

# UNIT 8

## Digital Modulation Techniques (PCM, DPCM, DM and ADM)

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# Outcomes

- Define PCM Principles: Sampling, Quantization and Encoding
  - Describe PCM Techniques: PCM, DPCM, DM and ADM
  - Discuss the issues related PCM techniques and possible solutions
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# Type of Modulation

## Analog Modulation



### Continuous Wave Modulation



Carrier signal

AM – Amplitude Modulation  
FM – Frequency Modulation  
PM – Phase Modulation

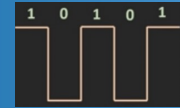
### Pulse Modulation



Carrier signal

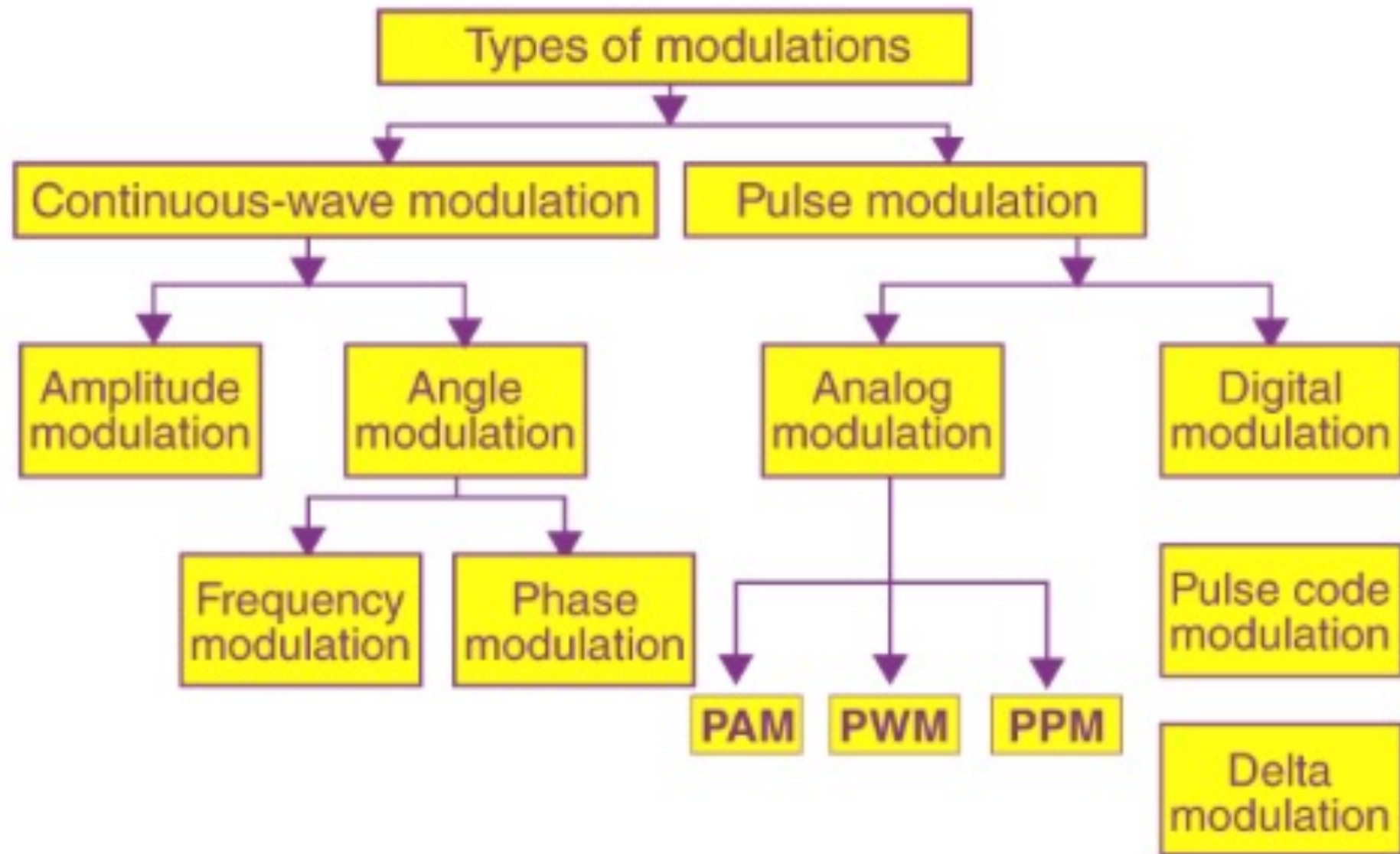
PAM – Pulse Amplitude Modulation  
PWM – Pulse Width Modulation  
PPM – Pulse Position Modulation

