels (dB)



$$A_{P_1 \text{ to } P_2} = 10 \log_{10} (P_2 / P_1) \text{ dB}$$

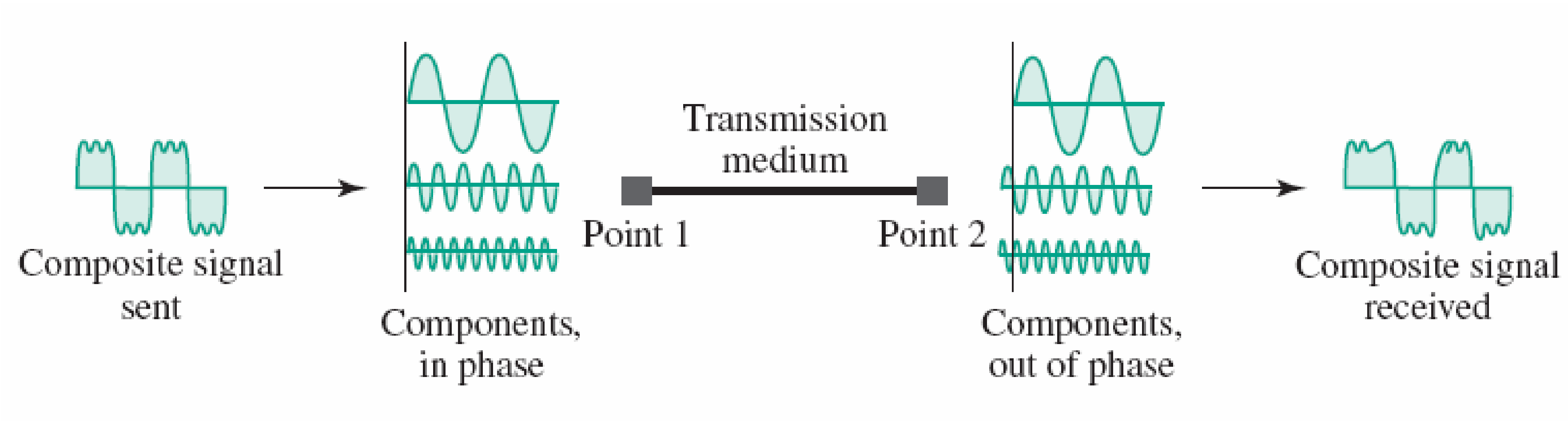
Exercise: Find the values of *a*, *b*, *c* and *x*





Transmission Impairment - Distortion

Distortion means that the signal changes its form or shape. Distortion occurs in a composite signal, made of different frequencies. Each signal component has its own propagation speed through a medium and, therefore, its own delay in arriving at the final destination.



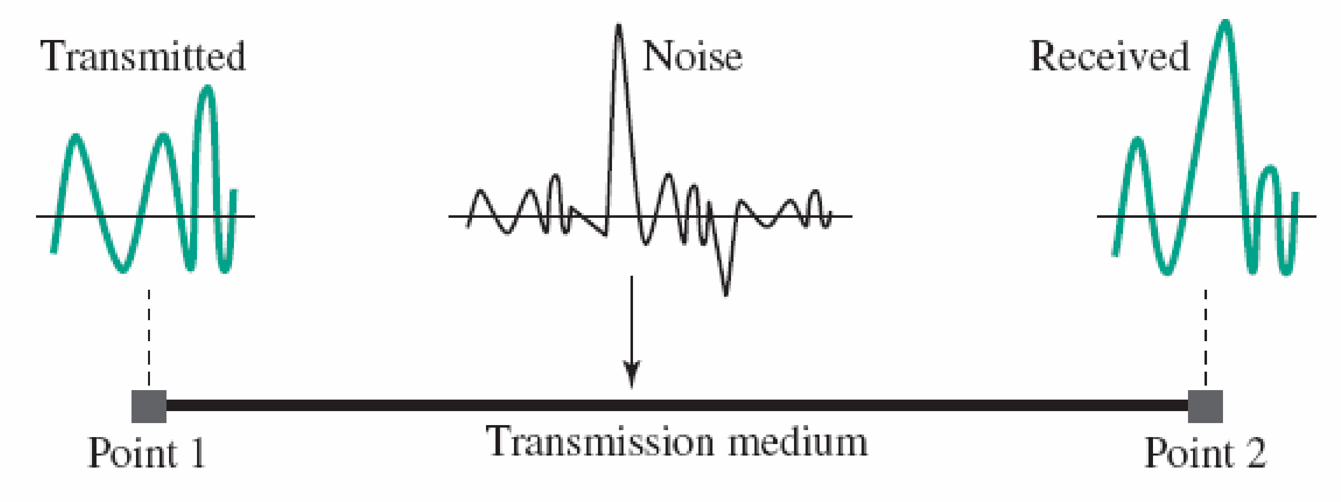
From [1]

