

Iqraq's Note

# Fundamental Signal Processing

# What is Signal Processing (SP)

- Field of science involving manipulation of signals to get the desired shaping
- Change from time domain to frequency and vice versa
- Smoothing, separating noise, filtering, extracting
- - Nature signal exist in continuous manner
  - $t_1, t_2 = -\infty, \infty$

# Flow of (SP)

- Need to convert to digital for computer to read



**Fig : Basic Elements of Signal Processing System**

# AD conversion

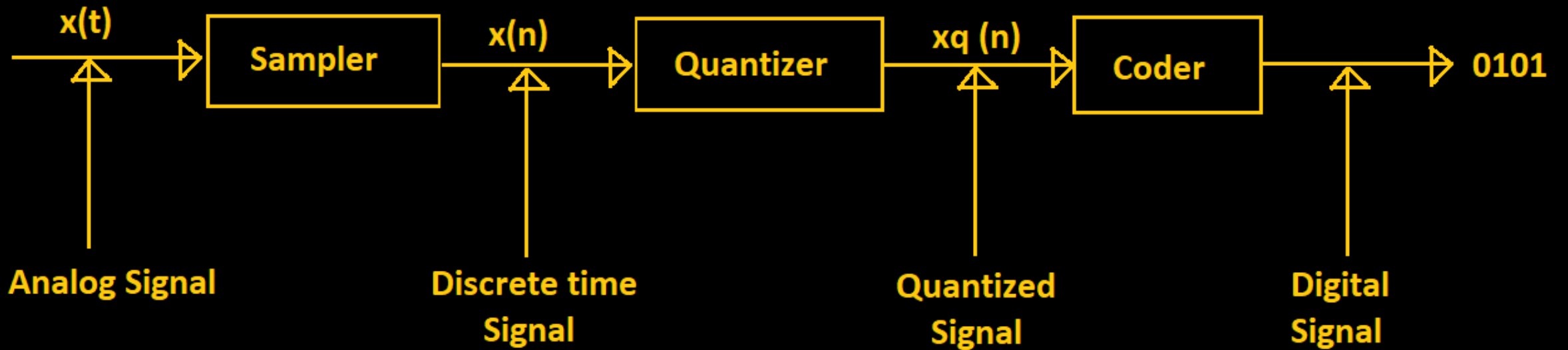


Fig : Elements of Analog to Digital Convertor

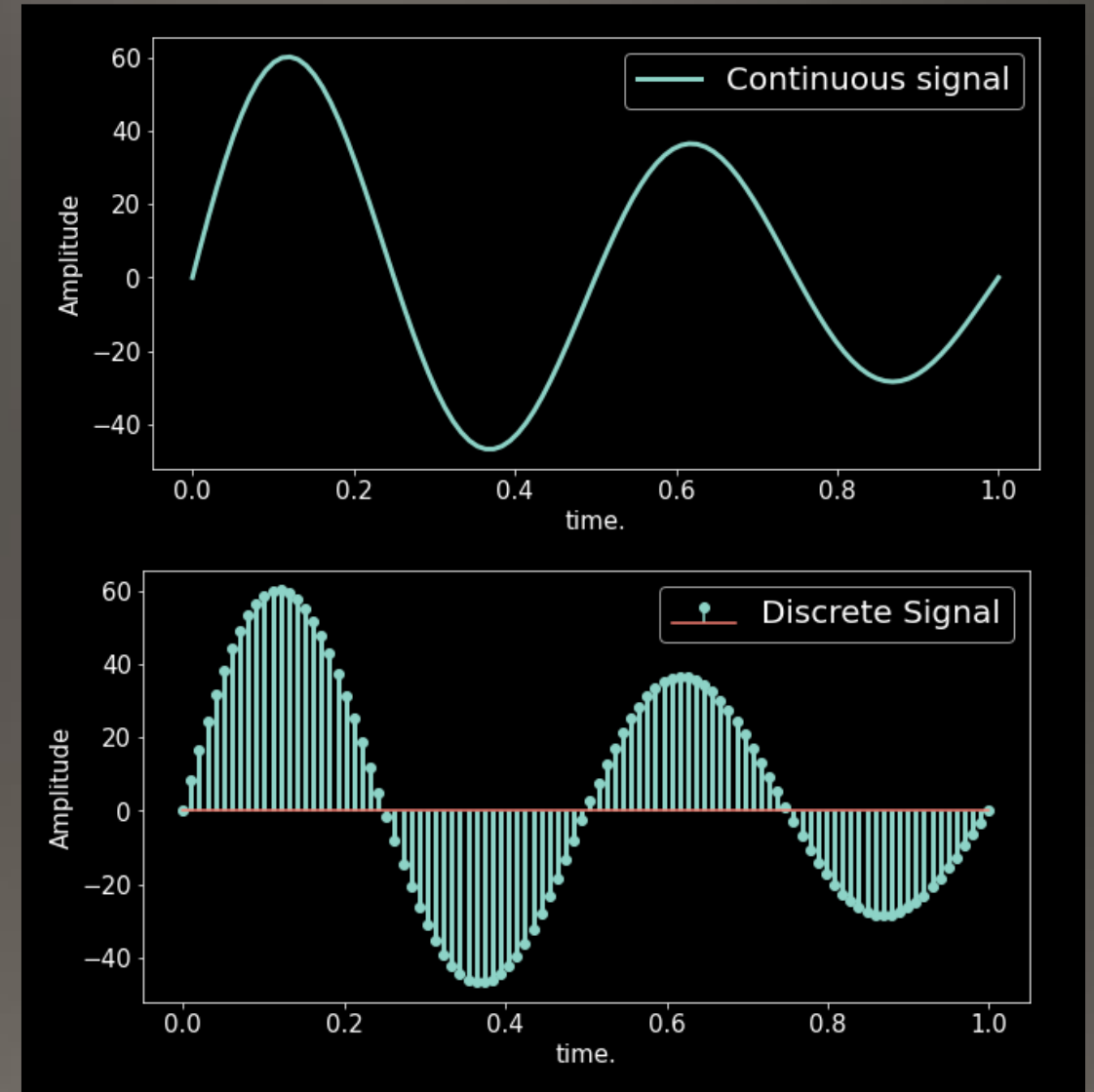
# What is sampling?

