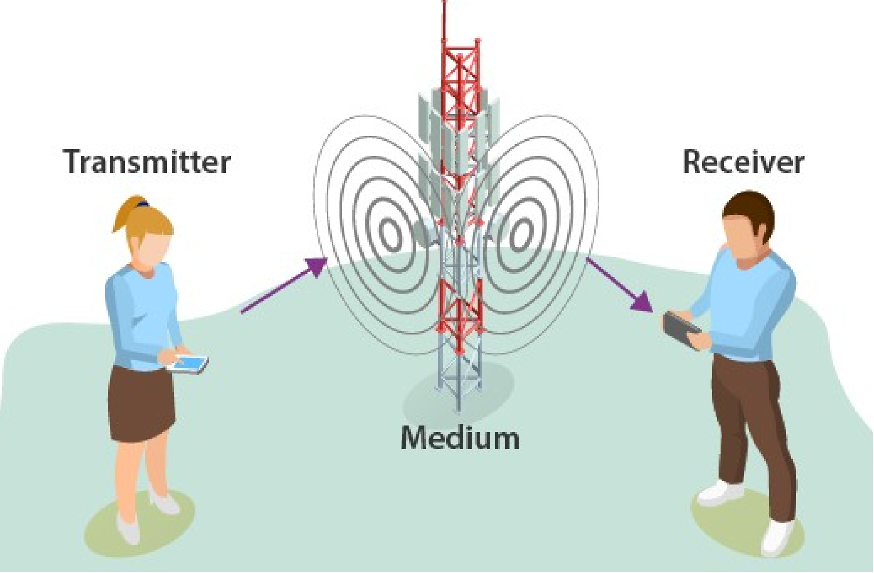
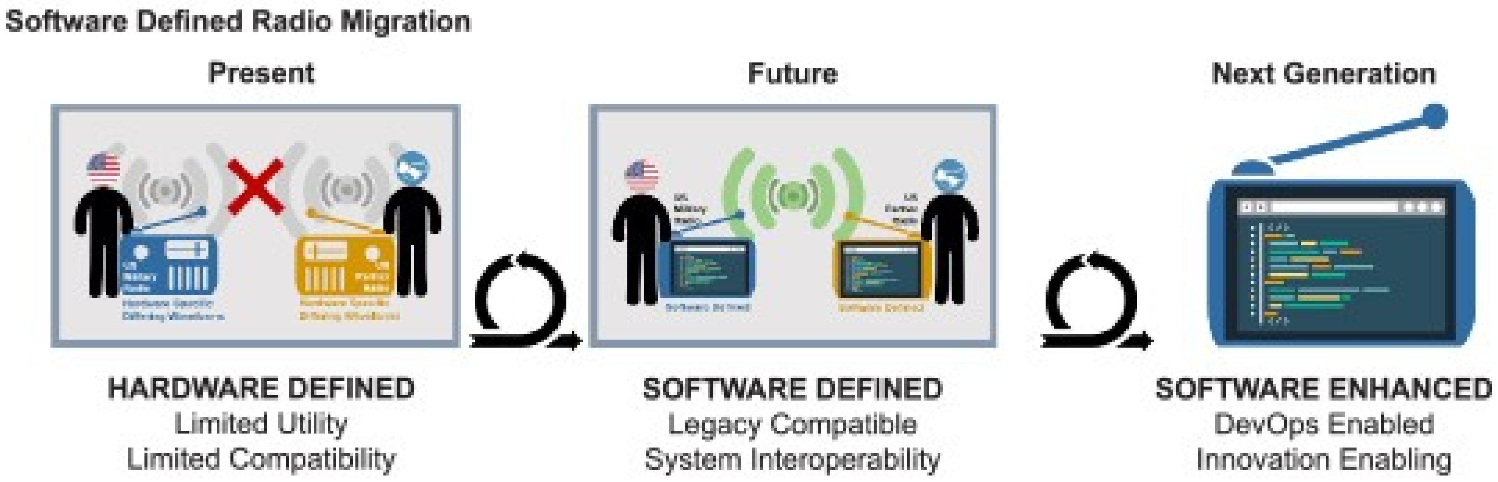




Introduction to Communication System

Sirisha Tadepalli

What is a Communication system?

- The transmission of information is called communication
- Information can be exchanged by one-to-one or one-to-many or many-many (Fig. 2)
- It can be wired or wireless (Electronics view)

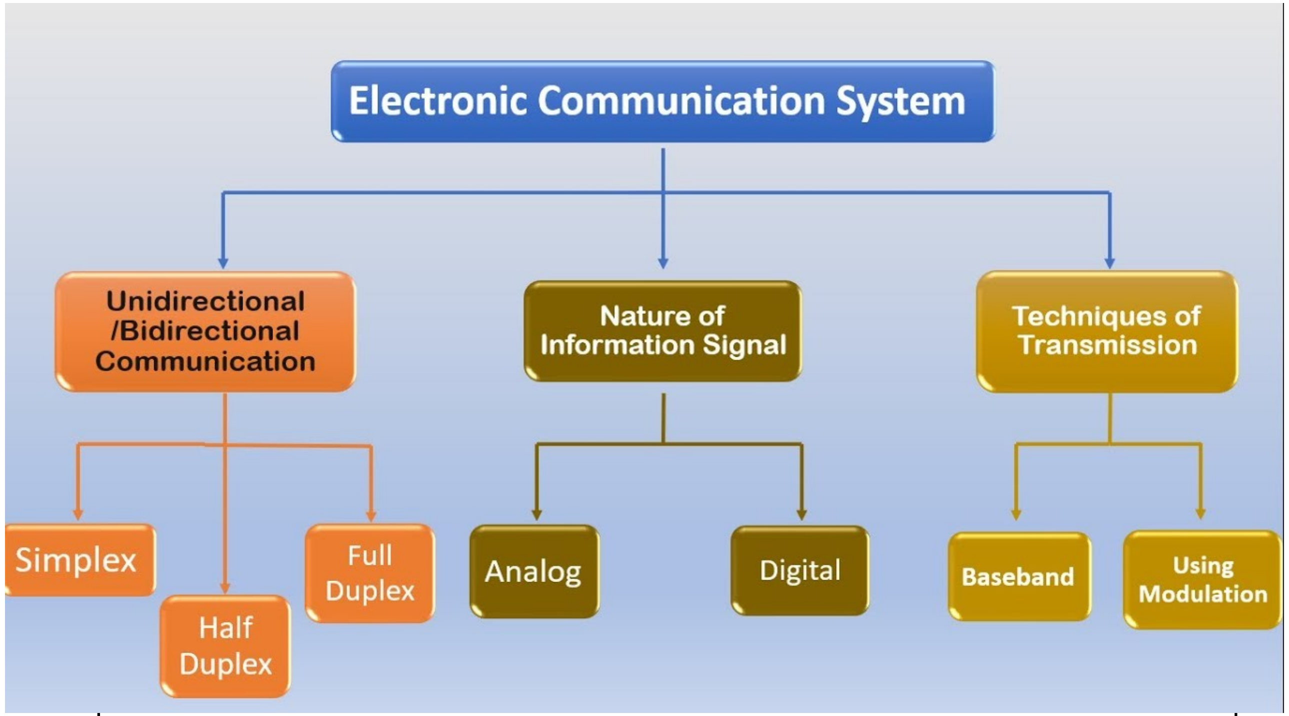
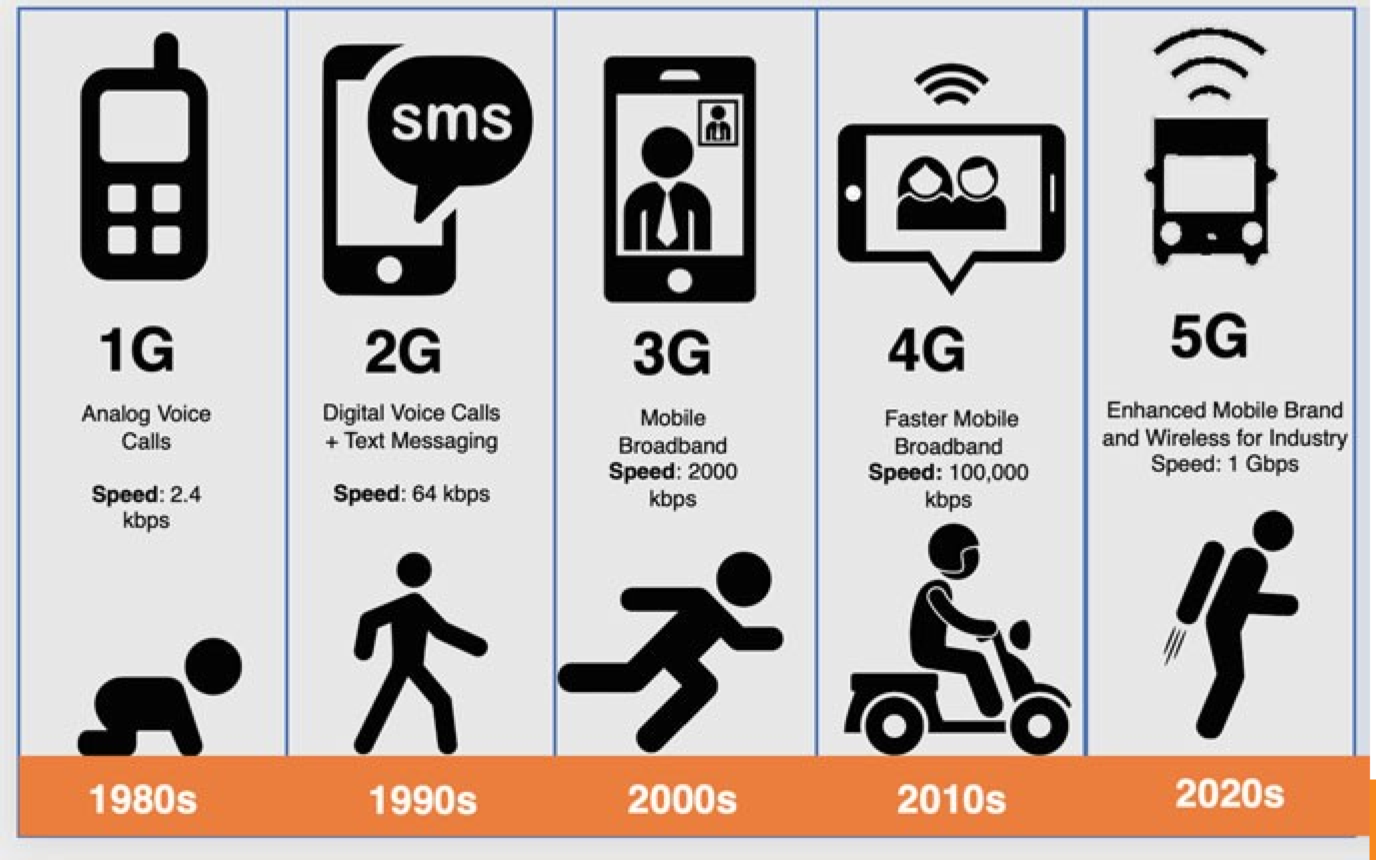


Fig.1 Basic classification of an electronic communication system [4]

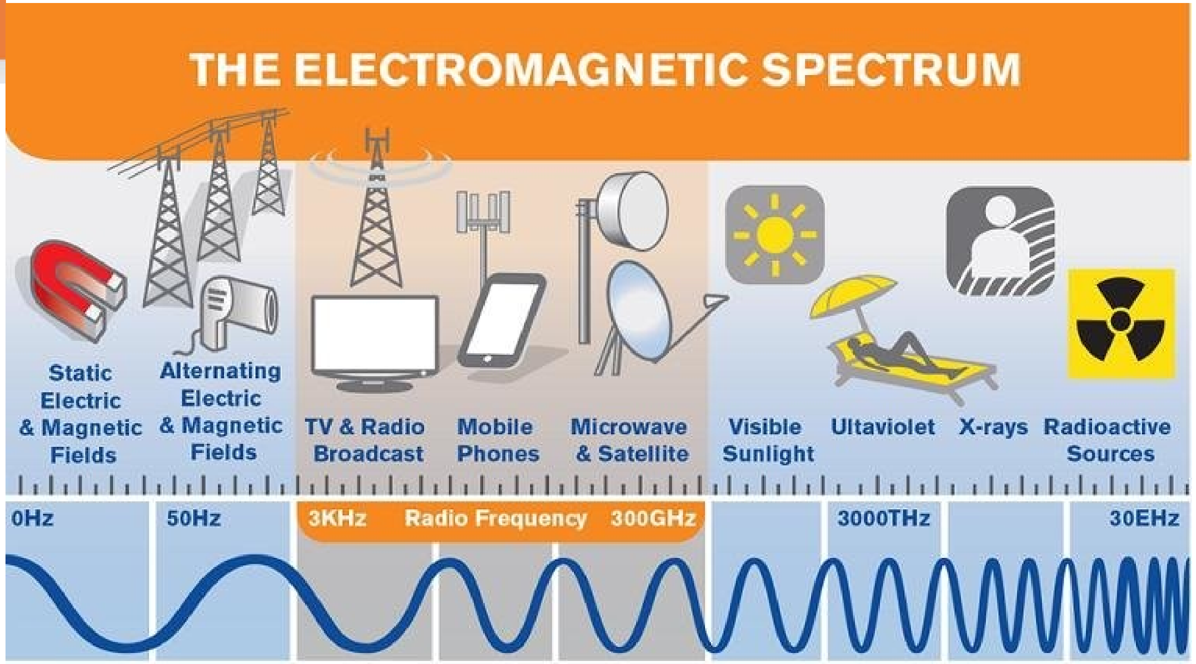
Fig. 2 Basic Information can be transmitted: one-to-many
Ex: Radio Broadcasting [3]

Why communication engineering for AI?

- **Foundational understanding for applying AI models**
  - Ex: Human-machine interaction, Machine-to-machine interaction, Computer Vision, and Natural language processing (NLP)
- **Hardware Interaction:** Many AI systems are designed to interact with the physical world through sensors and actuators.
  - Understanding communication engineering principles helps you grasp how these devices convert data into electrical signals and vice versa. Ex: IoT
- **Efficient Data Transmission:** Communication engineering knowledge equips you with techniques for optimizing data transmission and minimizing errors. Ex: Autonomous vehicles



Evolution of mobile technology



Syllabus

Unit 1

Introduction to communication systems, introduction to signals, different types of signals and their characteristics, concept of system, linear time-invariant (LTI) system, sinusoids- concept of frequency, in-phase and quadrature component, bandwidth, pass band and stop band, Introduction to SDR platforms and devices- MATLAB Simulink and GNU radio Companion (GRC), RTL-SDR and Adalm Pluto. Signal analysis/ spectrum analysis and visualization using SDR tools.

Unit 2

Need for modulation, analog modulation schemes, amplitude modulation (AM) and its types - AM-DSB-SC, AMDSB-TC, SSB. AM Demodulation schemes, angle modulation- frequency modulation (FM) -Narrowband and wideband, phase modulation, FM demodulation, implementation of analog modulation/demodulation schemes using SDR tools.

Unit 3

Quadrature amplitude modulation and demodulation, pulse analog modulation schemes, digital carrier modulation/demodulation Schemes- amplitude shift keying (ASK), frequency shift keying (FSK), phase shift keying (PSK), M-ary signaling, BPSK, QPSK, implementation of digital modulation/demodulation schemes using SDR tools. Multicarrier modulation- OFDM, MIMO, Prospects of AI in communication system- radio signal or modulation classification.

Reference textbooks

B1. Wyglinski, Alexander M., Robin Getz, Travis Collins, and Di Pu. Software-defined radio for engineers. Artech House, 2018.