Transmission Impairment - Noise

Noise is another problem. Several types of noise such as thermal noise, induced noise, crosstalk, and impulse noise may corrupt the signal.

Thermal noise is the random motion of electrons in a wire which creates an extra signal not originally sent by the transmitter.

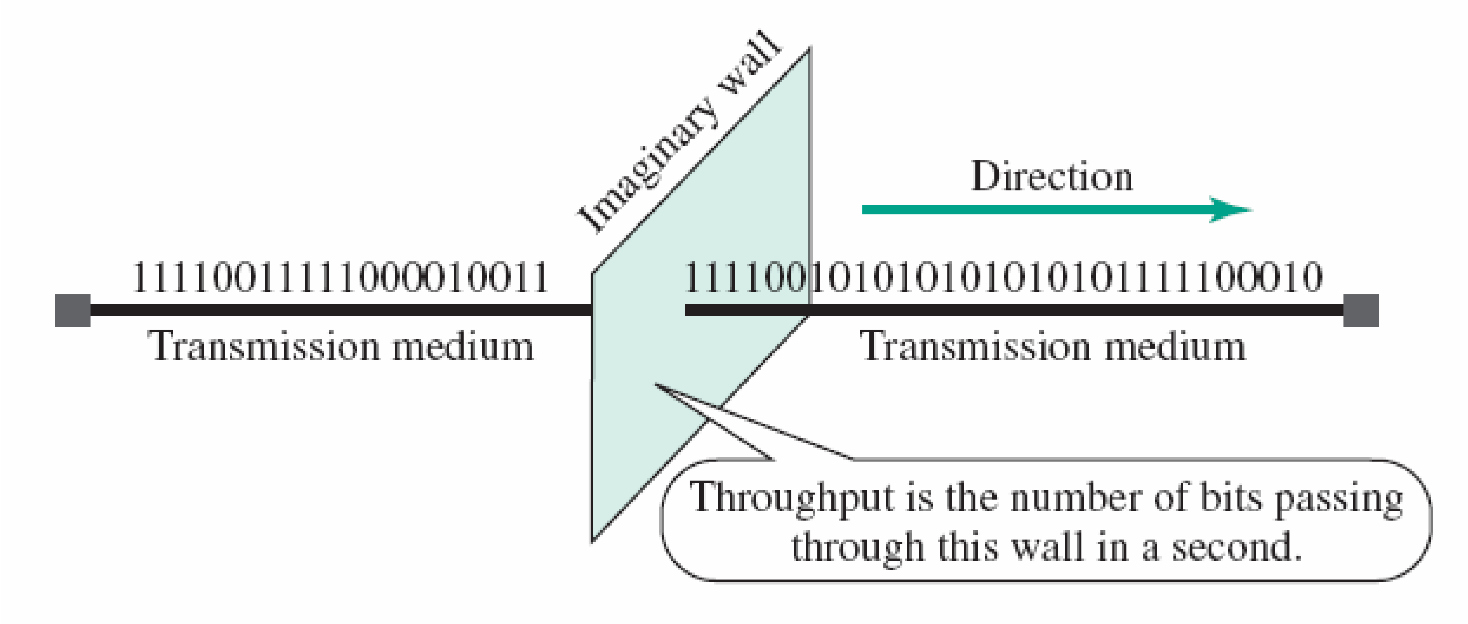
Induced noise comes from sources such as motors and appliances. These devices act as a sending antenna and the transmission medium acts as the receiving antenna.