- the reduction of a continuous-time signal to a discrete-time signal
- To know the value at instantaneous point
- Need to define sampling interval (T) by setting sampling frequency

- $f_s = 1\text{Hz}$

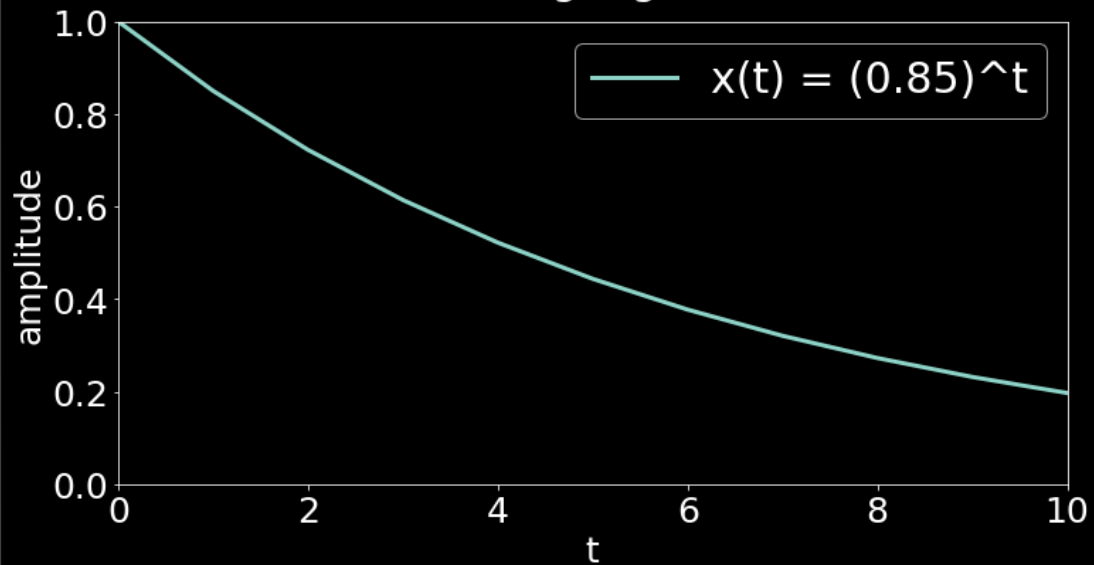
- $T = \frac{1}{f_s}$

- $T = 1\text{s}$

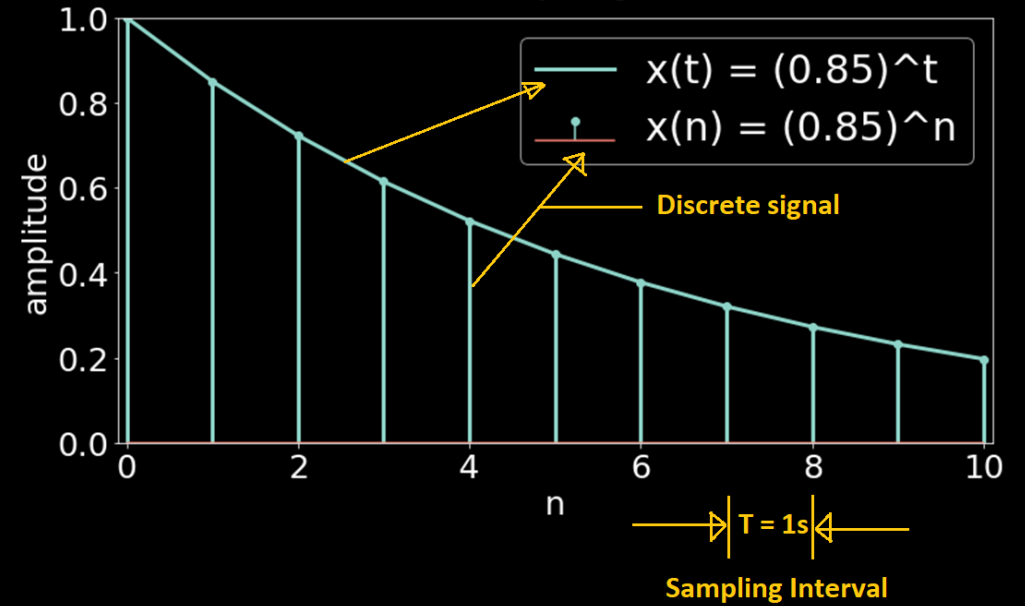


# Example

Analog Signal



Sampling



For sampling replace  $t = nT$ .