## Digital Modulation



ASK – Amplitude Shift Keying  
FSK – Frequency Shift Keying  
PSK – Phase Shift Keying

PCM – Pulse Code Modulation  
(Delta Modulation and Adaptive  
delta Modulation)

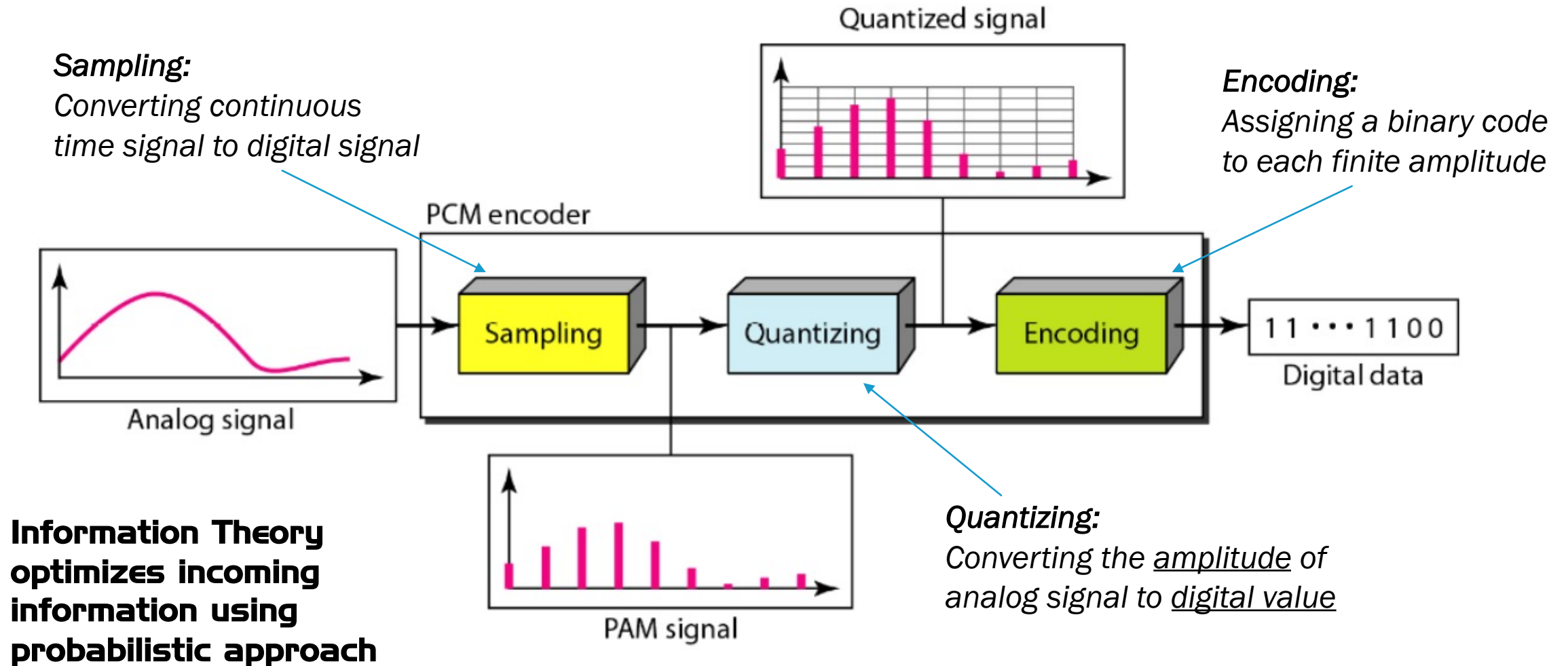




# PULSE CODE MODULATION



# Overview Pulse Code Modulation



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# Sampling

To convert the signal from continuous time to discrete time, a process called sampling is used.