From [1]

Throughput

The **throughput** is the measurement of how fast data can pass through an entity (such as a point or a network).



From [1]

Propagation Speed and Time

Propagation speed measures the distance a signal or a bit can travel through a medium in one second. The propagation speed of electromagnetic signals depends on the medium and on the frequency of the signal. For example, in a vacuum, light is propagated with a speed of 3×10^8 m/s. It is lower in air. It is much lower in a cable.

Propagation time measures the time required for a signal (or a bit) to travel from one point of the transmission medium to another. The propagation time is calculated by dividing the distance by the propagation speed.

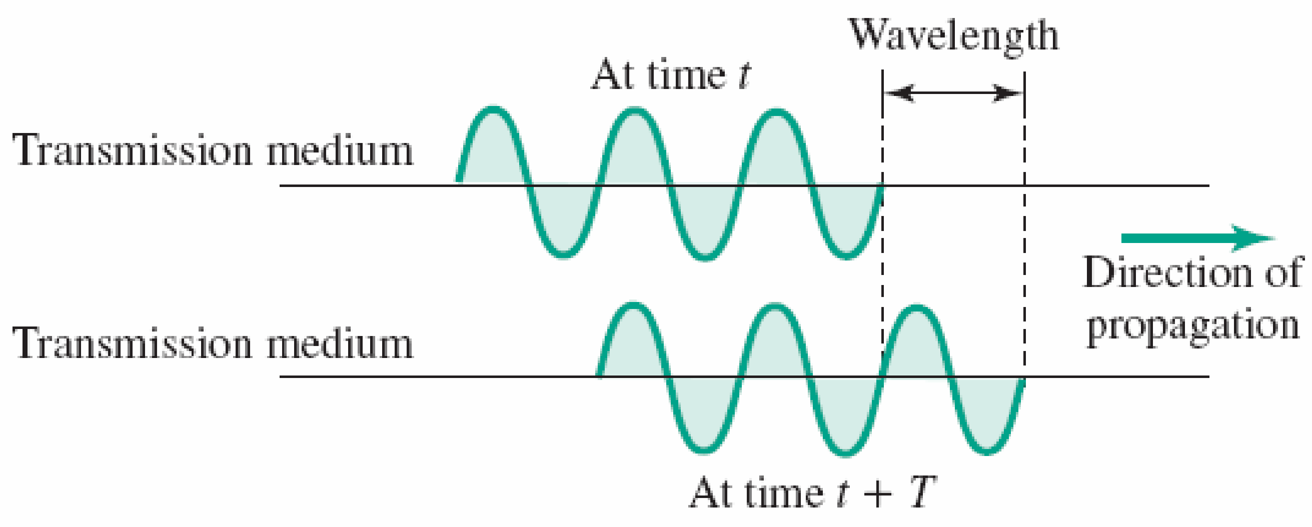
Propagation time = Distance/Propagation speed

Wavelength

Wavelength is the distance between repeating units of a propagating wave of a given frequency. While the frequency of a signal is independent of the medium, the wavelength depends on both the frequency and the medium.

Wavelength = Propagation speed × Period

If we represent wavelength by λ, propagation speed by c (speed of light), and frequency by f, we get $\lambda = c/f$



From [1]