Thus,

$$x(nT) = (0.85)^{nT}$$

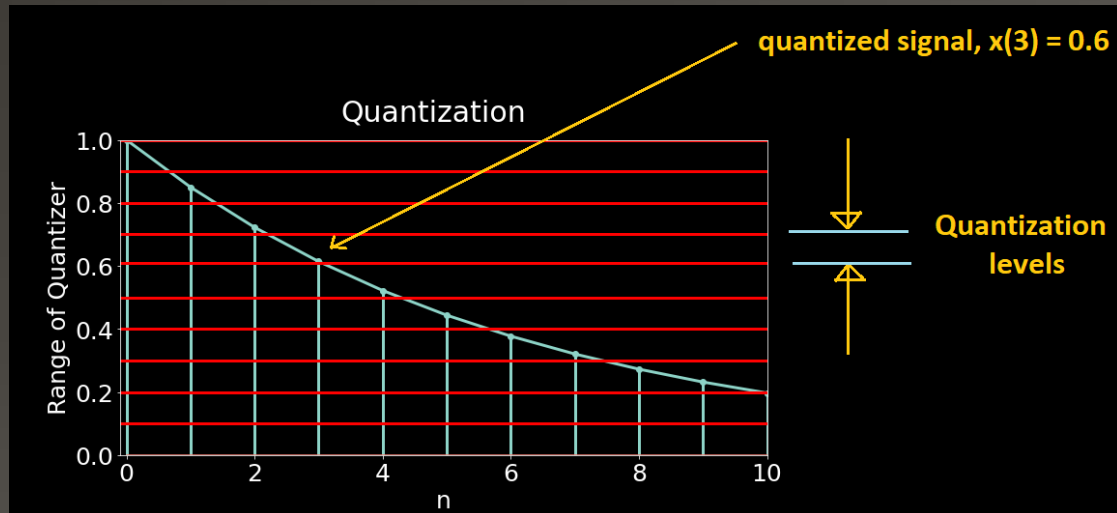
Since  $T = 1s$ , therefore,

$$x(n) = (0.85)^n$$

$x(n)$  is the discrete time signal with sampling interval of 1s.

# What is Quantization

- Process of converting amplitude of discrete signal into digital signal by expressing each sample value as a finite
- Accuracy = how many discrete levels are allowed to represent the magnitude of signal.



Discrete time ( $n$ )	Discrete signal $(0.85)^n$	Quantized signal
0	1	1
1	0.85	0.9
2	0.7224999999999999	0.7
3	0.6041249999999999	0.6
4	0.5220062499999999	0.5
5	0.44370531249999995	0.4
6	0.37714951562499993	0.4
7	0.3205770882812499	0.3
8	0.27249052503906246	0.3
9	0.23161694628320306	0.2
10	0.1968744043407226	0.2

Quantized signal after rounding discrete time signal

# Coding

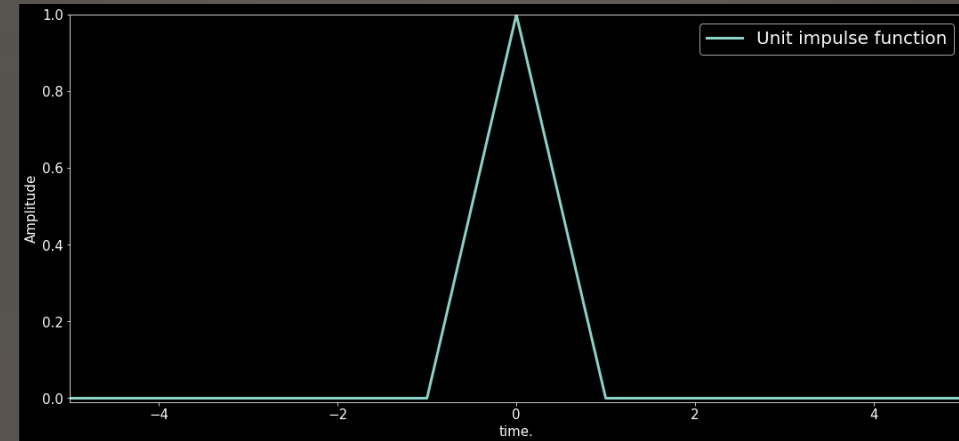
- Higher bit number will have higher quantized level –reading are more accurate
- Example of 4 bit ( $b_0 - b_3$ )

$b_3$	$b_2$	$b_1$	$b_0$	Quantized Levels
0	0	0	0	0.0
0	0	0	1	0.1
0	0	1	0	0.2
0	0	1	1	0.3
0	1	0	0	0.4
0	1	0	1	0.5
0	1	1	0	0.6
0	1	1	1	0.7
1	0	0	0	0.8
1	0	0	1	0.9
1	0	1	0	1.0