B2. Qasim Chaudhari, Wireless Communications from the Ground Up: An SDR Perspective, 2018

B3. Andrew Barron, Software Defined Radio: for Amateur Radio Operators and Shortwave Listeners, 2019

B4. C.R. Johnson and W.A. Sethares, Software Receiver Design: Build Your Own Digital Communication System in Five Easy Steps, Cambridge University Press, 2011

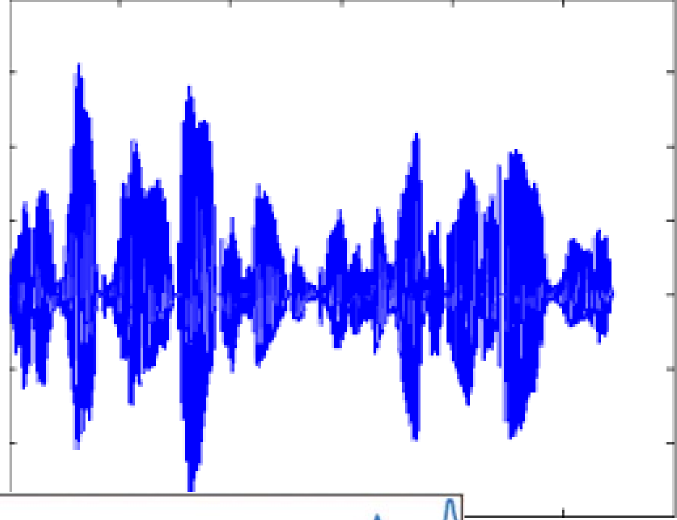
B5. Proakis, John G., Masoud Salehi, and Gerhard Bauch. Contemporary communication systems using MATLAB. Cengage Learning, 2012.

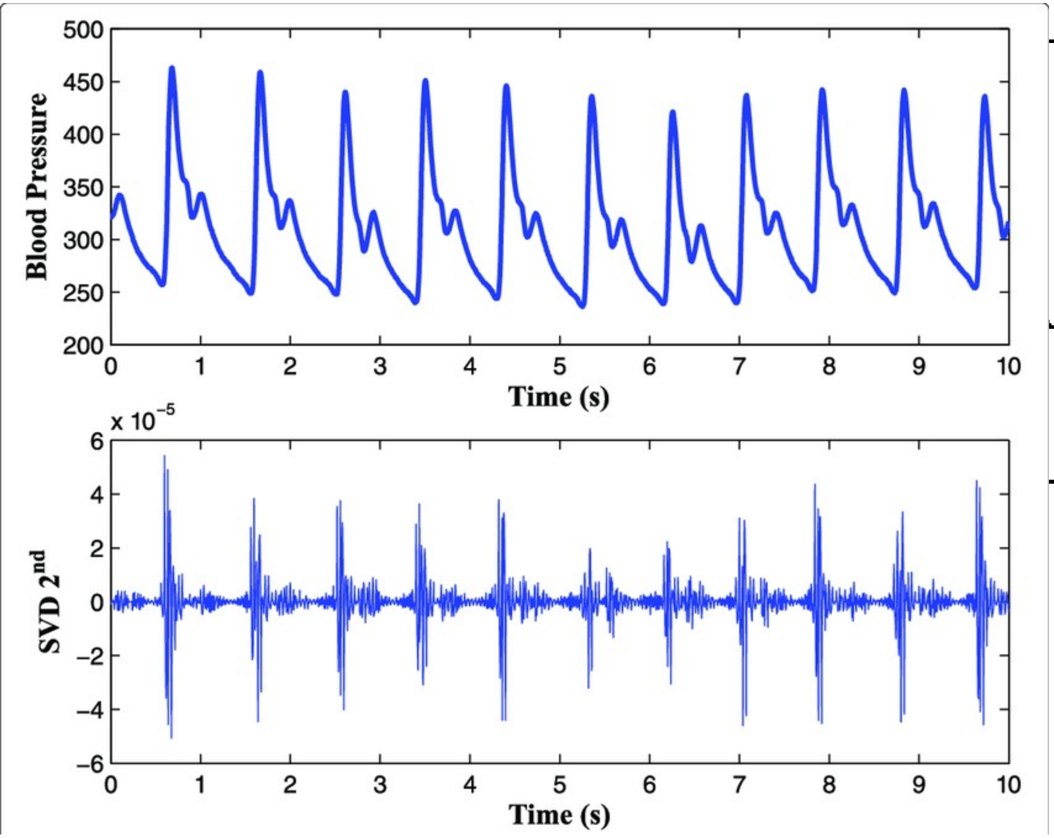
B6. Wyglinski, Alexander M., and Di Pu. Digital communication systems engineering with software defined radio. Artech House, 2013.

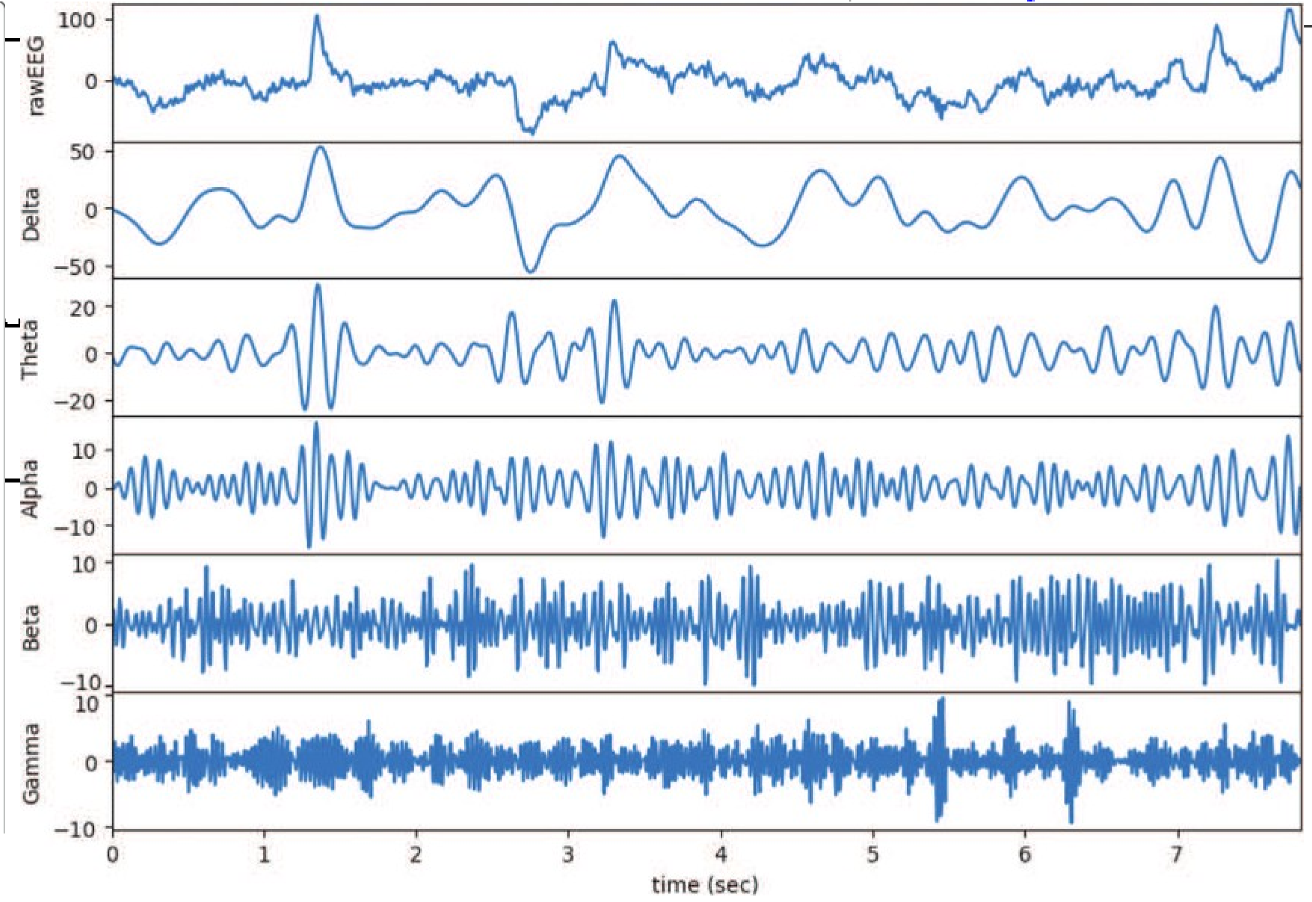
Introduction to signals

What is a signal?

- A fundamental quantity of representing information
- Information can be of any form
- Ex: Analog, digital, Sound, temperature, intensity, Pressure, etc..,





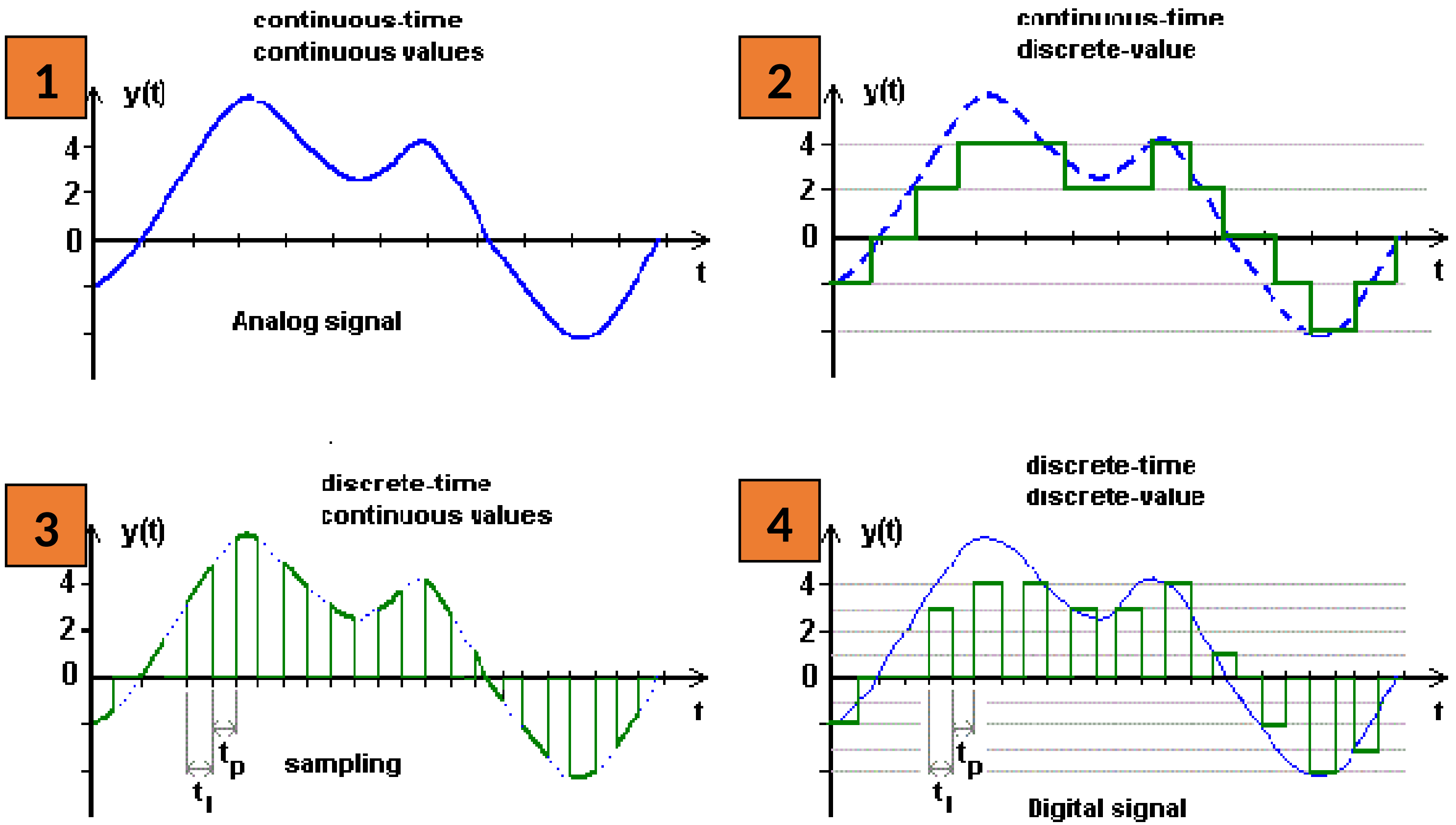


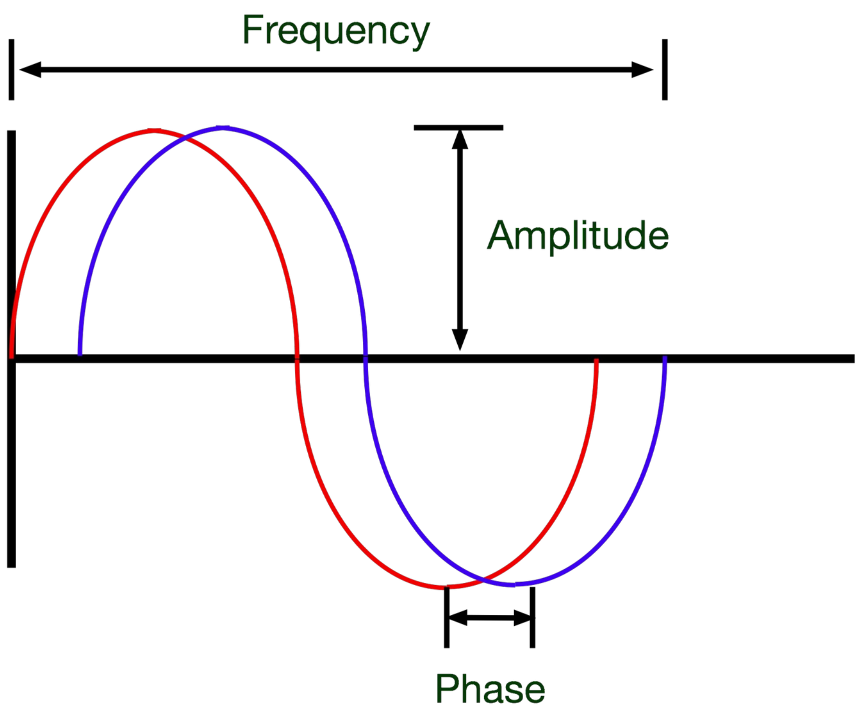
Types of Signals: or classification

1. Analog signal: continuous t and continuous y(t)
2. Continuous-time signal: continuous t and discrete y(t)
3. Discrete signal: discrete t and continuous y(t)
4. Digital signal: discrete t and discrete t

Characteristics of a signal

- Amplitude (height)
- Frequency (width)
- Phase (angle)





Classification of systems

- A **system** is an entity that processes one or more input signals to produce one or more output signals
- Classification:
  - No. of inputs: SISO, MIMO
  - Type of signal: Continuous-time system, discrete-time, hybrid system, and digital or analog systems
  - One-dimensional and multi-dimensional
- Example: Speech recognition system
- Amplification and noise reduction
- ECG abnormalities detection system