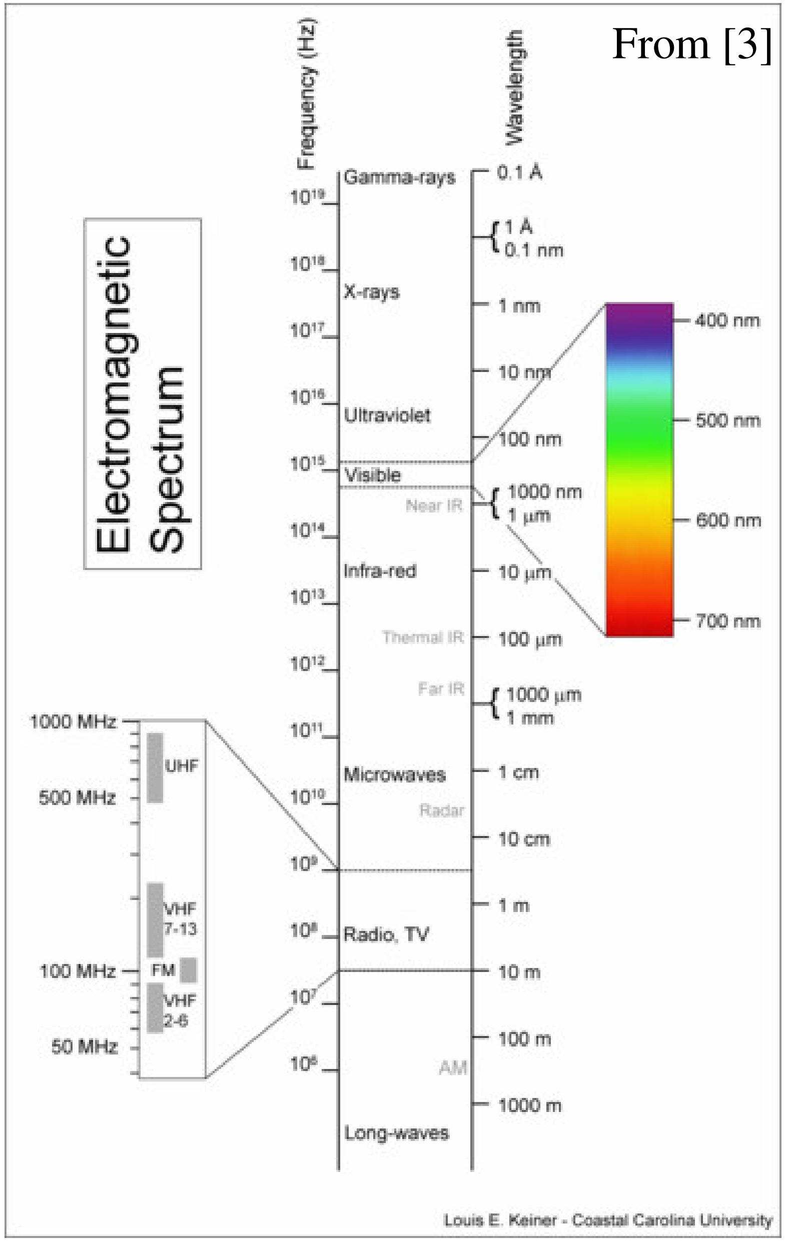
The Electromagnetic Spectrum

Electromagnetic Radiation is a self-propagating wave in space with electric and magnetic components. EM radiation carries energy and momentum, which may be imparted when it interacts with matter.