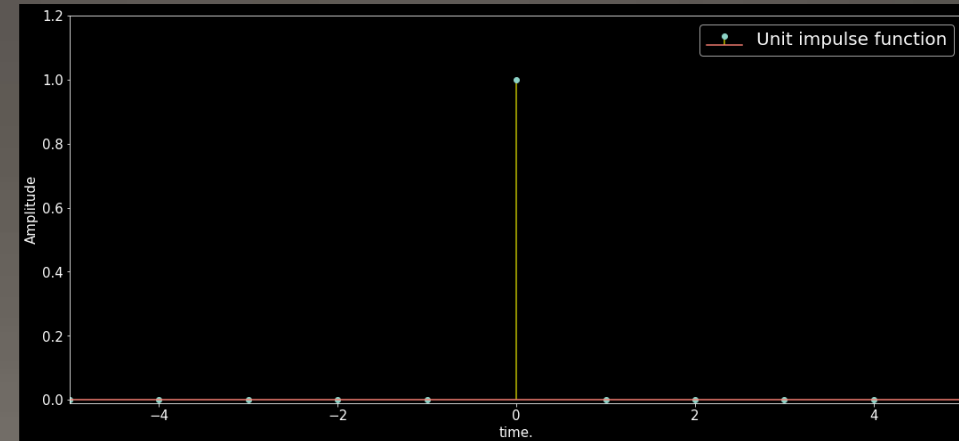


# Fundamentals for Positive continuous time signal ( $t > 0$ ) and discrete time signal

- Unit impulse signal
- $S(t) = 1$  for  $t = 0$   
and
- $S(t) = 0$  for  $t \neq 0$



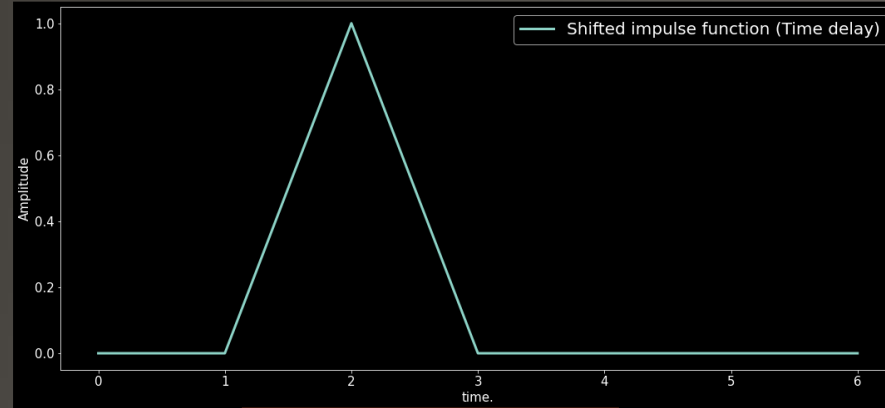
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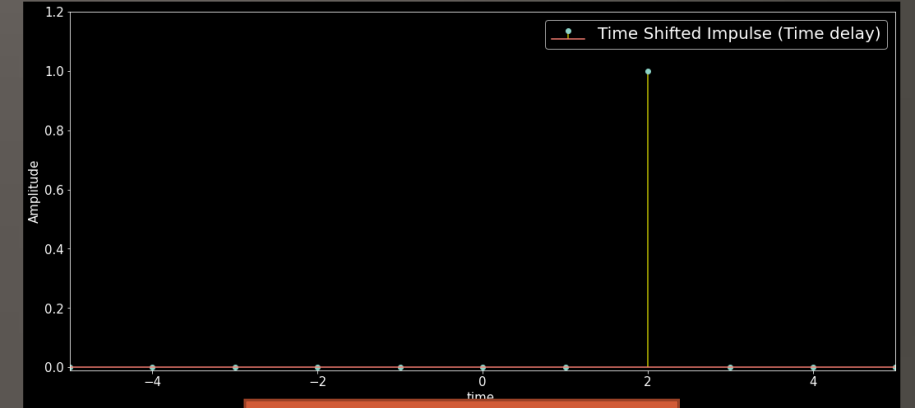
Discrete

## 1. Time Delay

1. Move signal towards positive time axis
2.  $S(t - 2) = 1$



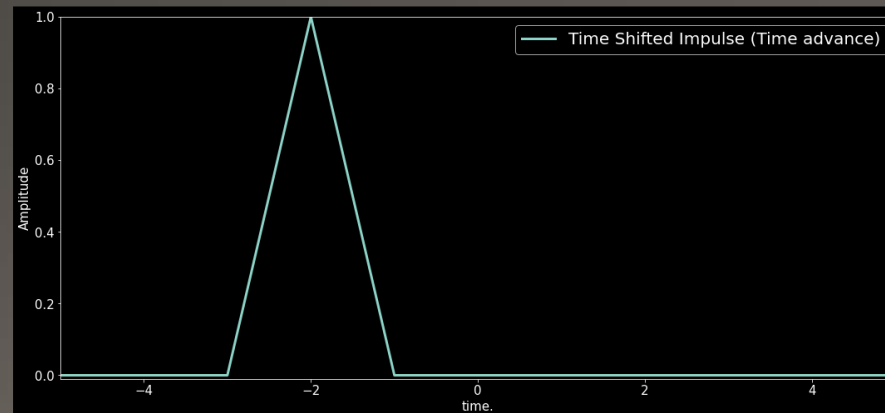
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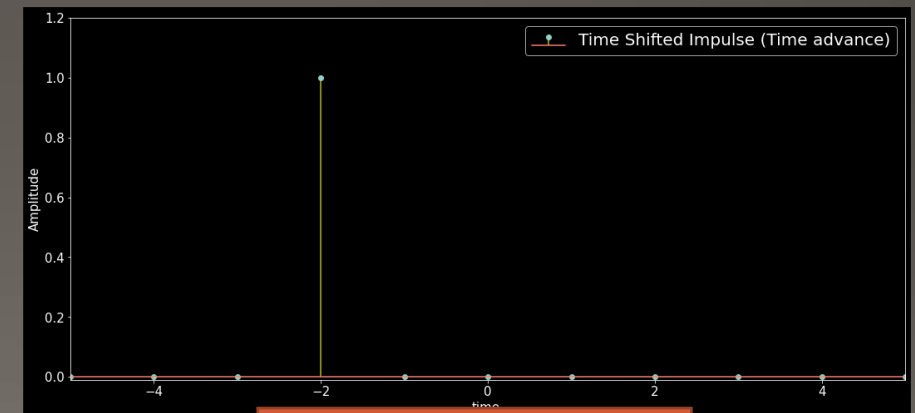
Discrete