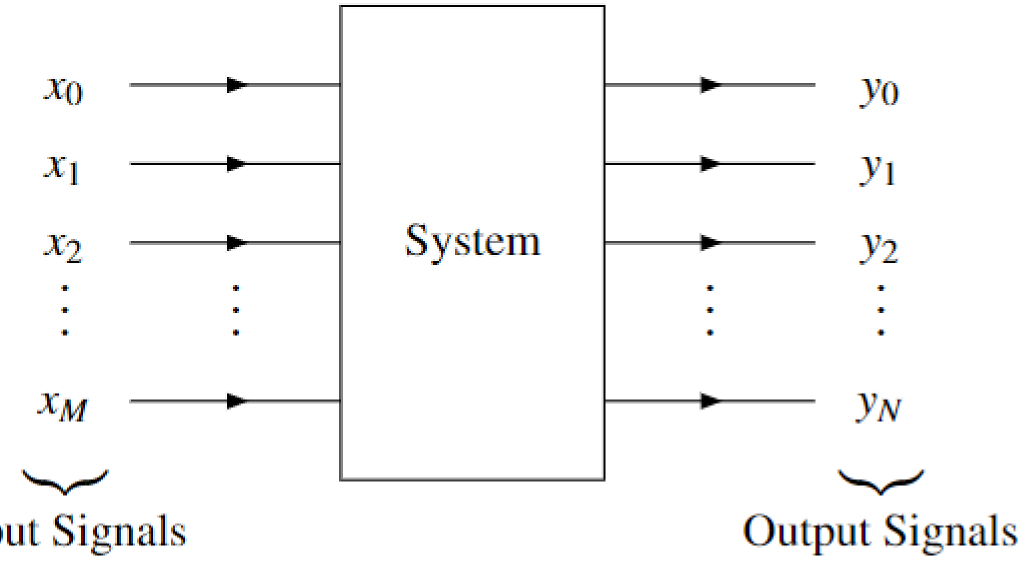
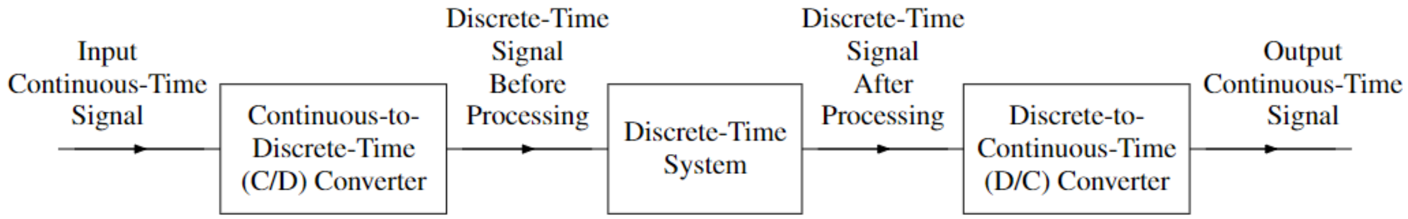


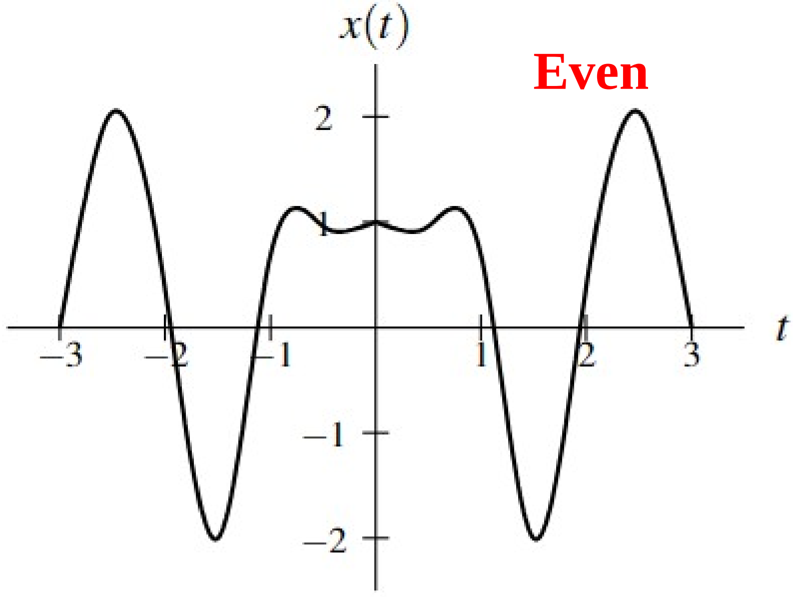
Figure 1.4: System with one or more inputs and one or more outputs.

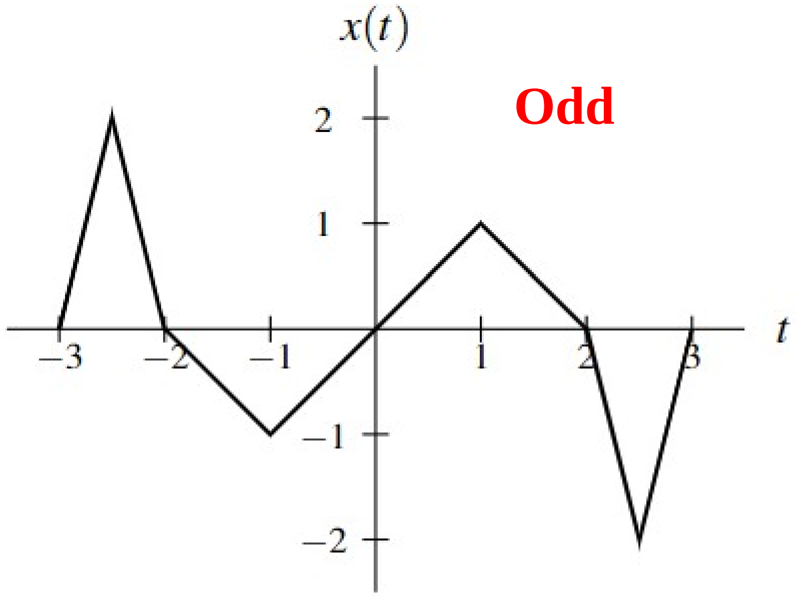


(a)

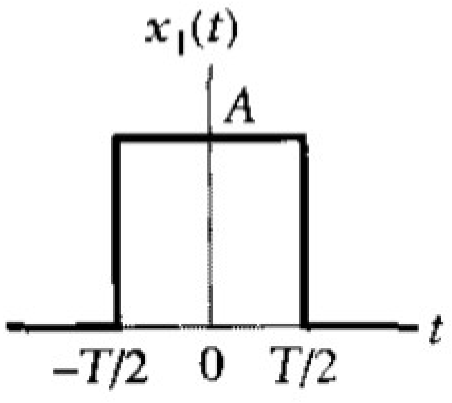
Symmetry of Functions and Sequences

- **Even** and **Odd signals**
- A function x is said to be even if it satisfies
- $x(t) = x(-t)$ or $x(n) = x(-n)$
- Geometrically, an even function or sequence is symmetric with respect to the vertical axis
- A function x is said to be **odd** if it satisfies
- $x(t) = -x(-t)$ for all t (where t is a real number)
- $x(n) = -x(-n)$ for all n (where n is an integer)
- One can easily show that an odd function or sequence x must be such that **$x(0) = 0$**, assuming that the domain of x includes 0

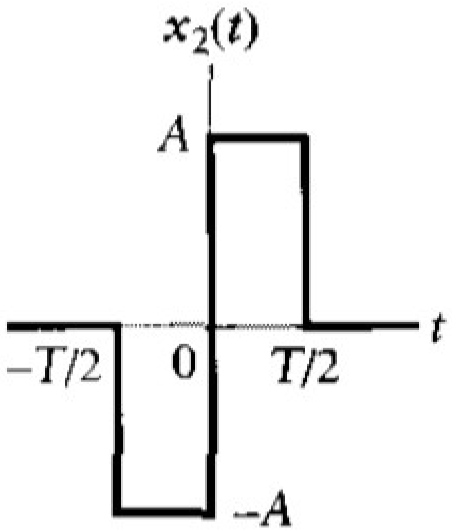




Examples



(a)



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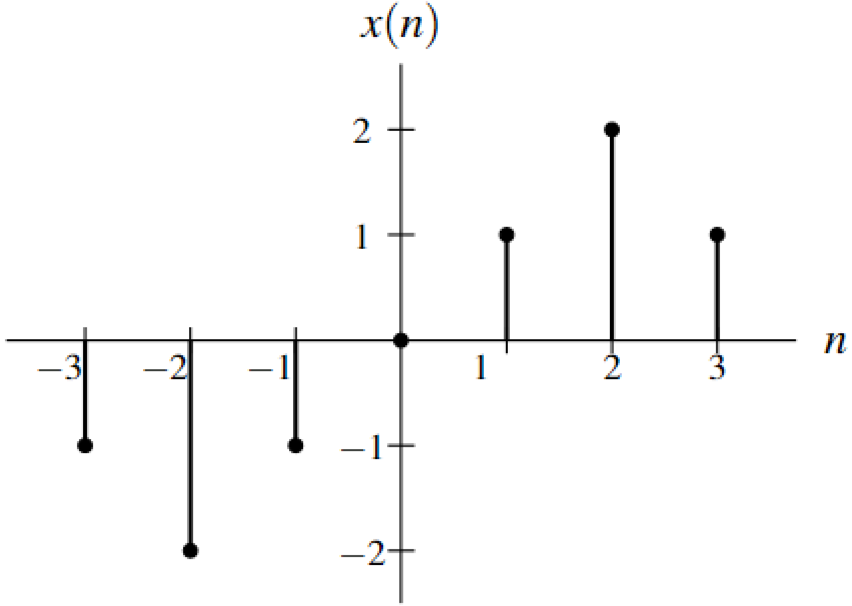
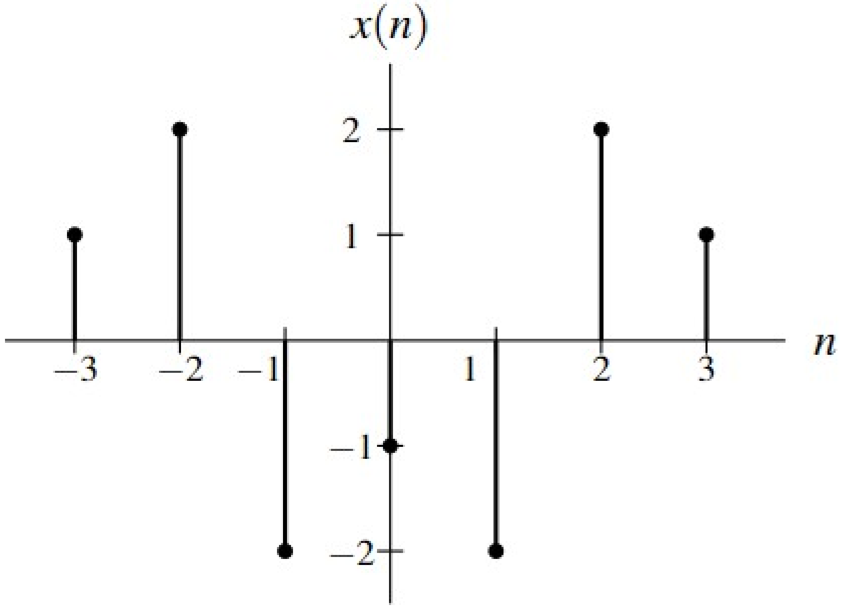
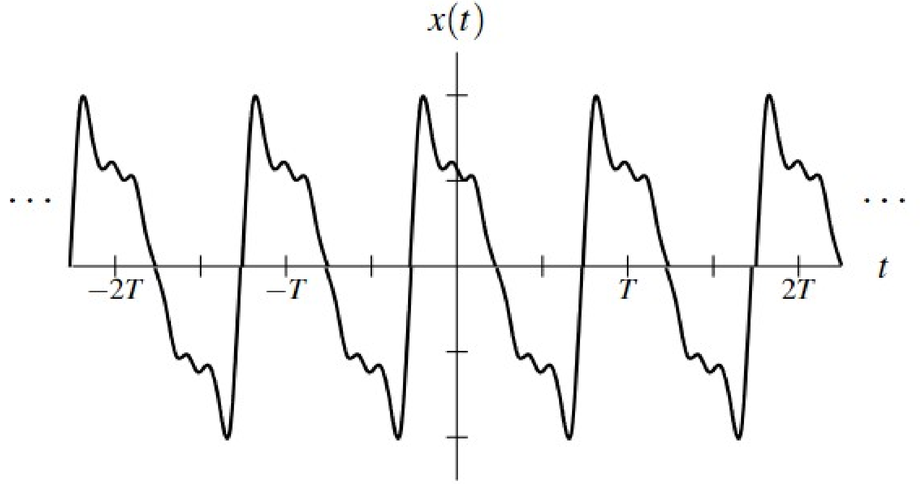
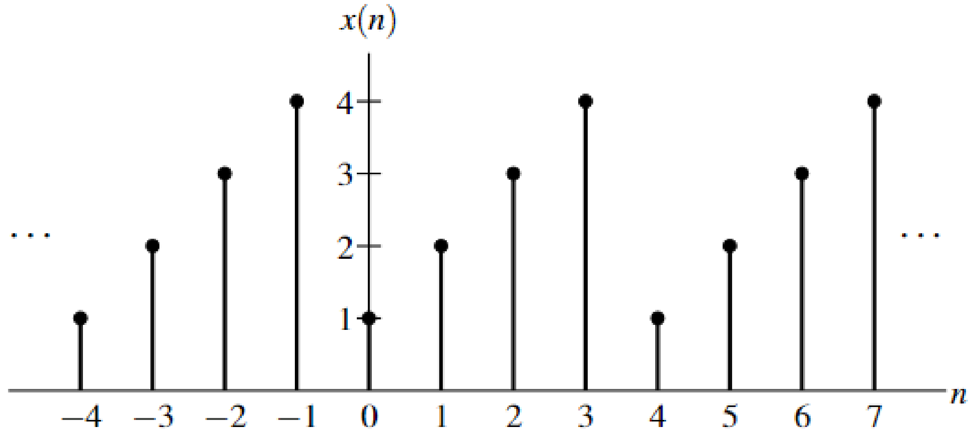




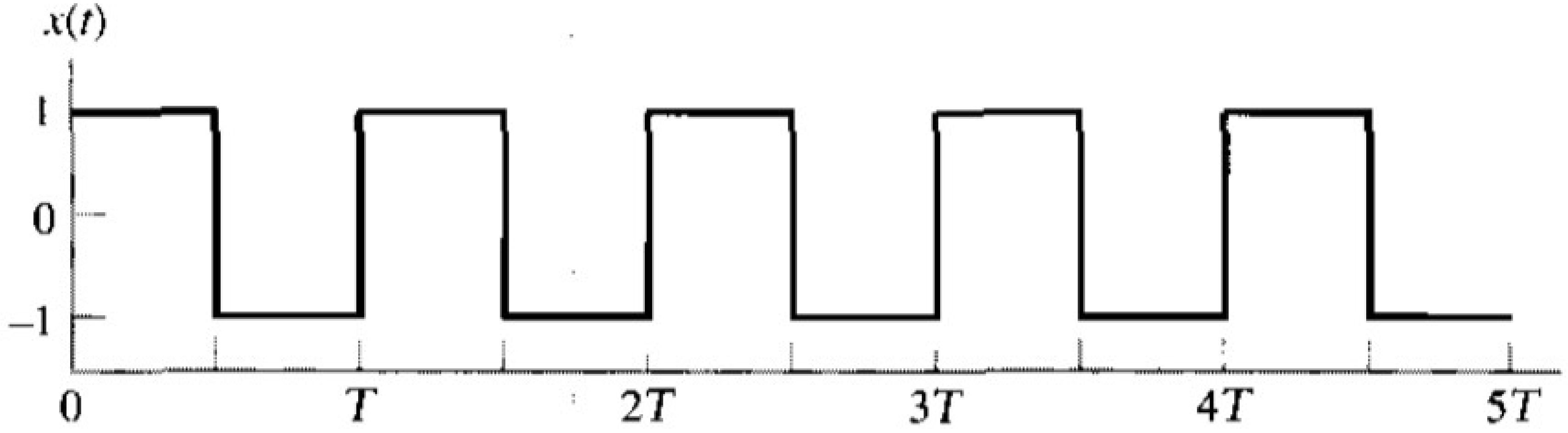
Periodicity of Functions and Sequences

- A function **x** is said to be periodic with period T (or simply T -periodic) if, for some strictly positive real constant T
- **x(t) = x(t + T) for all t (where t is a real number)**
- A T-periodic function x is said to have the frequency 1/T and angular frequency 2π/T
- A sequence x is said to be periodic with period N (or simply N-periodic) if, for some strictly positive integer N,
- **x(n) = x(n + N) for all n (where n is an integer)**
- An N-periodic sequence x is said to have a frequency 1/N and angular frequency 2π/N
- A function or sequence that is not periodic is said to be **Aperiodic**.