The value of the signal is measured at certain interval in time. Each measurement refers to as a **sample**.

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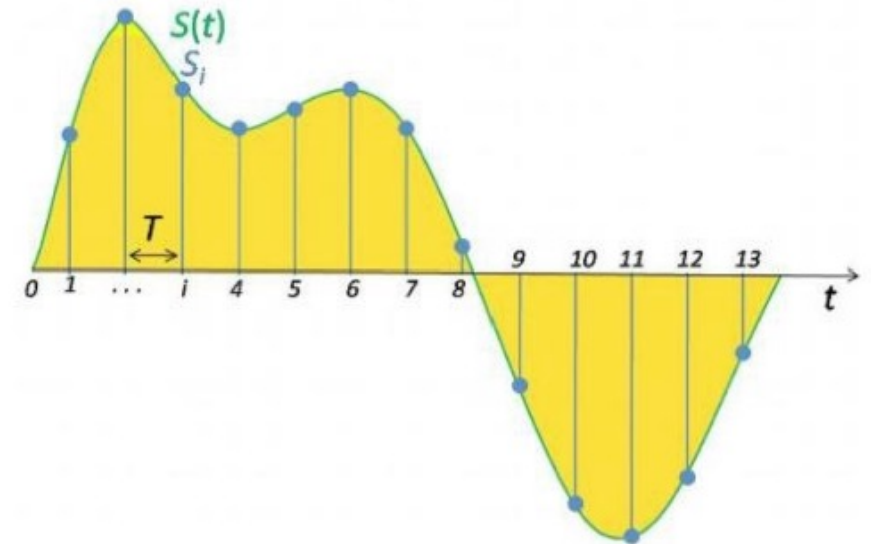
# Sampling cont.

- Sampling a signal: Analog  $\rightarrow$  Digital conversion by reading the value at discrete points
- The process of taking samples of information signal at a rate of **Nyquist's sampling frequency**