## 2. Time advance

1. Move signal towards negative time axis
2.  $S(t - 2) = 1$



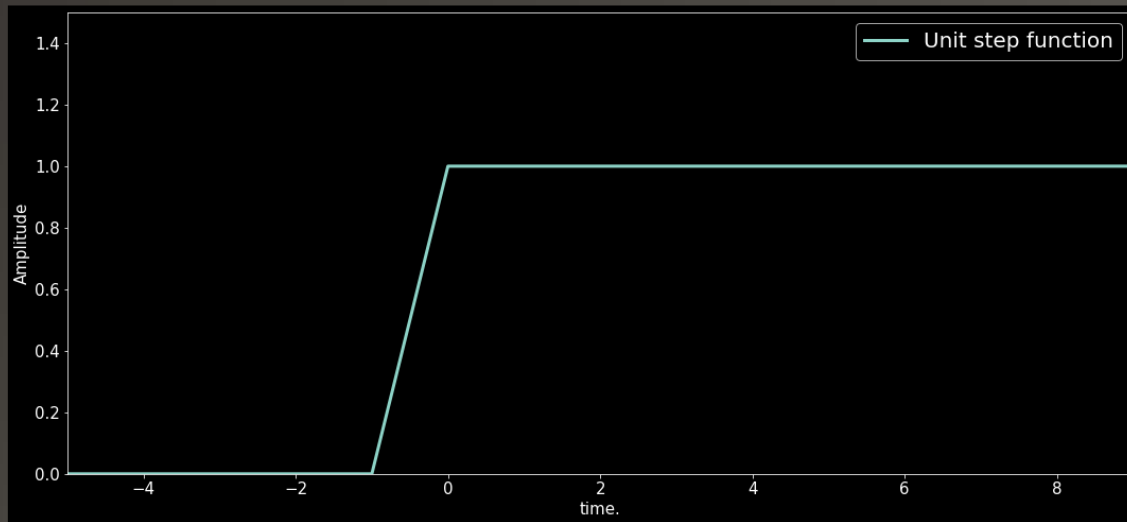
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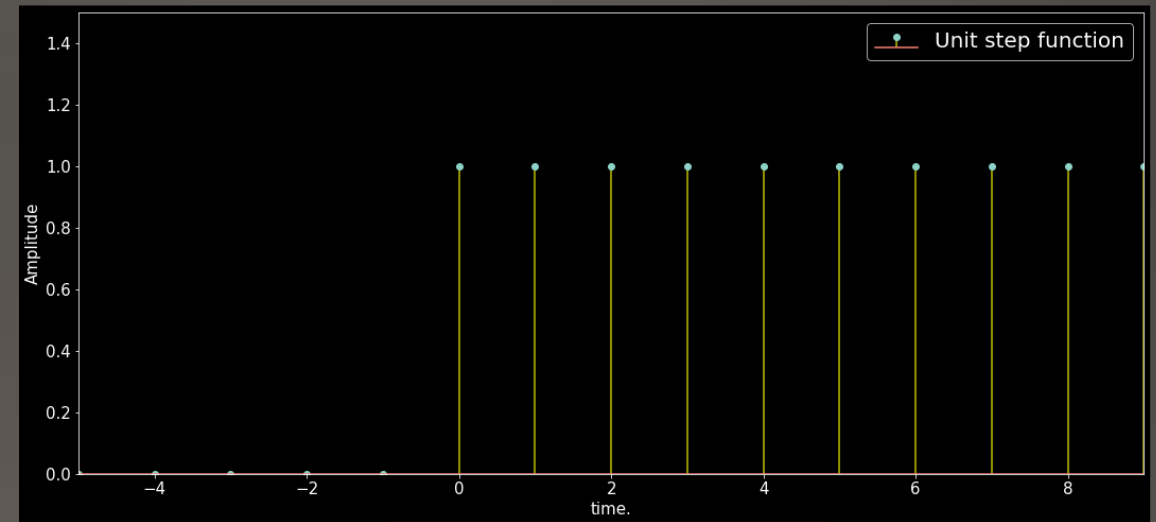
Discrete

### 3. Unit Step Signal

1. Function having magnitude of 1 at time equal to and greater than zero
2.  $u(t) = 1$  for  $t \geq 0$  &  $u(t) = 0$  for  $t < 0$