What is the frequency of the signal? If T =0.2s



(a)

Signal Energy and Power

The energy E contained in the function x is defined as **E =**

The average power P contained in the function x is given by $$P = \lim_{T \to \infty} \frac{1}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} |x(t)|^2 \, dt$$

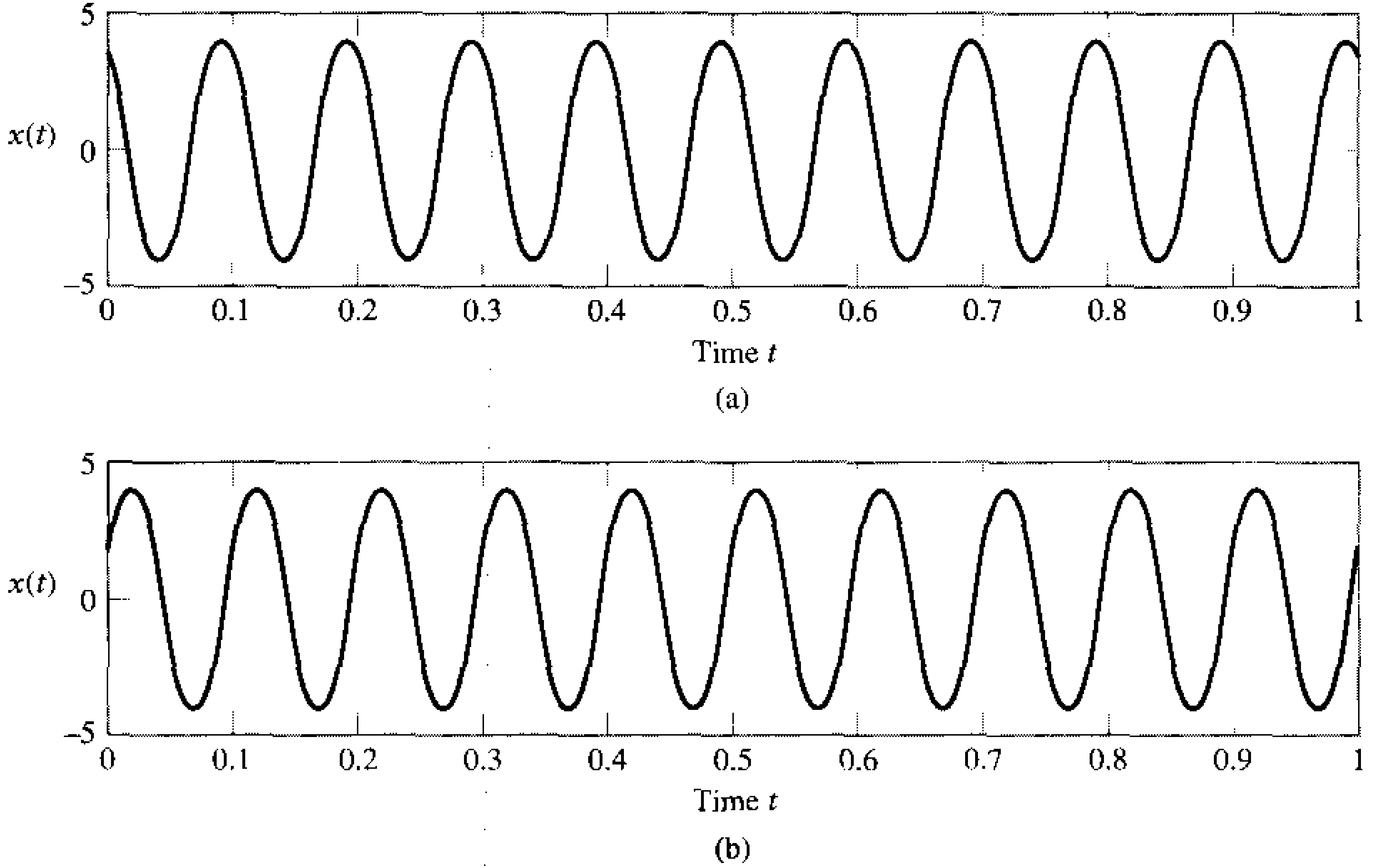
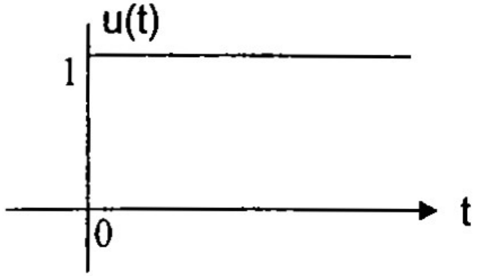


FIGURE 1.29 (a) Sinusoidal signal $A\cos(\omega t + \phi)$ with phase $\phi = +\pi/6$ radians. (b) Sinusoidal signal $A\sin(\omega t + \phi)$ with phase $\phi = +\pi/6$ radians.

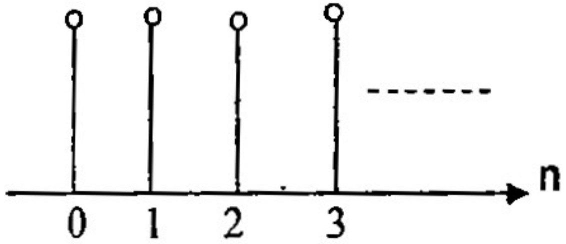
Unit step Function

- The unit-step function (also known as the Heaviside step function), denoted u(t), is defined as

$$u(t) = \begin{cases} 1; \; t > 0 \\ 0: \; t < 0 \end{cases}$$



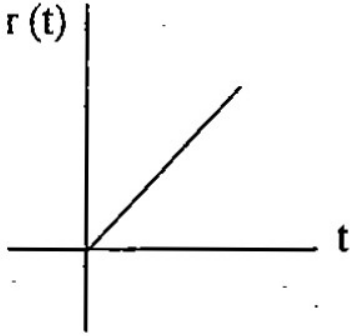
$$u(n) = \begin{cases} 1; \; n \geq 0 \\ 0; \; n < 0 \end{cases}$$



Ramp function

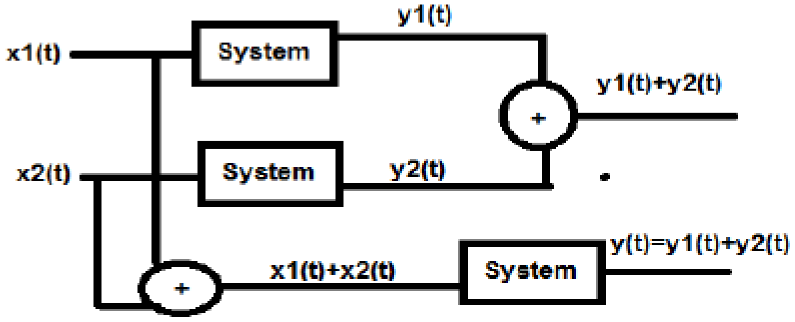
- Ramp function, r(t) is defined as

$$r(t) = \begin{cases} t: \; t > 0 \\ 0: \; t = 0 \end{cases}$$

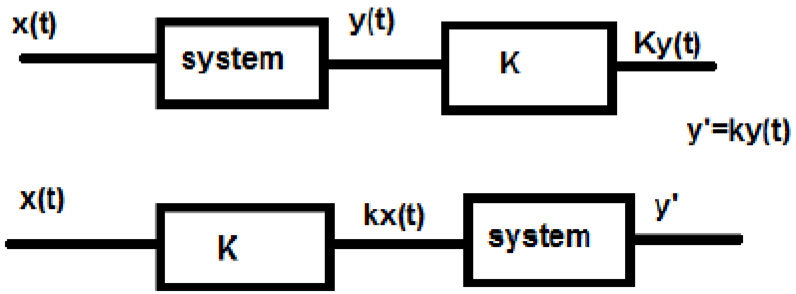


Linear time-invariant system (LTI)

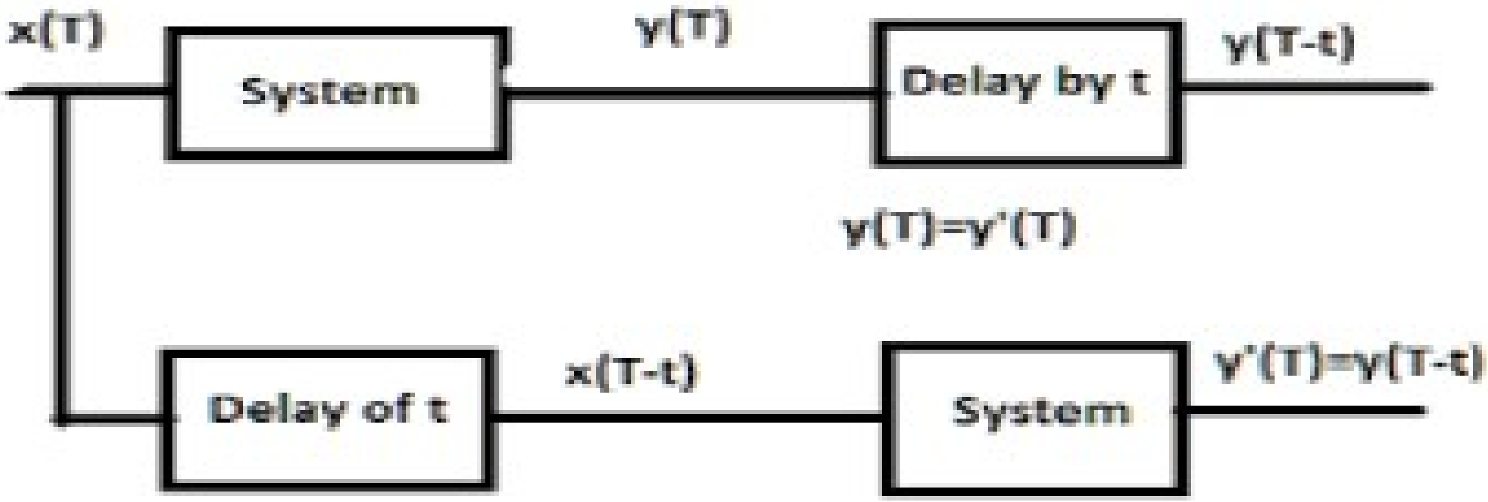
- Linear system and Time invariant system → LTI system
- **Linear system** follows superposition: Law of additivity and Law of homogeneity
- **Time invariant system**



Law of adiitivity



Law of homogeneity



Time invariant

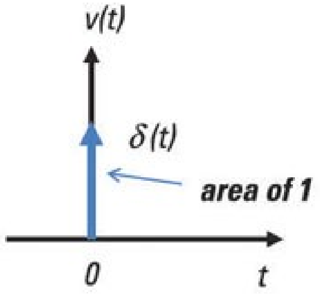
Linear time-invariant system (LTI)

- Why the LTI system?
- The effect of a system on the spectrum of a signal can be analyzed easily if and only if the system is LTI
- **Impulse response**

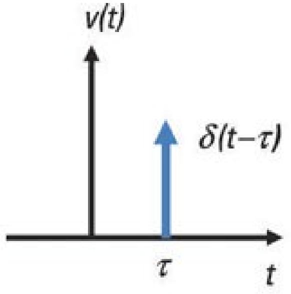
$x(t) = \delta\,(t)$ $y(t) = h(t)$

unit impulse → System → Impulse response

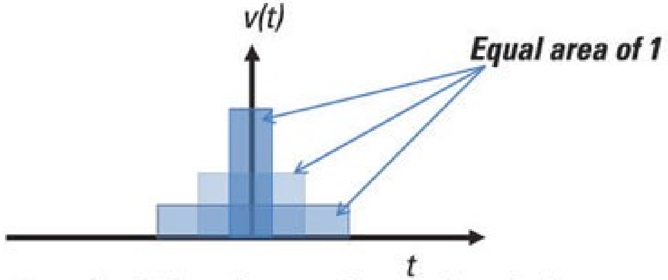
- The impulse response completely characterizes the behavior of an LTI system
- $\delta[n-i] \rightarrow$ LTI $\rightarrow h[n-i]$ (Time-invariance–works for any constant i)