## Nyquist's sampling theorem

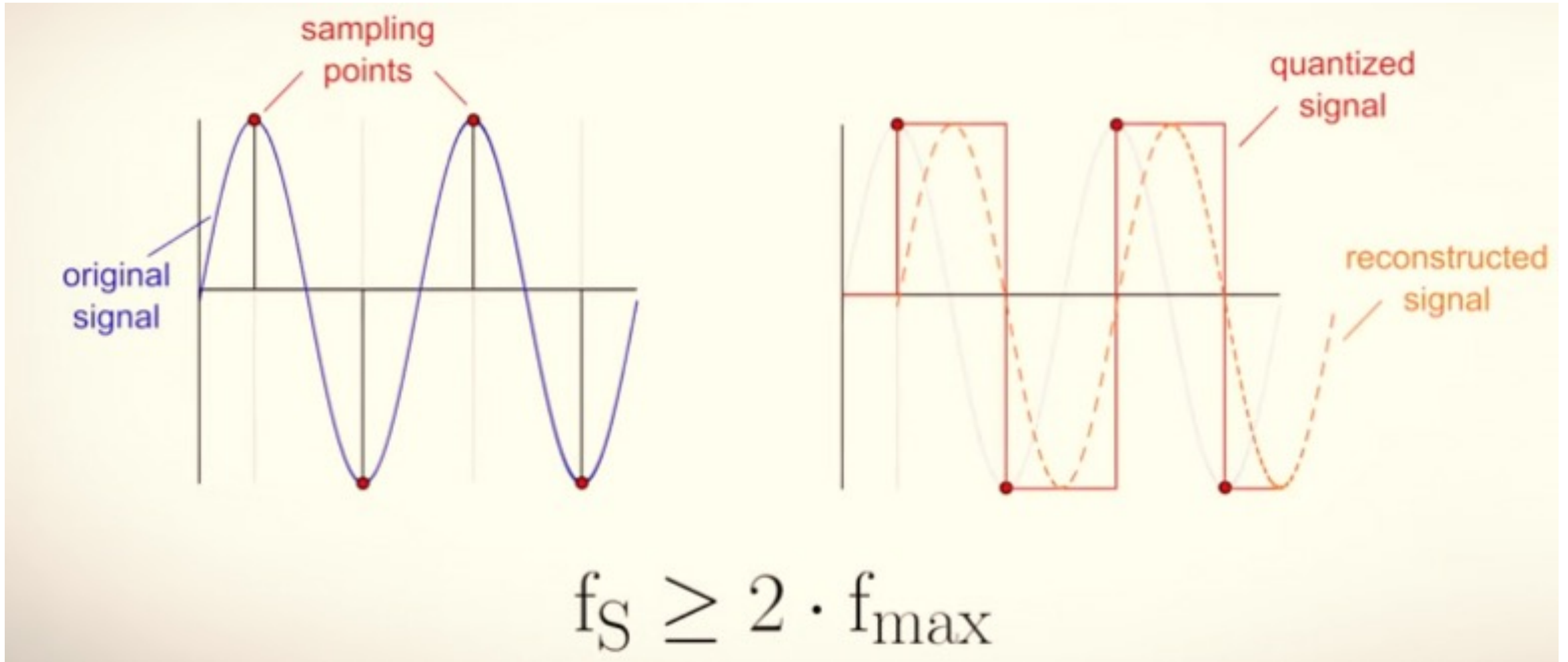
Original information signal can be reconstructed at the receiver with minimum distortion if the sampling rate the pulse