Continuous

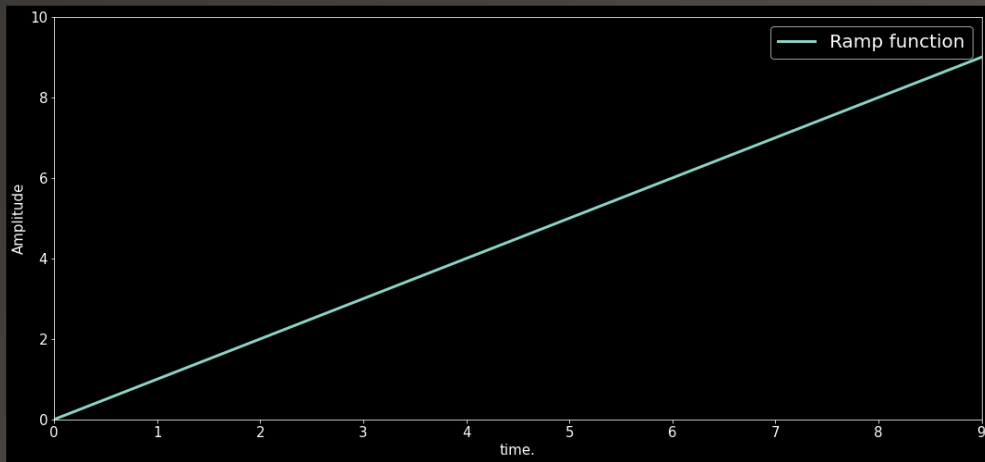


Discrete

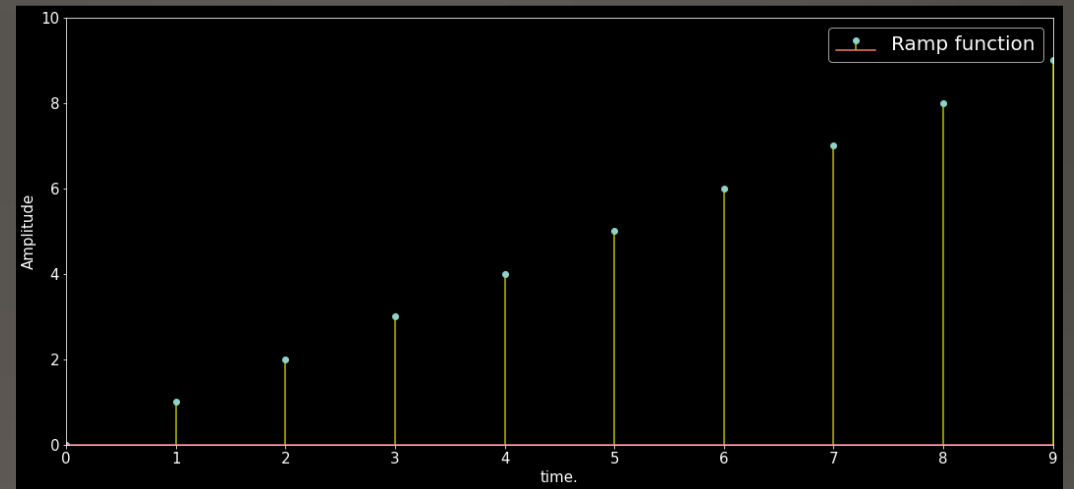
## 4. Unit Ramp Signal

- Function having a magnitude of  $t$  at  $t \geq 0$

1.  $x(t) = 1$  for  $t \geq 0$  &  $x(t) = 0$  for  $t < 0$



Continuous



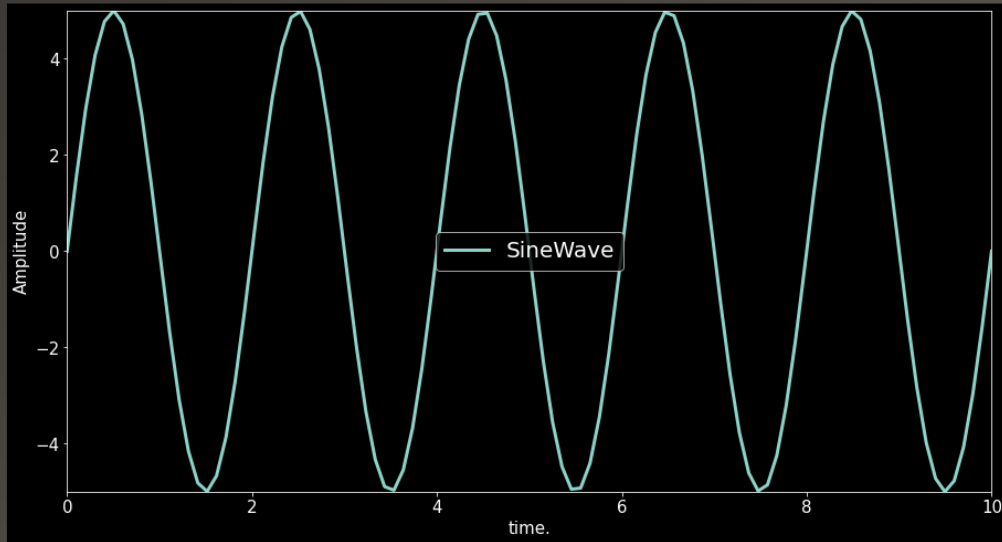
Discrete

## 5. Sinusoidal Signal

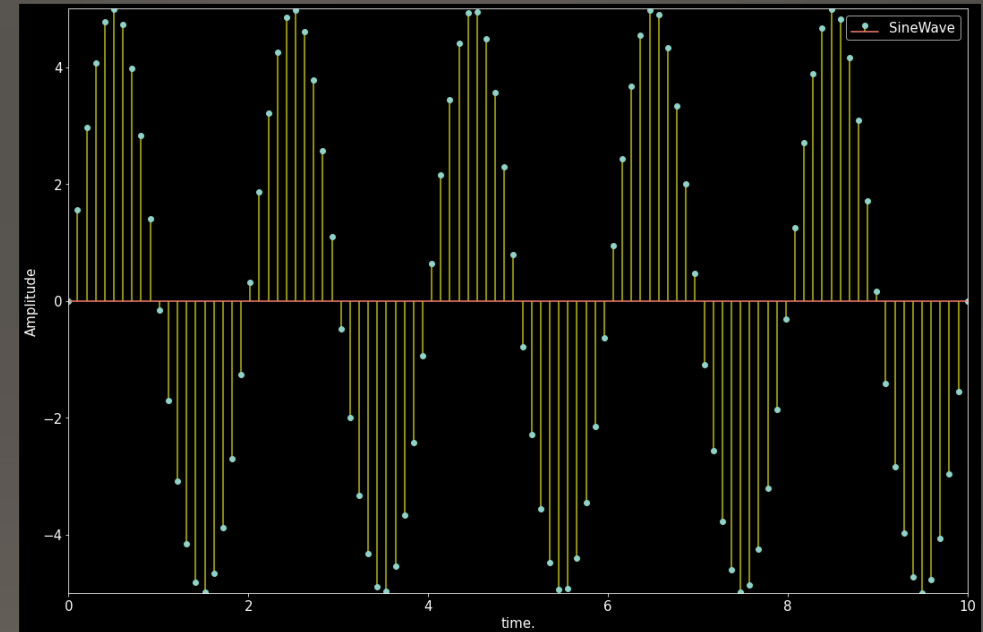