Ideal Impulse at t = 0

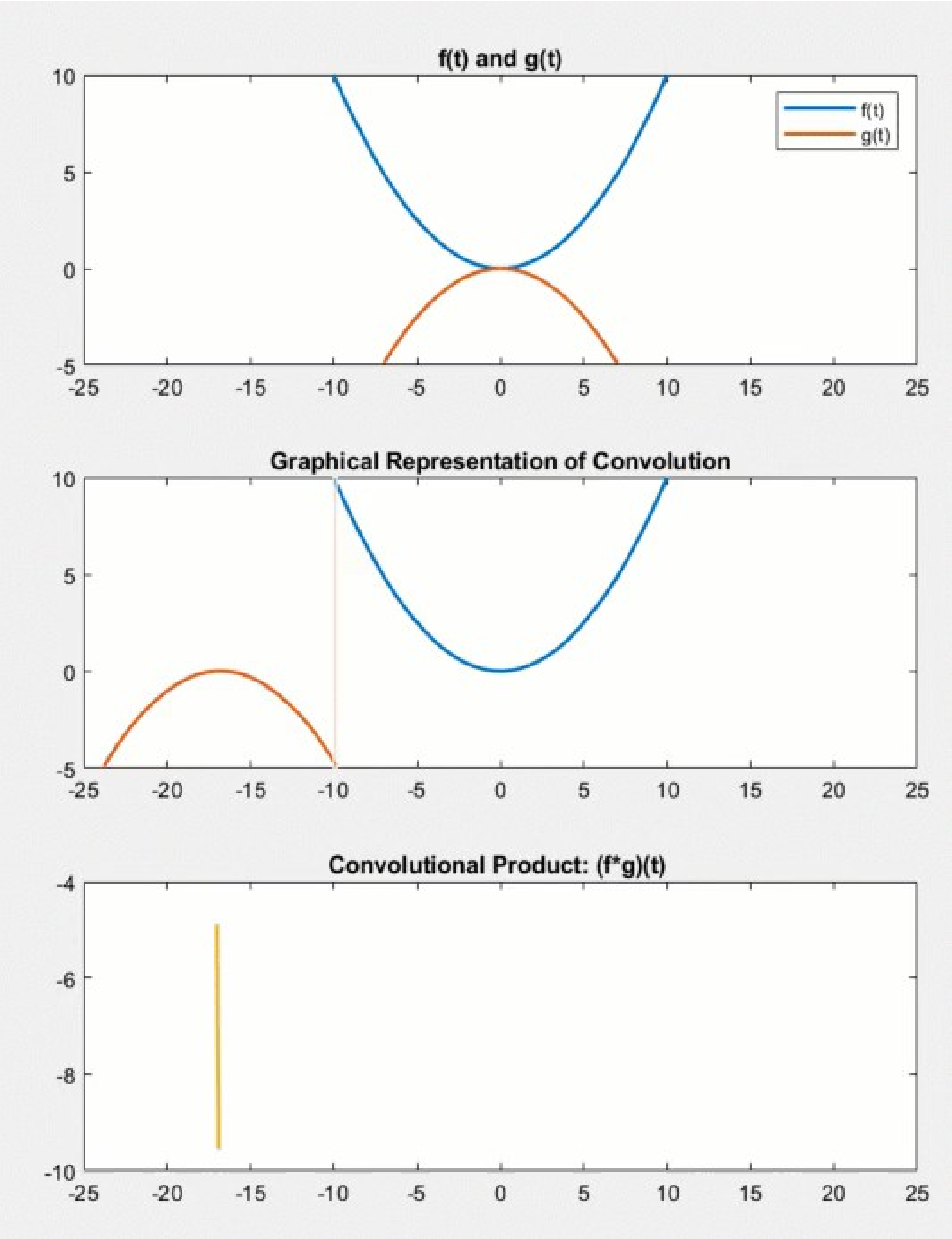


Delayed Ideal Impulse at τ = 0



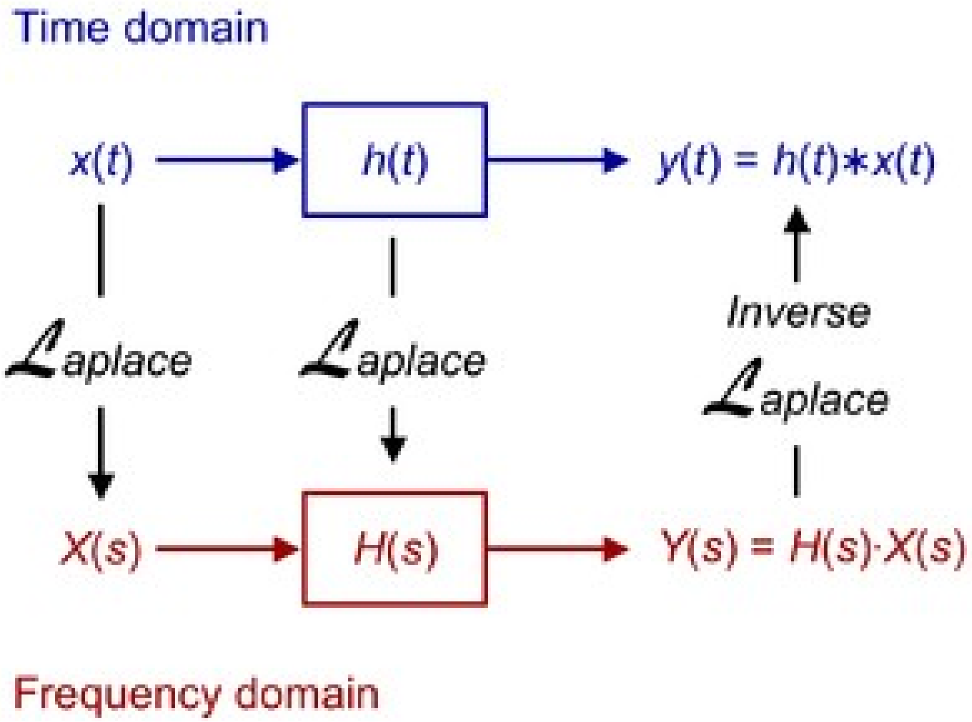
Rectangular Pulses (approach as an impulse)

Linear time-invariant system (LTI)



convolution is an integral that expresses the amount of overlap of one function $f(t)$, as it is shifted over function $g(t)$, for a continuous-time signal it is expressed as:

$$(f * g)(t) \approx^{def} \int_{-\infty}^{\infty} f(\tau)g(t-\tau)dr$$



Linear time-invariant system (LTI)

For a Discrete signal

x[n] → LTI → y[n] ¿ $\sum_{i=-\infty}^{\infty} x(i)h(n-i) = x[n] * h[n]$

If the length of vector x is N and the length of vector h is M

- The response vector Y of length M+ N -1:
- In other words, length(y) = length(x) + length(h)- 1:

- If the nonzero values of x[n] are in the interval [ax, bx] and the nonzero values of h[n] are in the interval [ah, bh] then the nonzero values of the output y[n] are in the interval [ax + ah, bx + bh]:

Qn. Consider an LTI system with impulse response h[n]=[2 4 1 3], $0 \le n \le 3$
Find the response of the system to input signal $x[n] = \delta[n]$

Sol: $y(n) = x(n) * h(n) = \sum_{k=-\infty}^{\infty} x(k)h(n-k)$

Since both $x(k) = 0 \; for \; k < 0 \quad and \quad h(n-k) = 0 \; for \; k > n$

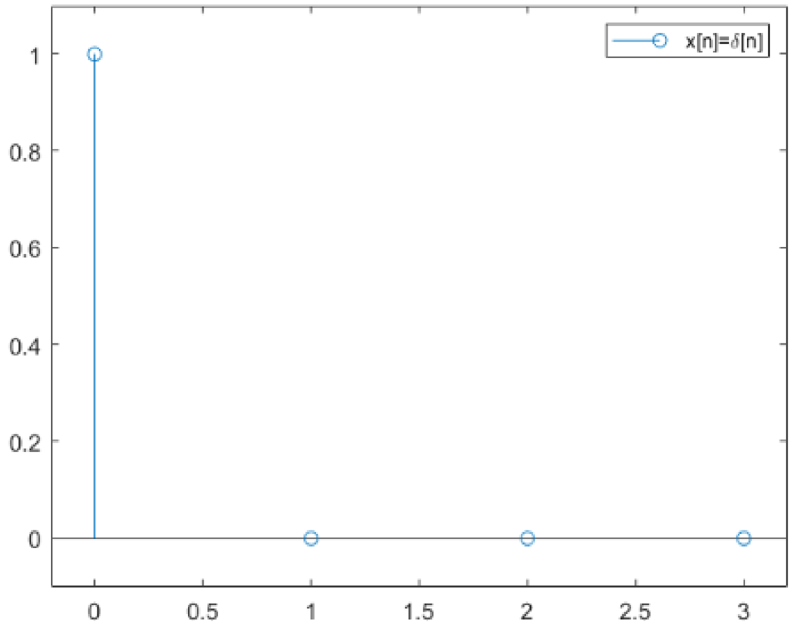
$$y(n) = \sum_{k=0}^{n} x(k)h(n-k)$$

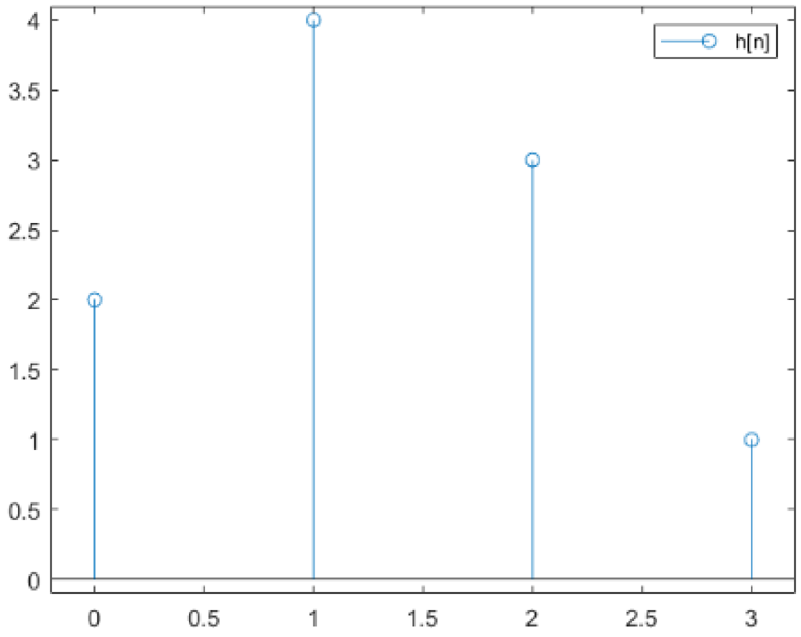
$$y(0) = \sum_{k=0}^{0} x(k)h(n-k) = x(0)h(0) = 2 \qquad y(1) = \sum_{k=0}^{1} x(k)h(1-k) = x(0)h(1) + x(1)h(0) = 4 + 0 = 4$$

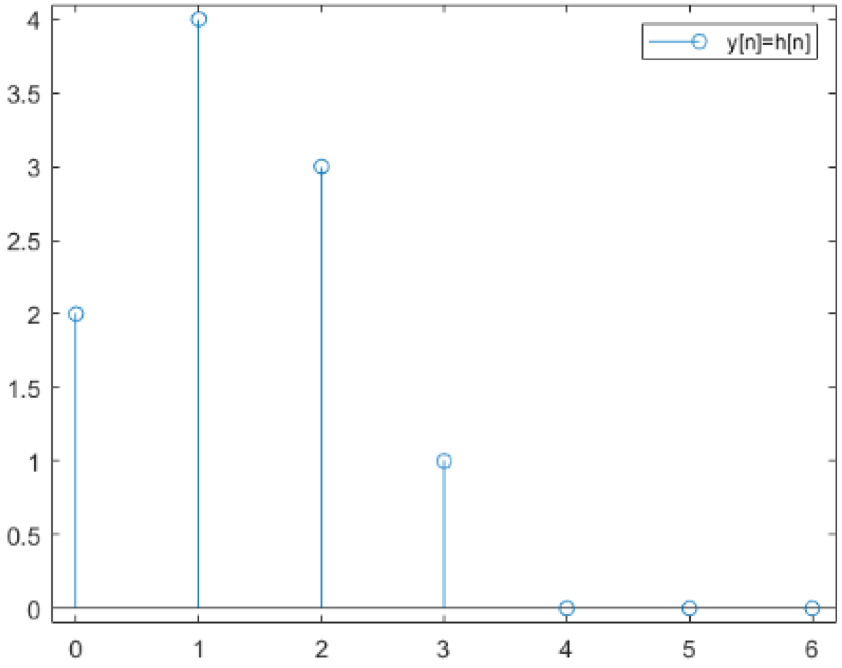
Similarly, we can obtain

$$y(2) = 1, y(3) = 3, y(4) = 0, y(5) = 0, y(6) = 0 \qquad y(n) = 0 \; for \; n \ge 7$$

Verification : Computationally





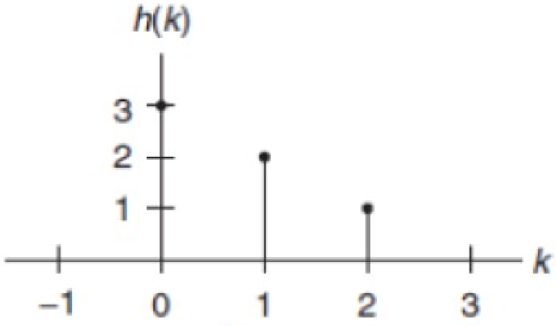


What will be the response y[n] for input ?

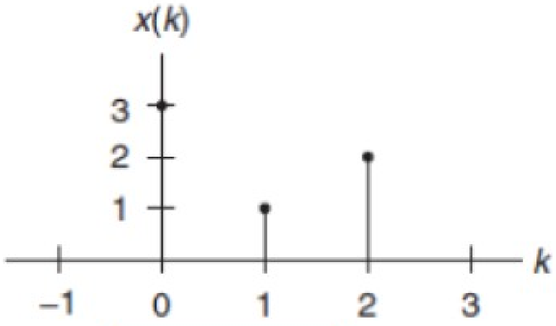
Convolution 2

Example:

$$x[n] = \begin{cases} 3 & n = 0 \\ 2 & n = 1 \\ 1 & n = 2 \\ 0 & elsewhere \end{cases}$$



Length =3



Length =3

$$h[n] = \begin{cases} 3 & n = 0 \\ 1 & n = 1 \\ 2 & n = 2 \\ 0 & elsewhere \end{cases}$$

Solution:

Convolution sum using the table method.

k:	-2	-1	0	1	2	3	4	5	
$x(k)$:			**3**	**1**	**2**				
$h(-k)$:	1	2	3						$y(0) = 3 \times 3 = 9$
$h(1-k)$		1	2	3					$y(1) = 3 \times 2 + 1 \times 3 = 9$
$h(2-k)$			1	2	3				$y(2) = 3 \times 1 + 1 \times 2 + 2 \times 3 = 11$
$h(3-k)$				1	2	3			$y(3) = 1 \times 1 + 2 \times 2 = 5$
$h(4-k)$					1	2	3		$y(4) = 2 \times 1 = 2$
$h(5-k)$						1	2	3	$y(5) = 0$ (no overlap)

Convolution Length = $N_1 + N_2 - 1 = 3 + 3 - 1 = 5$

Sinusoids

- Why do we need to study Sinusoids?

✓For eg. Modulation schemes

- Used in communication systems

✓Speech signal in low frequency region (80-400 Hz)

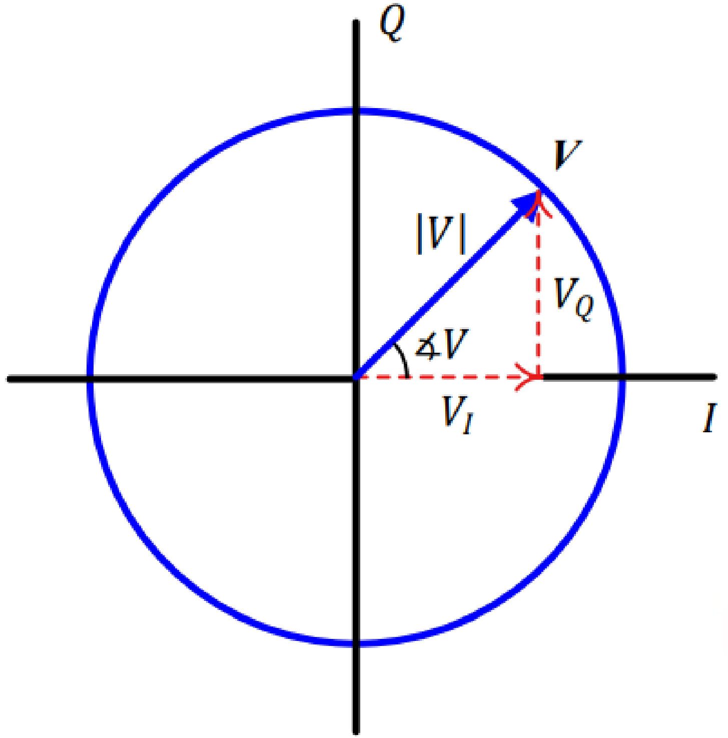
$$y_{\mathrm{LFR}}[n] = \sum_{i=1}^{M} A_i[n] \cos\left(\omega_i[n]n + \phi_i[n]\right)$$

- Any signal can be represented as the sum of sinusoids

- Many systems can be characterized by their response to sinusoids

✓Eg. Filters, Equalizers

Complex number



V is a complex number in this IQ-plane

$I \rightarrow Inphase$
$Q \rightarrow Quadrature$

$$V_I = |V| \cos \measuredangle V$$
$$V_Q = |V| \sin \measuredangle V$$

$|V|$ and $\measuredangle V$ are the magnitude and angle of V with respect to I-axis,