modulation system equal to or greater than twice the maximum information signal frequency

$$f_s \geq 2f_{max}$$



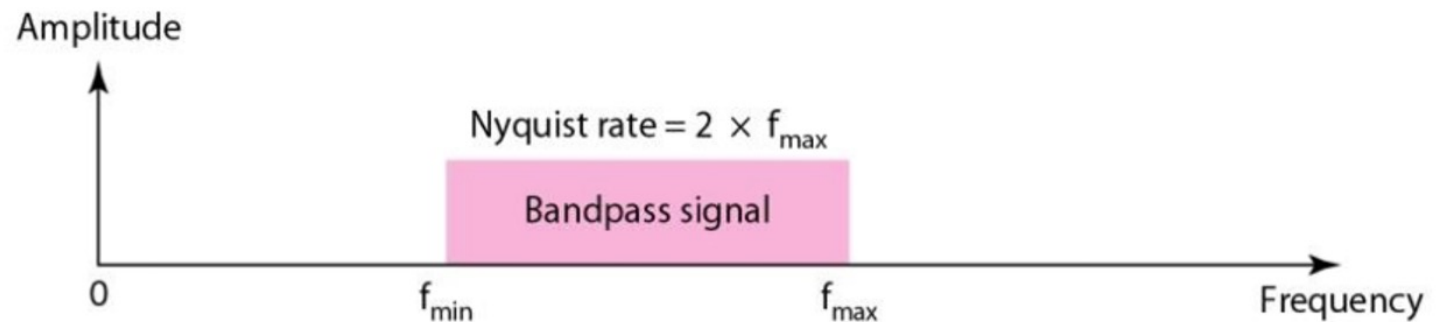
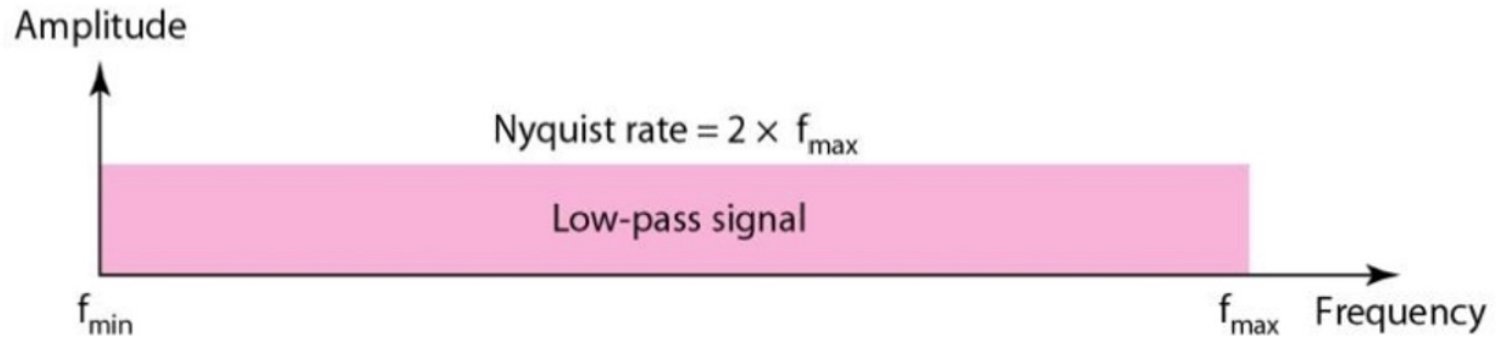