- Oscillations that repeat over a fixed interval of time period of the signal

1.  $x(t) = A\sin(2\pi ft)$

2.  $A$  – Amplitude,  $f$  = Frequency



Continuous



Discrete

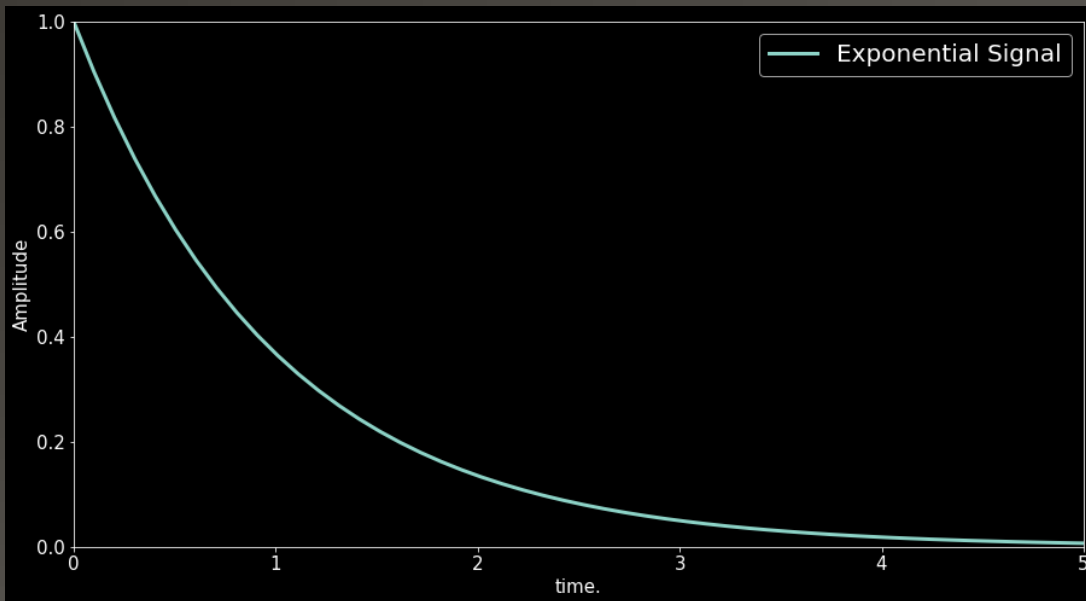
## 5. Unit Exponential Signal

1. 1 at  $t(0)$  and exponentially decaying for time greater than zero, for  $t > 0$