Magnitude and Phase

In polar representation of complex numbers, the magnitude of V in an IQ-plane

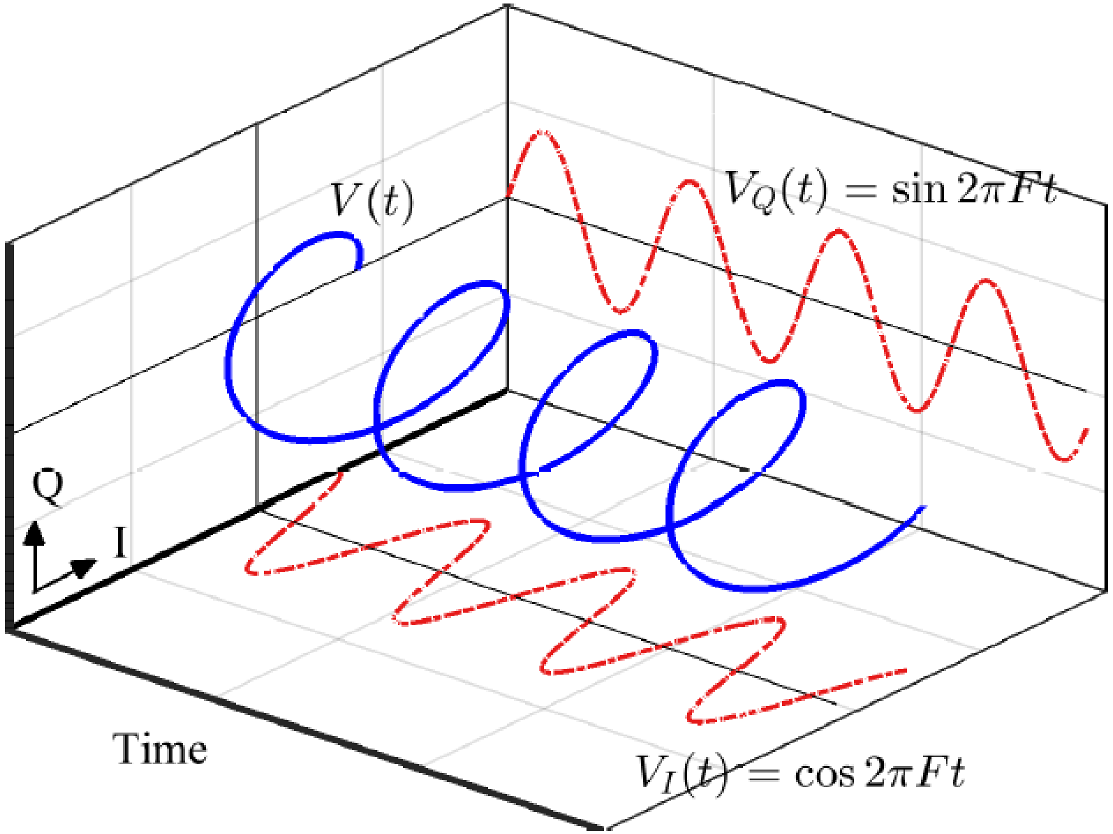
$$|V| = \sqrt{V_I^2 + V_Q^2}$$

the phase $\measuredangle V$

$$\tan^{-1} V_Q / V_I.$$

Complex Sinusoid

V rotating anticlockwise in a circle at a constant rate with time



Instead of a complex number V, now V(t) can be treated as a signal with time as independent variable and we call it a complex sinusoid.

Complex sinusoid is made up of 2 real sinusoids

Complex Sinusoid

Concept of Frequency

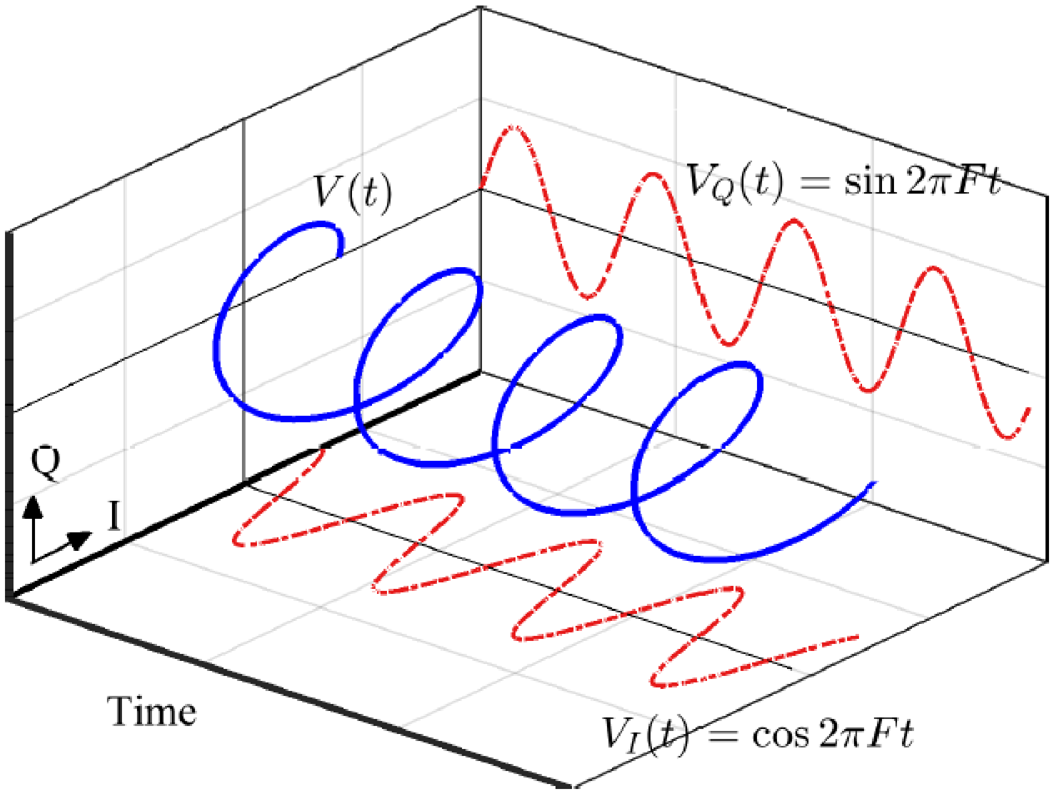
A constant rate implies that in any given duration Δt, the change in phase $\Delta\theta$ is a constant

$$\text{angular velocity} = \frac{\Delta\theta}{\Delta t}$$

The frequency of this complex sinusoid

$$F = \frac{1}{2\pi} \cdot \frac{\Delta\theta}{\Delta t}$$

V rotating anticlockwise in a circle at a constant rate with time



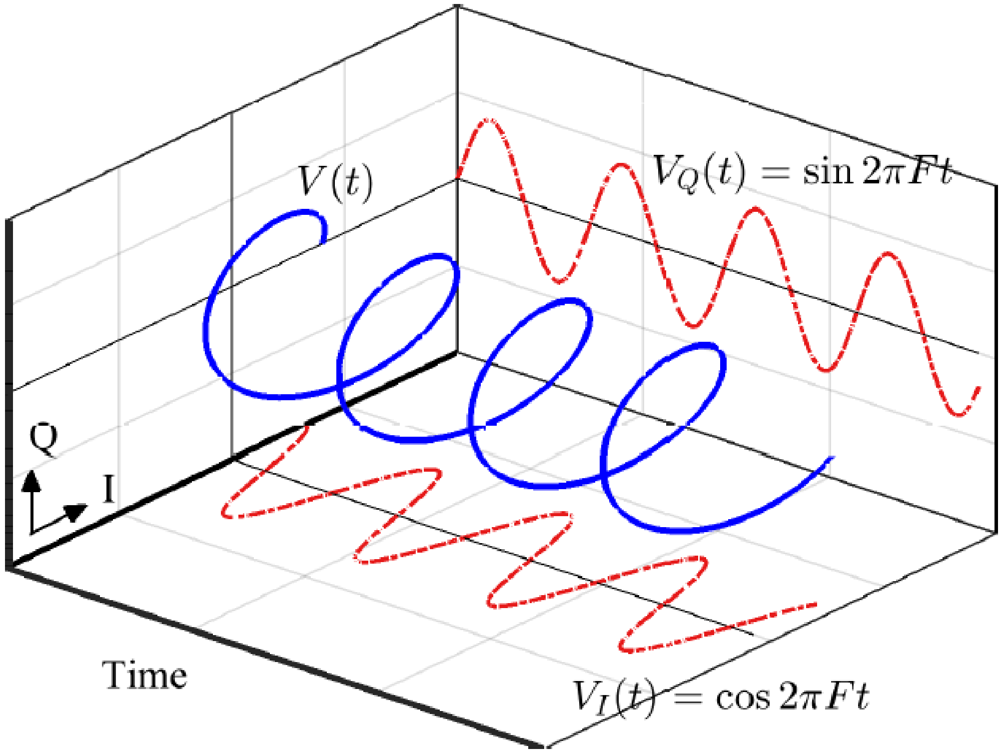
Inphase and Quadrature Component

Observe the projection of v(t) into 2D plane

Projection from a 3D to a 2-D plane formed by time and I-axis

↓

Inphase part



Projection from a 3D to a 2-D plane formed by time and Q-axis

↓

Quadrature Part

Complex sinusoid with frequency F is composed of two real sinusoids

$$V_I(t) = \cos 2\pi F t$$
$$V_Q(t) = \sin 2\pi F t$$

$$F = \frac{1}{T}$$

F is the continuous frequency with units of cycles/second or Hertz (Hz).

Range of continuous frequency values $\boxed{-\infty < F < \infty}$

When one tunes to an FM radio station at 88 MHz, one is actually listening to a station broadcasting a radio signal at a carrier frequency of 88×10^6 Hz, which means that the transmitter is oscillating at a frequency of $88,000,000$ cycles/second. Accordingly, that wave is completing one period in $T = 1/F = 11.4$ ns.

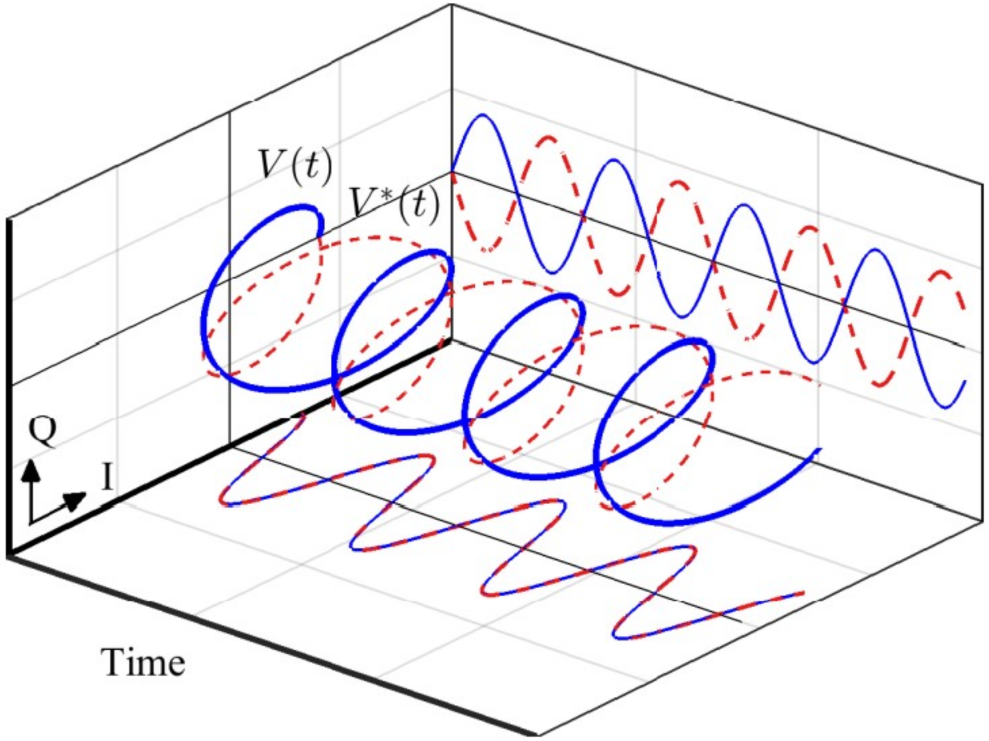
In summary

- Frequency is the rate of rotation of a complex sinusoid in time *IQ*-plane
- This rate of rotation can be changed from very slow (close to 0) to as fast as possible (close to $+\infty$)
- A clockwise direction of rotation implies a negative frequency, while anti-clockwise rotation implies a positive frequency

Anti-clockwise rotation (+ve F) = $I.\cos\theta - Q.\sin\theta$; $Q.\cos\theta + I.\sin\theta$
Clockwise rotation (-ve F) = $I.\cos\theta + Q.\sin\theta$; $Q.\cos\theta - I.\sin\theta$

Real Sinusoid

- How to produce only one real sinusoid in complex IQ-plane?
- Rotating two complex sinusoids in opposite directions to each other



Two complex sinusoids $V(t)$ and $V^*(t)$ rotating in time IQ-plane

- V(t) with a frequency F shown as solid blue line while V*(t) with frequency –F shown as dashed red line
- The I parts exactly fall on top of each other and hence are the same cos 2πFt
- The Q parts carry exactly the same amplitude with opposite signs to each other and hence are sin 2πFt and – sin 2πFt, respectively.

Observations

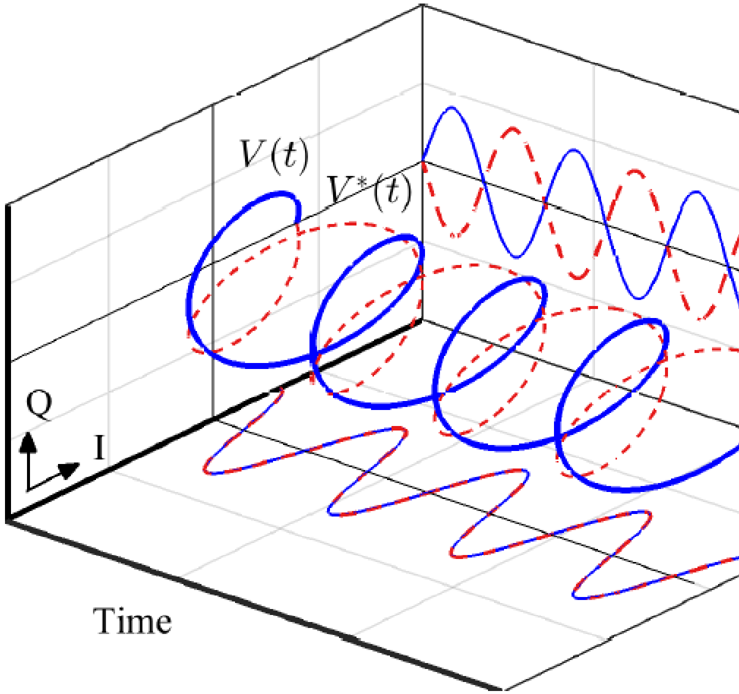
I parts : $\cos 2\pi F t$

Q parts : $\sin 2\pi F t$ and $-\sin 2\pi F t$

$$|V \cdot V^*| = |V|^2 = V_I^2 + V_Q^2$$
$$\measuredangle (V \cdot V^*) = 0$$

$I \quad \rightarrow \qquad \{V + V^*\} = V_I + V_I \qquad V_I = \frac{1}{2}\{V + V^*\}$

$Q \quad \uparrow \qquad \{V + V^*\} = V_Q - V_Q \qquad 0 = \frac{1}{2}\{V + V^*\}$