# Sampling – Nyquist Theorem



# Sampling Cont.

- Infinite bandwidth cannot be sampled.
- The sampling rate must be at least 2 times the highest frequency, not the bandwidth



# Sampling examples

## Question 1:

- A complex low-pass signal has a BW of 200 kHz. What is the minimum sampling rate?

## Answer

- The BW of the low-pass signal is between 0 and  $f_{\max}$
- $f_{\max} = 200 \text{ KHz}$ ,
- Therefore, minimum sampling rate,  $f_s$  is 400,000 samples per second

## Question 2

- A complex bandpass signal has a bandwidth of 200kHz. What is the minimum sampling rate for this signal

## Answer

- We cannot find the minimum sampling rate in this case because we do not have where the BW starts or ends. We don't know maximum frequency in the signal.

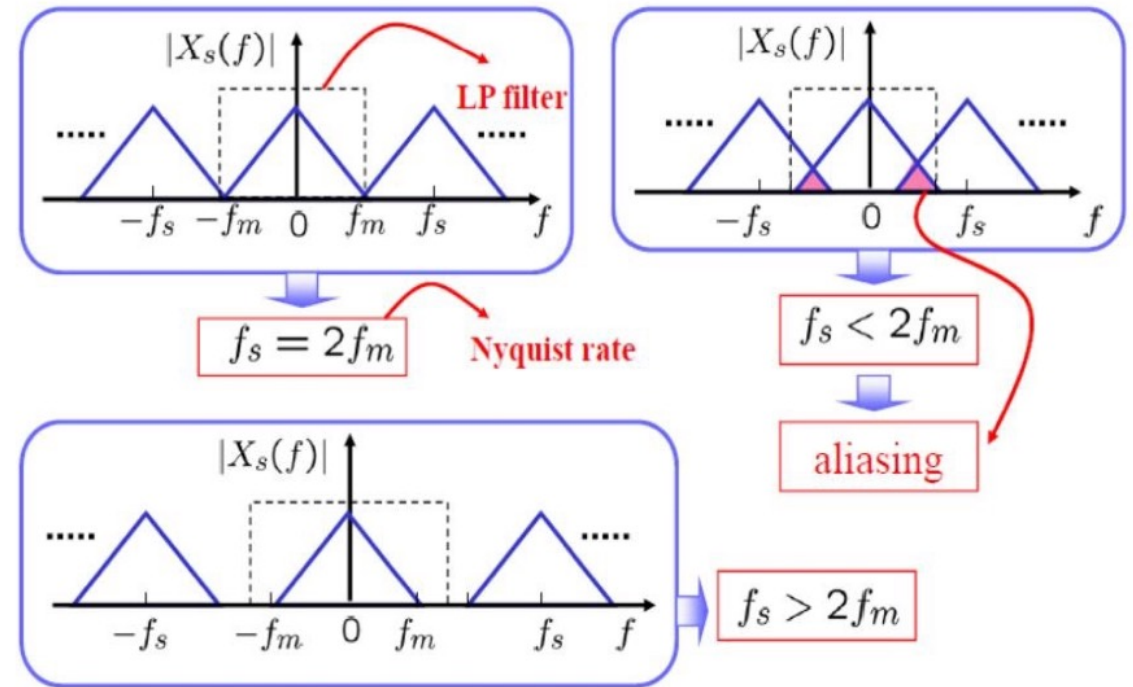
# Undersampling and oversampling

## Undersampling

- Undersampling is essentially sampling too slowly, or sampling at a rate below the Nyquist frequency for a particular signal of interest.
- Undersampling leads to **aliasing** and the original signal cannot be properly reconstructed

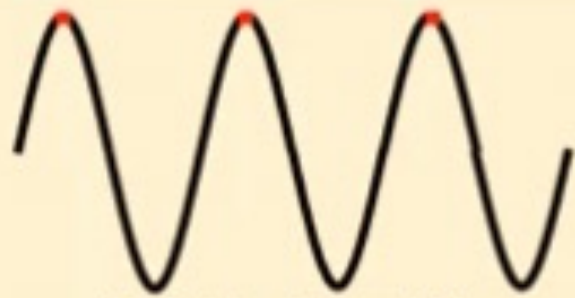
## Oversampling

- Oversampling is a sampling at a rate beyond twice the highest frequency component of the interest in the signal and is usually desired.



Effect of aliasing

# Impact of Nyquist theorem



100 Hz Sine Wave

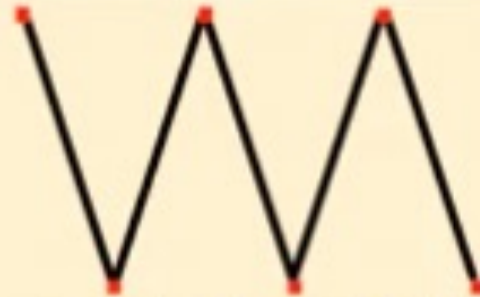


Aliased Signal

Sampled at 100 Hz

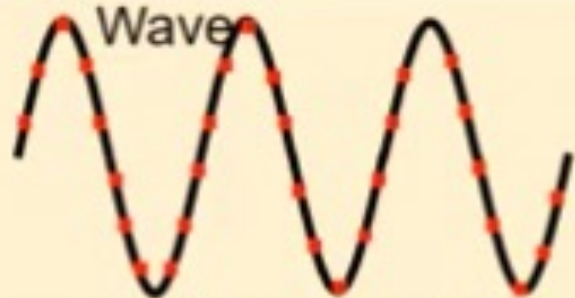


100 Hz Sine Wave

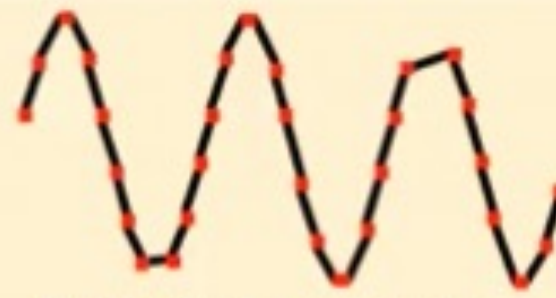
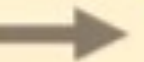