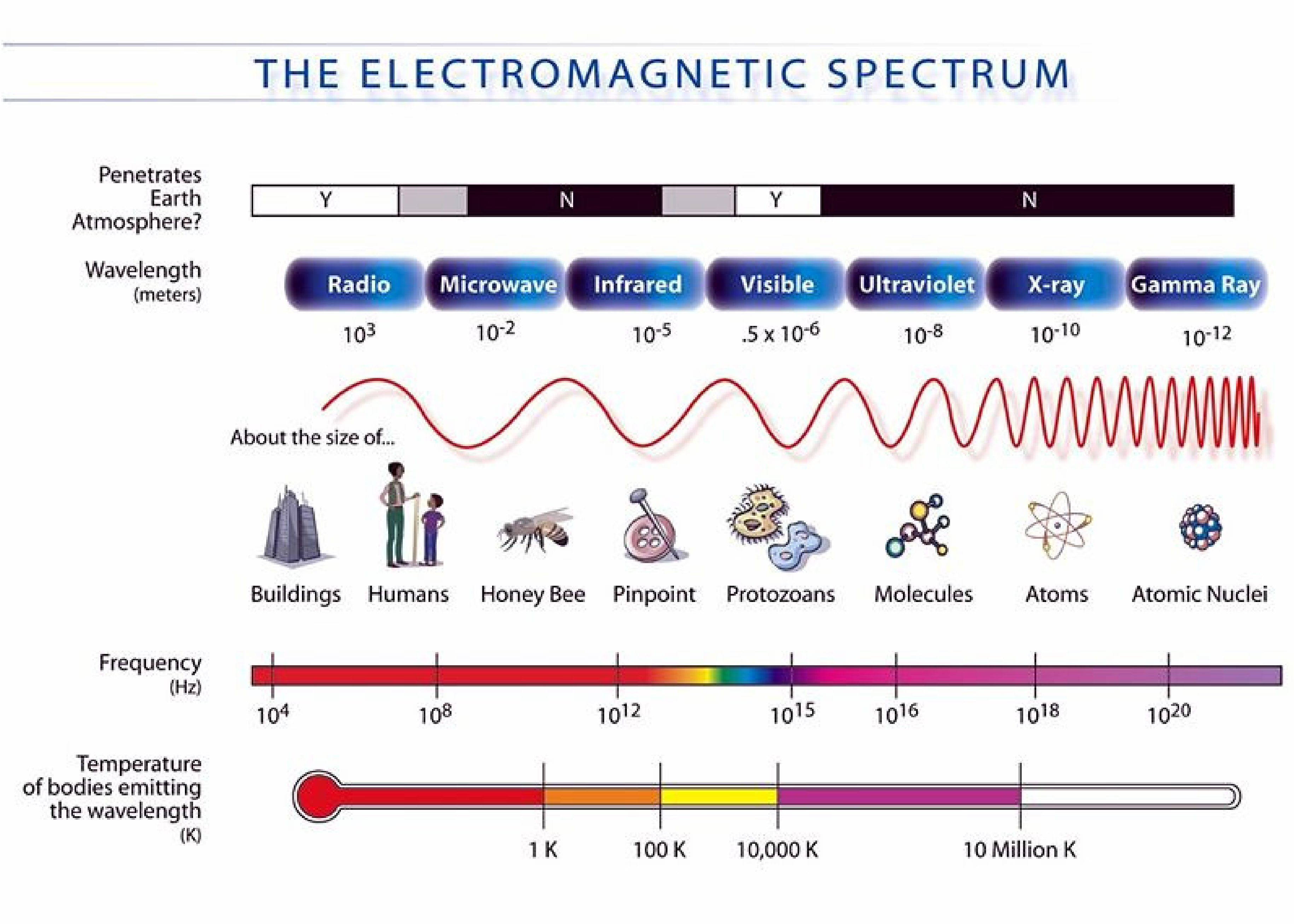
EM-waves are classified according to their varying frequencies (and consequent wavelengths). They include in order of increasing frequency, radio waves, microwaves, infrared radiation, visible light, ultraviolet radiation, X-rays and gamma rays.

All EM-radiation travel through vacuum at a propagation speed of 299,792,458 m/s (approx. 3 x 10^8 m/s), i.e. the speed of light.

The **electromagnetic spectrum** is the range of all possible electromagnetic radiation.



From [3]


THE ELECTROMAGNETIC SPECTRUM
Penetrates Earth Atmosphere?
Y
N
Y
N
Wavelength (meters)
Radio
Microwave
Infrared
Visible
Ultraviolet
X-ray
Gamma Ray
10^3
10^{-2}
10^{-5}
$.5 \times 10^{-6}$
10^{-8}
10^{-10}
10^{-12}
About the size of...
Buildings
Humans
Honey Bee
Pinpoint
Protozoans
Molecules
Atoms
Atomic Nuclei
Frequency (Hz)
10^4
10^8
10^{12}
10^{15}
10^{16}
10^{18}
10^{20}
Temperature of bodies emitting the wavelength (K)
1 K
100 K
10,000 K
10 Million K

From [3]