Addition of 2 complex sinusoids

$$I \quad \rightarrow \qquad \cos 2\pi F t = \frac{1}{2}\{V(t) + V^*(t)\}$$

$$Q \quad \uparrow \qquad 0 = \frac{1}{2}\{V(t) + V^*(t)\}$$

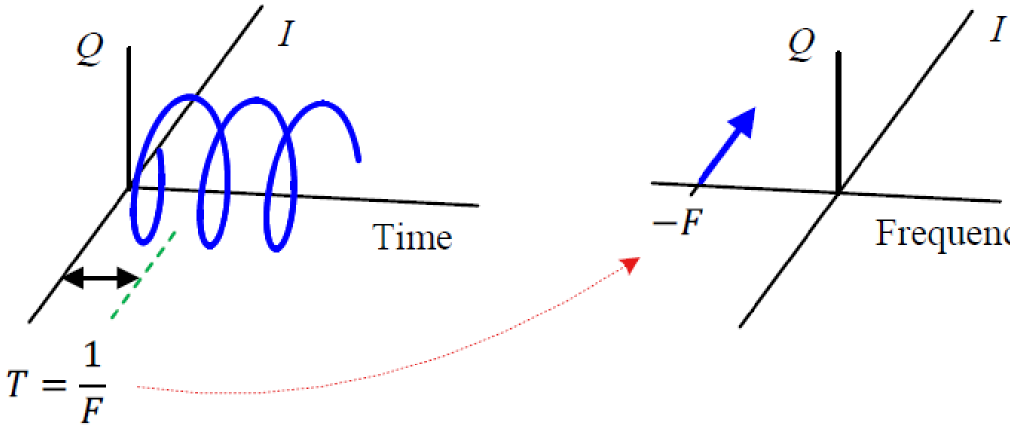
Subtraction of 2 complex sinusoids

$$I \quad \rightarrow \qquad 0 = \frac{1}{2}\{V(t) - V^*(t)\}$$

$$Q \quad \uparrow \qquad \sin 2\pi F t = \frac{1}{2}\{V(t) - V^*(t)\}$$

Frequency Domain representation

Complex sinusoid:



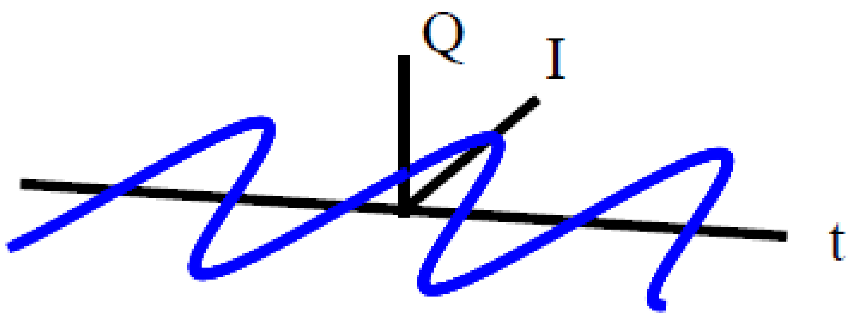
Representation of a complex sinusoid in frequency domain. A r
value of F implies clockwise rotation

Real cosine

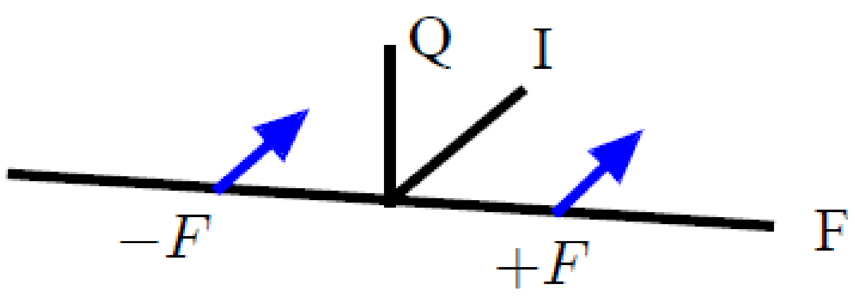
$$I \quad \rightarrow \quad \cos 2\pi F t = \frac{1}{2}\left\{V(t) + V^*(t)\right\}$$

$$Q \quad \uparrow \quad 0 = \frac{1}{2}\left\{V(t) + V^*(t)\right\}$$

Cosine in Time IQ-Plane



Cosine in Frequency IQ-Plane



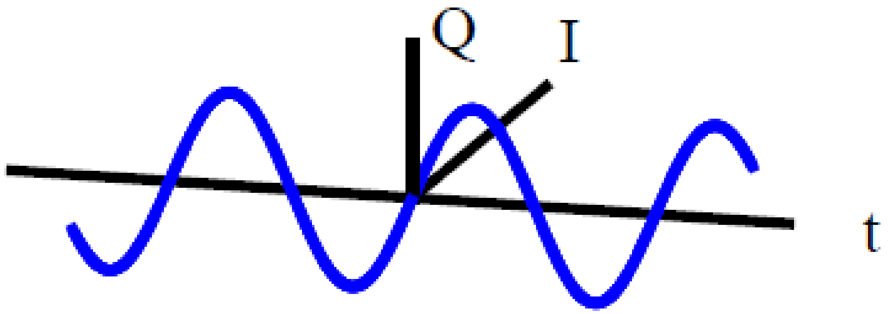
A frequency domain representation of a cosine wave should be two impulses, one at frequency +F due to V(t) and the other at frequency −F contributed by V * (t).

Real sine

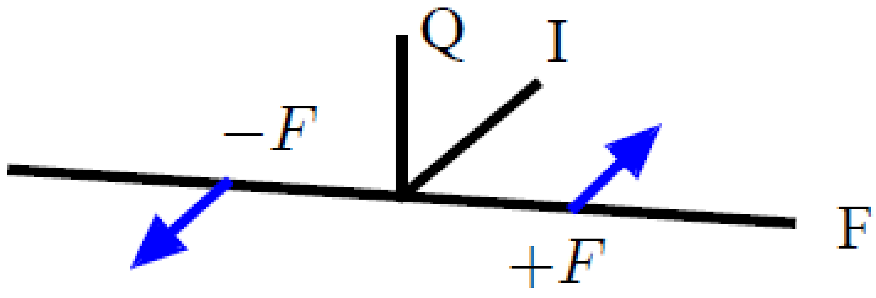
$$I \quad \rightarrow \qquad 0 = \frac{1}{2}\{V(t) - V^*(t)\}$$

$$Q \quad \uparrow \quad \sin 2\pi F t = \frac{1}{2}\{V(t) - V^*(t)\}$$

Sine on Q-axis in Time IQ-Plane

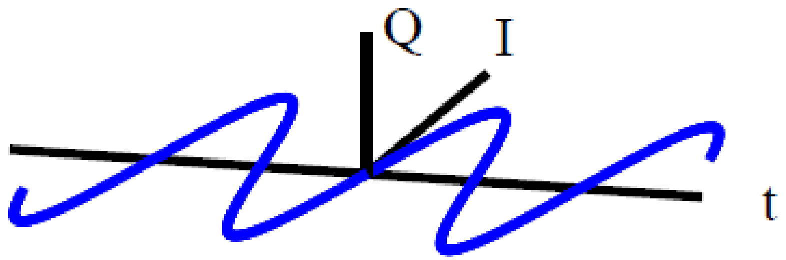


Sine in Frequency IQ-Plane

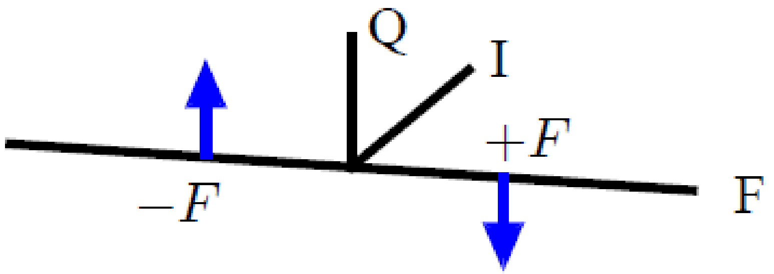


what is the frequency domain representation of a sine wave on time and I-axis?

Sine on I-axis in Time IQ-Plane



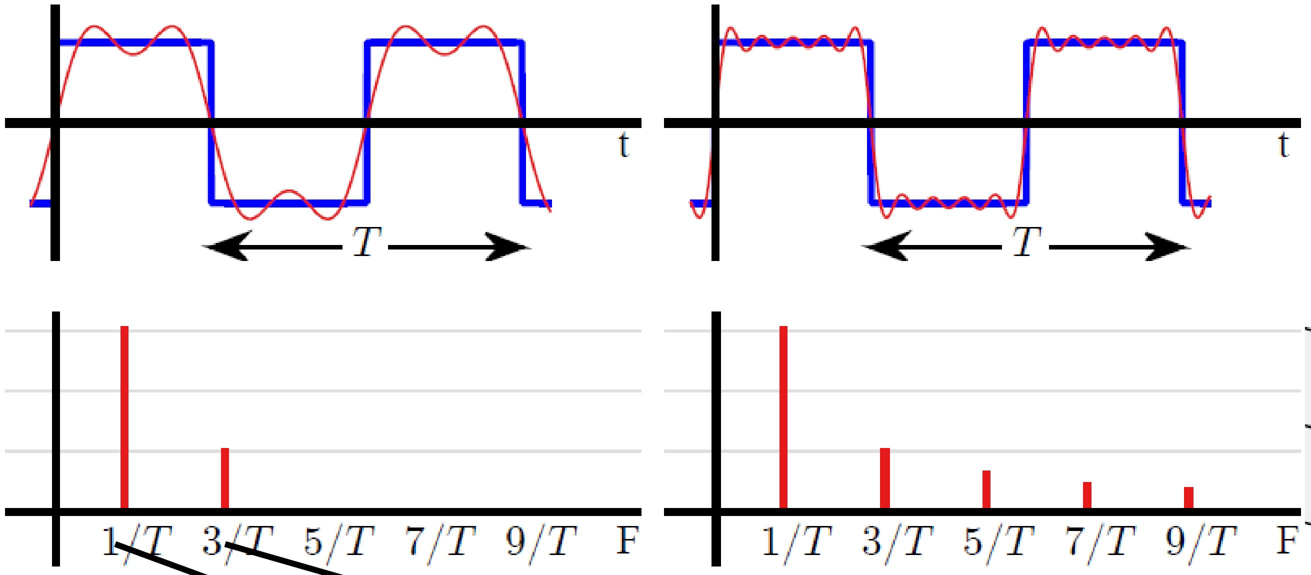
Sine in Frequency IQ-Plane



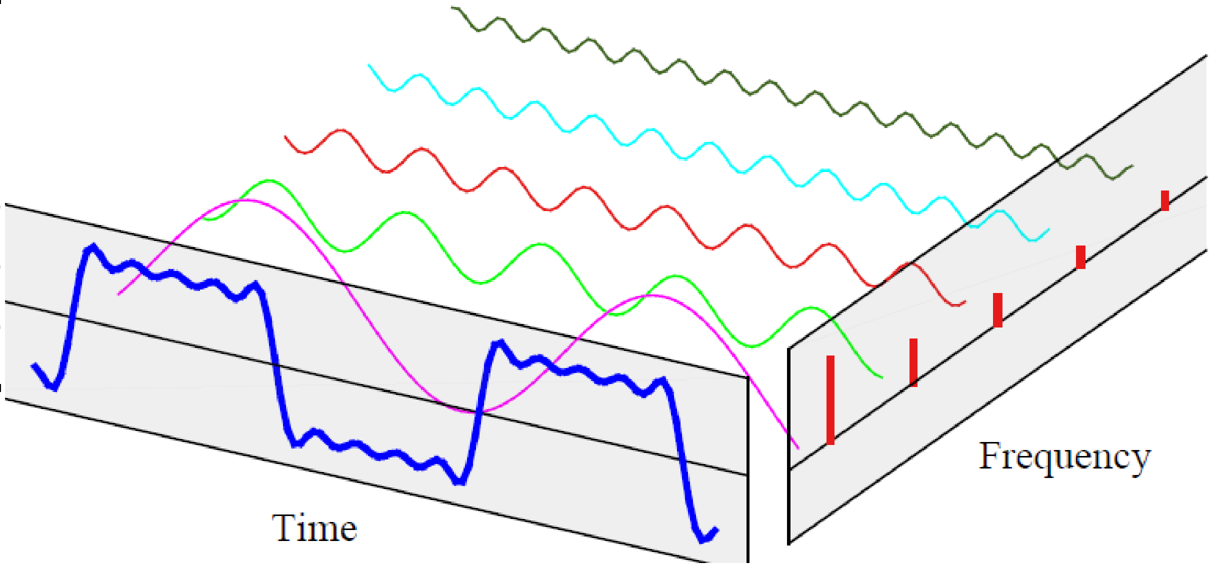
By rotating the time waveform clockwise by 90°, the sine wave can be put on time I-axis

Complex sinusoids form a basic unit of signal construction of any shape

5 sinusoidal components adding together to generate the approximate square wave in the time domain.



(a) 2 sinusoidal components (b) 5 sinusoidal components



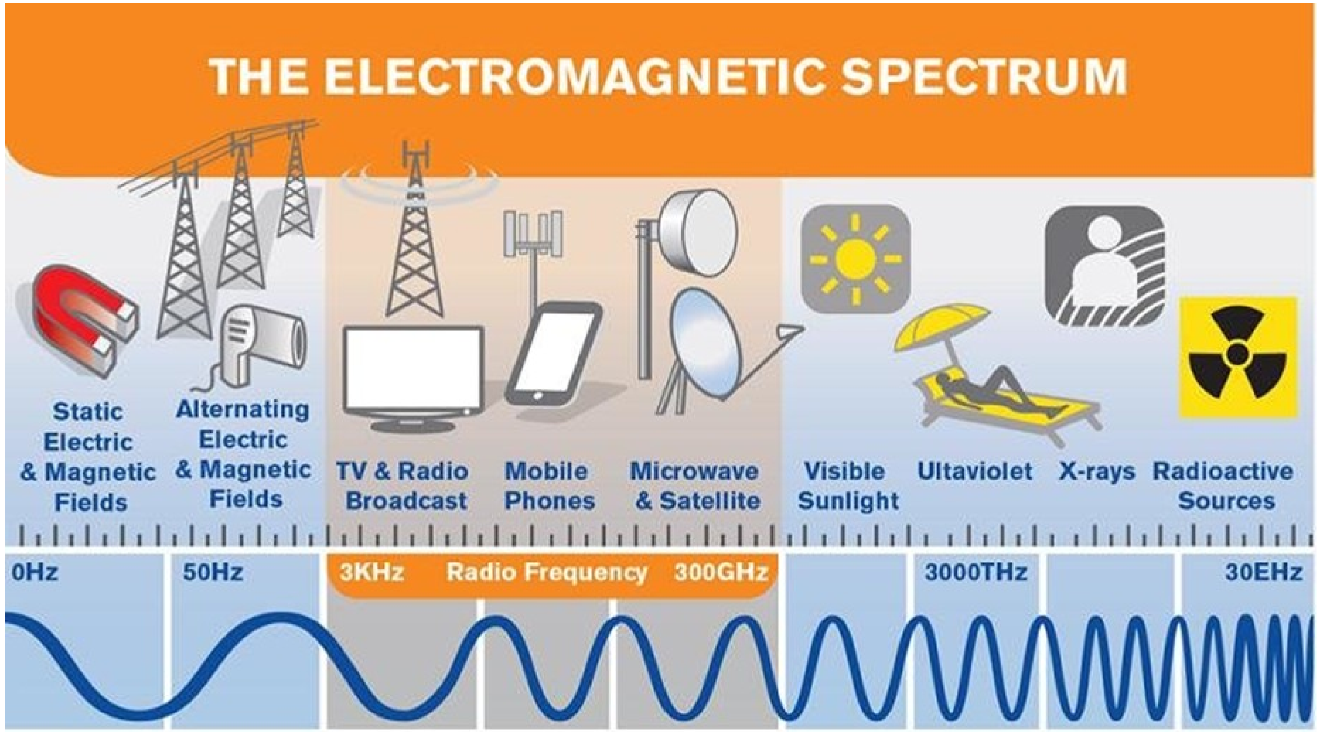
$$s(t) = \sin(2\pi F t) + \frac{1}{3}\sin(2\pi 3 F t) = \sin\left(2\pi \frac{1}{T} t\right) + \frac{1}{3}\sin\left(2\pi \frac{3}{T} t\right)$$

$$s(t) = \sin(2\pi F t) + \frac{1}{3}\sin(2\pi 3 F t) + \frac{1}{5}\sin(2\pi 5 F t) + \frac{1}{7}\sin(2\pi 7 F t) + \frac{1}{9}\sin(2\pi 9 F t)$$