Adequately Sampled  
for Frequency Only

Sampled at 200 Hz



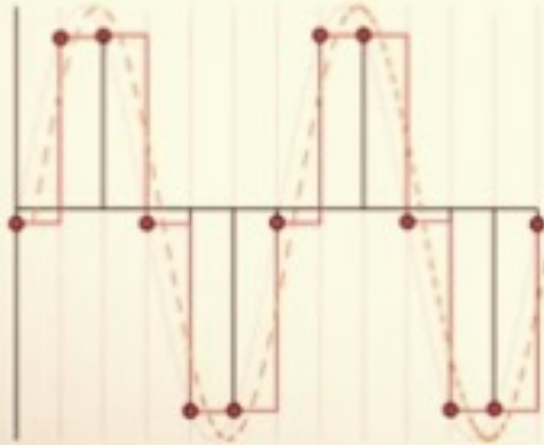
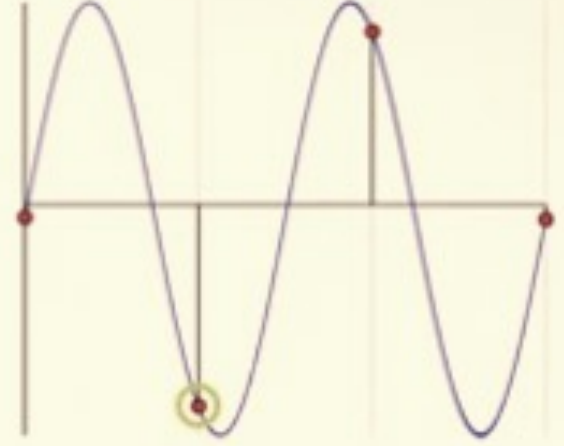
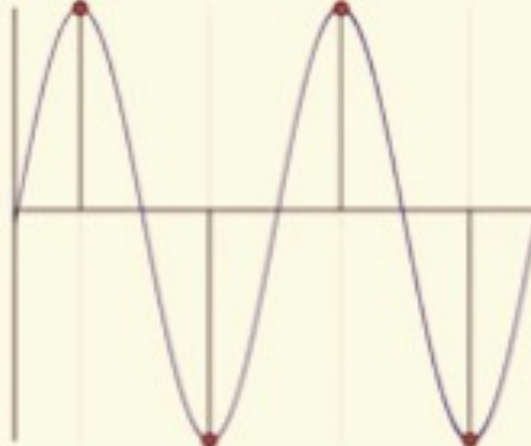
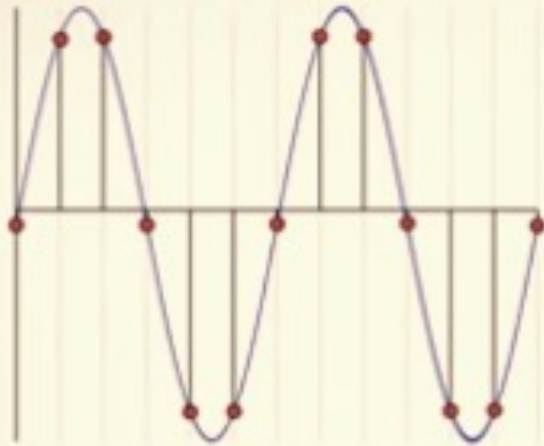
100 Hz Sine Wave



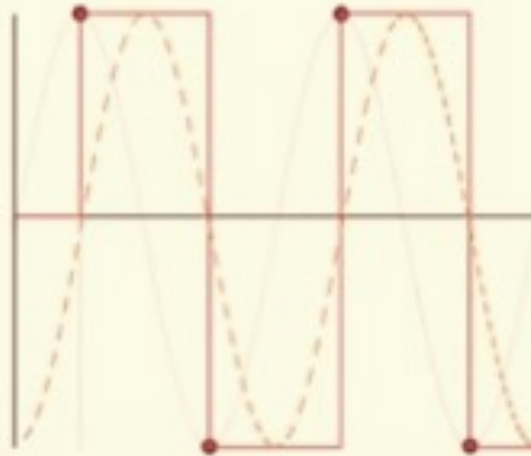
Adequately Sampled  
for Both Frequency  
and Shape