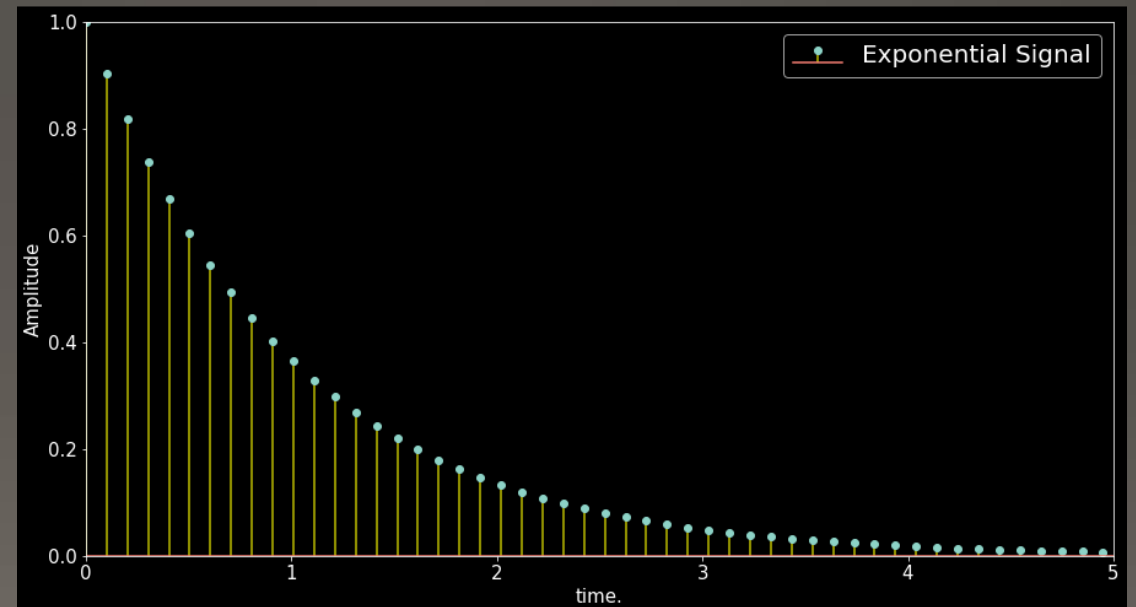
2.  $x(t) = 1$  for  $t = 0$

and

$$x(t) = e^{-t} \text{ for } t > 0$$



Continuous



Discrete

# Sampling and reconstruction

- Convert continuous time signal  $x(t)$  to discrete  $x(n)$  by replacing  $t$  with  $nT$ .
- $x(n) = x(nT)$
- Where  $x(n)$  is the discrete time signal obtained by taking samples of the continuous time signal  $x(t)$  every  $T$  seconds. The  $T$  between successive samples is called the sampling period.
- $T = \frac{1}{f_s}$
- Where  $f_s$  is called sampling rate or the sampling frequency (Hz)

# Nyquist sampling theorem

- States that the sampling frequency should be greater or equal than twice the maximum frequency of the signal

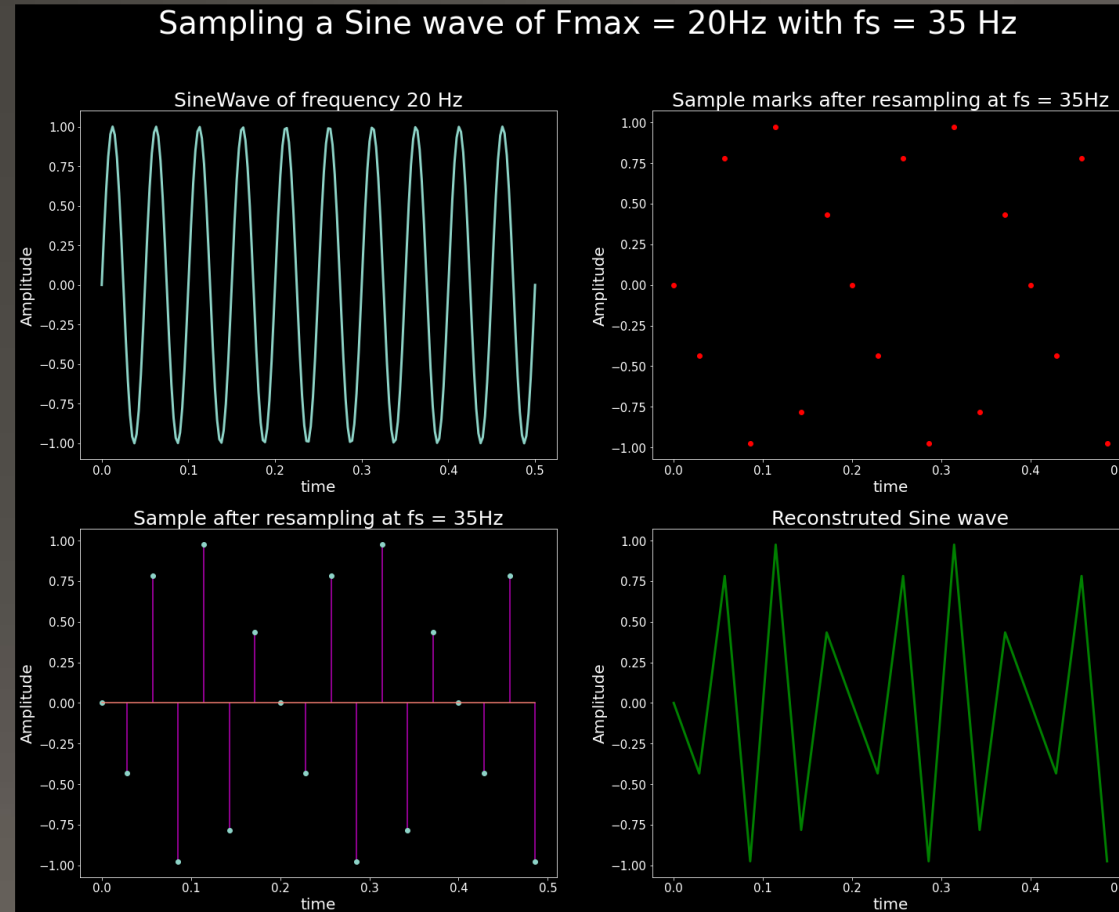
If  $f_{\max}$  is the maximum frequency of the signal then

$$F_s \geq 2F_{\max}$$

This theorem is important if we want to reconstruct the signal after sampling.



# Lower sample rate – low accuracy of reconstructed signal



# Higher sampling rate = higher accuracy of reconstructed signal