Spectrum and Bandwidth

Spectrum contains the frequencies of complex sinusoids that sum up in time domain to form that signal

Bandwidth of a signal is the range of frequencies of complex sinusoids present in that signal



$$\mathrm{BW} = F_H - F_L$$

Band-limited signal

$$S(F) = \begin{cases} 0, & 0 \le |F| \le F_L \\ x, & |F_L| \le |F| \le |F_H| \\ 0, & F_H \le |F| \le \infty \end{cases}$$

Signals with fast irregular variations → wide bandwidth

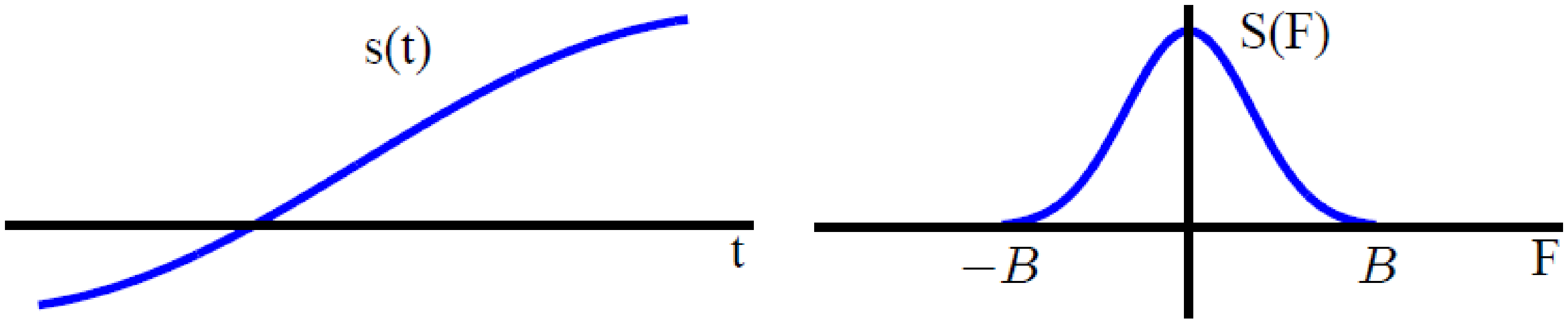
Frequency Support

- A signal cannot be limited in both time and frequency domains.
- A band-limited signal is then referred to as a signal with most of its energy concentrated within a certain amount of frequency range.

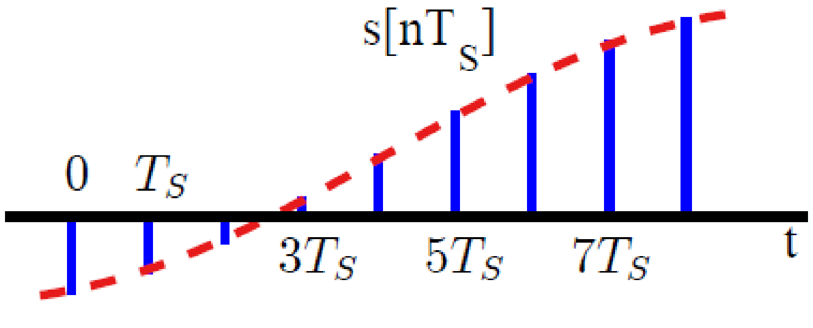
For example, Federal Communications Commission (FCC) defines bandwidth as the band in which 99% of the signal power is contained.

Sampling a Continuous-Time Signal

Sampling is the conversion of a continuous-time signal into a discrete-time signal obtained by taking the samples of the continuous-time signal at discrete-time instants



A continuous-time signal in time and frequency domains



Sampling a continuous-time signal

- Consider a band-limited continuous-time signal s(t) and its frequency domain representation S(F) with bandwidth B shown in above Figure
- A discrete-time signal s[n] can be obtained by taking samples of s(t) at equal intervals of T_S seconds.

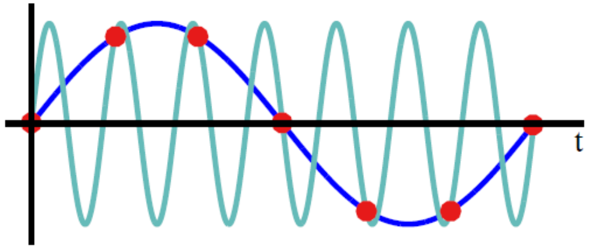
$$s[n] = s(t)\Big|_{t=nT_S}$$

T_S = Sampling period or sampling interval

While F_S = Sampling frequency = $1/T_S$

- Consider a continuous-time signal

$$s(t) = A\cos(2\pi F t + \theta)$$



$$s[n] = s(t)\big|_{t=nT_S}$$

- Sampling of the same signal at a rate Fs =1/Ts

$$= A\cos(2\pi F n T_S + \theta) = A\cos(2\pi F\, n/F_S + \theta)$$

$$= A\cos(2\pi\, F/F_S\, n + \theta)$$

F/Fs above is the frequency of a discrete-time sinusoid s[n]

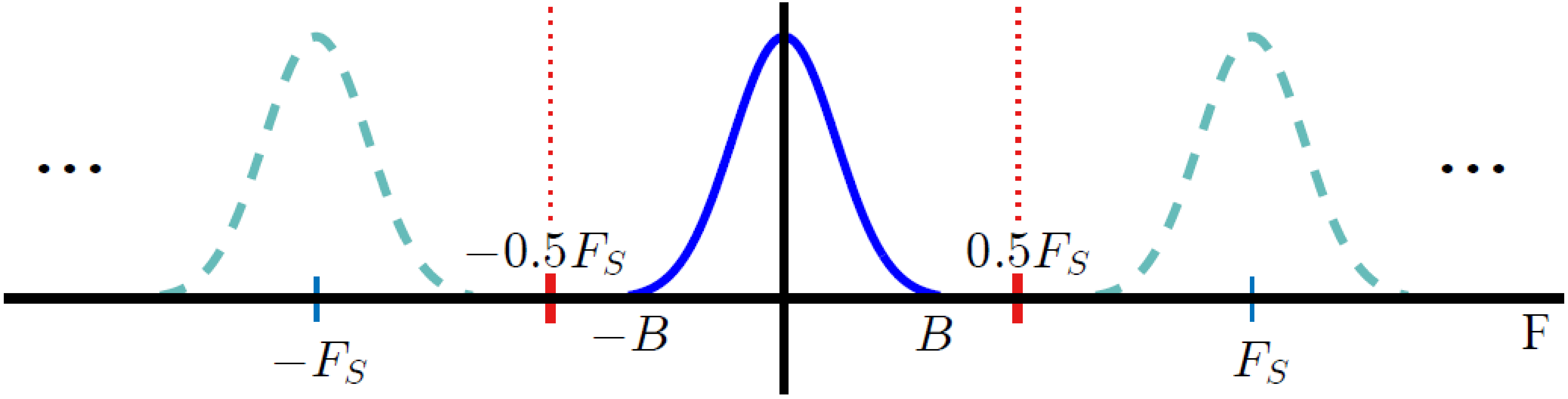
- Let us sample at the same rate Fs another sinusoid with continuous frequency F+kFs, where k = ±1,±2, · · ·

- s[n] = Acos {2π(F + kFs)nTs + θ}

$$= A\cos\left(2\pi\frac{F + kF_S}{F_S}n + \theta\right) = A\cos\left(2\pi\frac{F}{F_S}n + 2\pi k n + \theta\right)$$

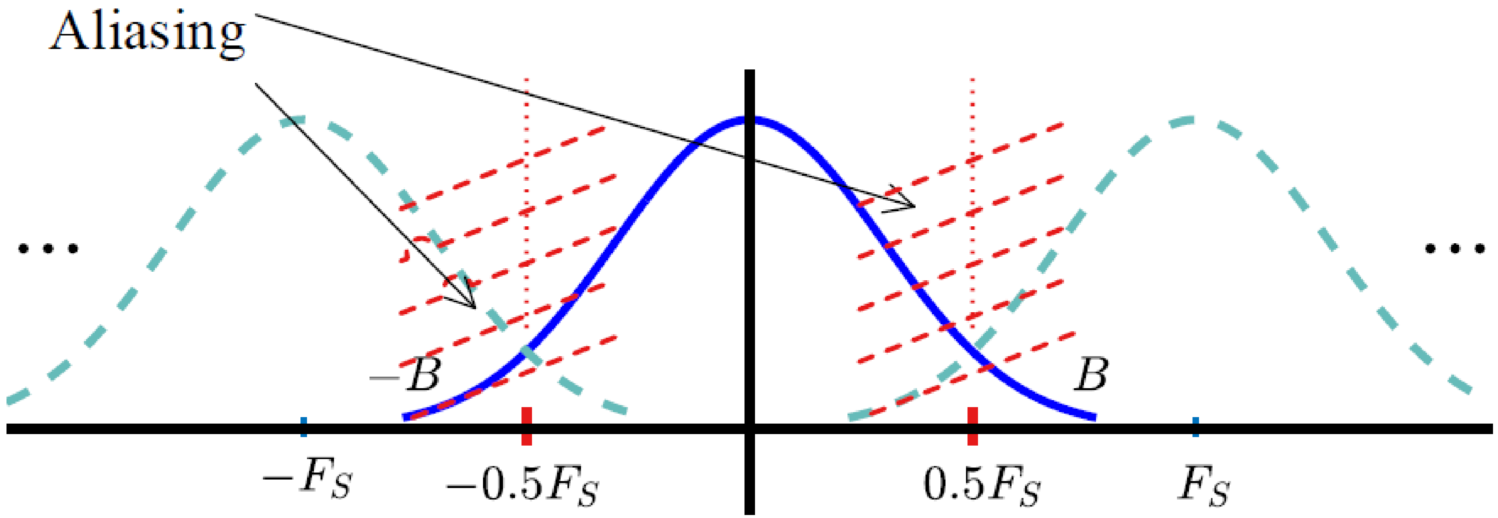
$$= A\cos\left(2\pi\frac{F}{F_S}n + \theta\right)$$

- This expression is the same as the discrete-time sinusoid



Spectrum after sampling. Dotted red lines indicate the baseband or Nyquist band

If a continuous-time signal has a bandwidth B greater than 0.5Fs, it will appear as an alias



Digital Filters

- In signal processing, the function of a filter is to remove unwanted parts of the signal, such as random noise, or to extract useful parts of the signal, such as the components lying within a certain frequency range.
- The primary functions of a filter are:
    a) To confine a signal into a prescribed frequency band or channel for example as in an anti-aliasing filter or a radio/TV channel selector,
    b) To decompose a signal into two or more sub-band signals for sub-band signal processing, for example in music coding,
    c) To modify the frequency spectrum of a signal, for example in audio graphic equalizers, and
    d) To model the input-output relation of a system such as a mobile communication channel, voice production, musical instruments, telephone line echo, and room acoustics.

Nyquist Rate or Sampling theorem

- The highest frequency (or bandwidth) B of a continuous-time signal s(t) should be less than 0.5Fs to prevent any distortion in the sampled signal s[n]

$$\boxed{F_S > 2B}$$

Passband and Stopband

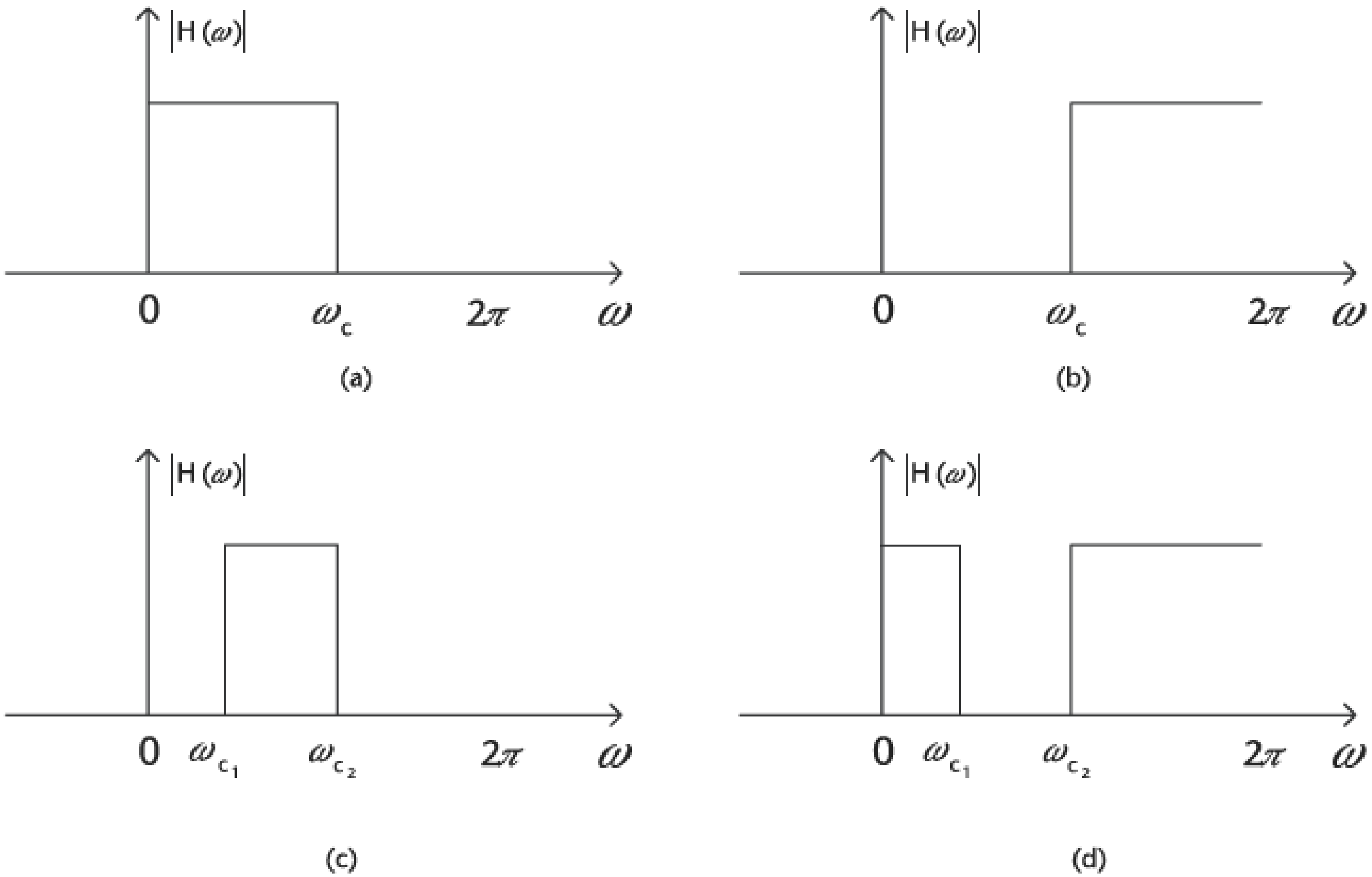


Figure 2.40 Ideal magnitude response characteristics of four types of filters on the frequency range $[0, 2\pi]$. (a) Lowpass filter, (b) highpass filter, (c) bandpass filter, where $(\omega_{c_1}, \omega_{c_2})$ is passband, and (d) bandstop filter, where $(\omega_{c_1}, \omega_{c_2})$ is stopband.