Sampled at 1 kHz

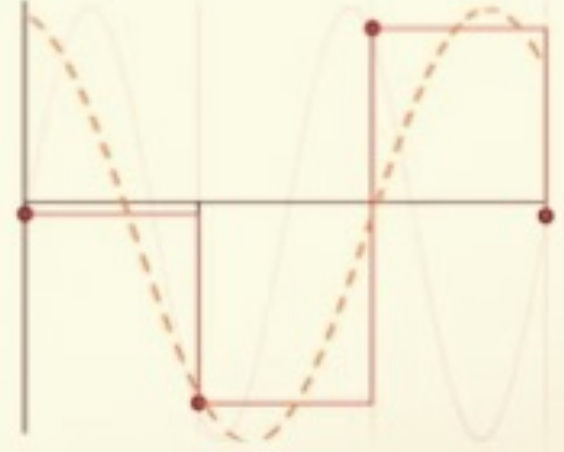
# Sampling and Aliasing



reconstruction OK



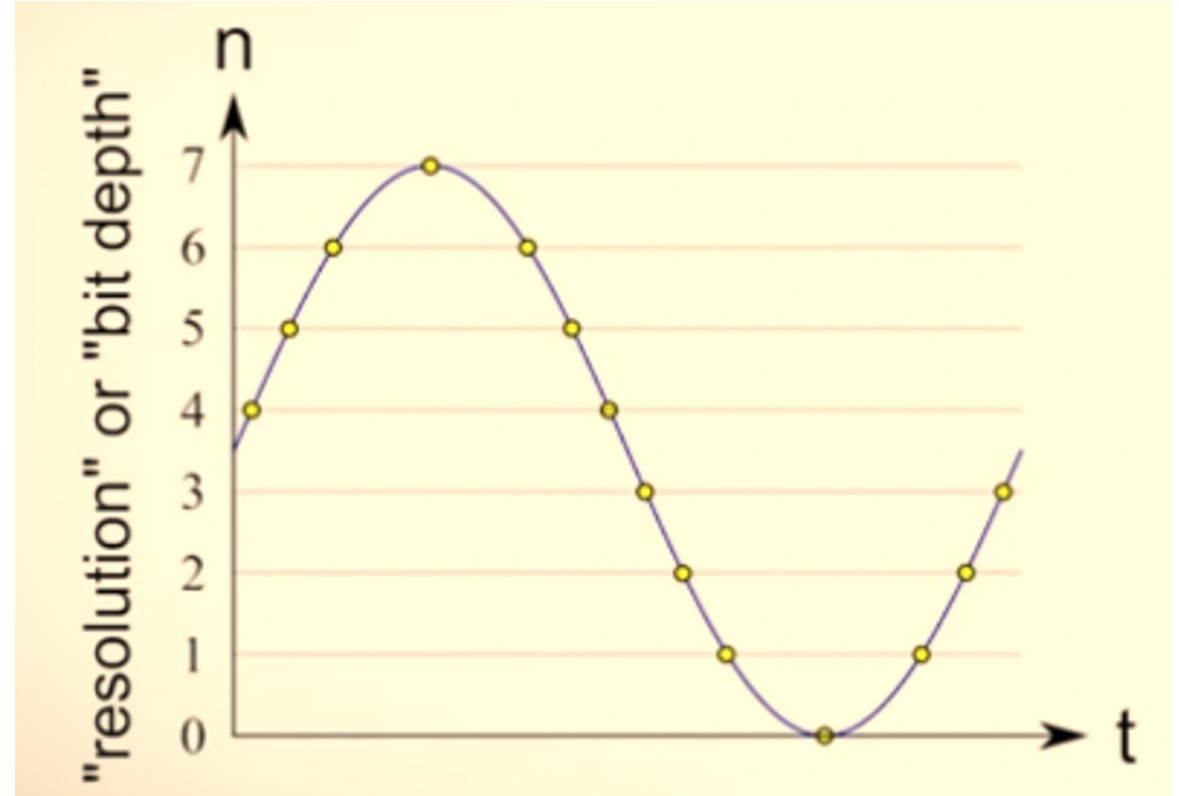
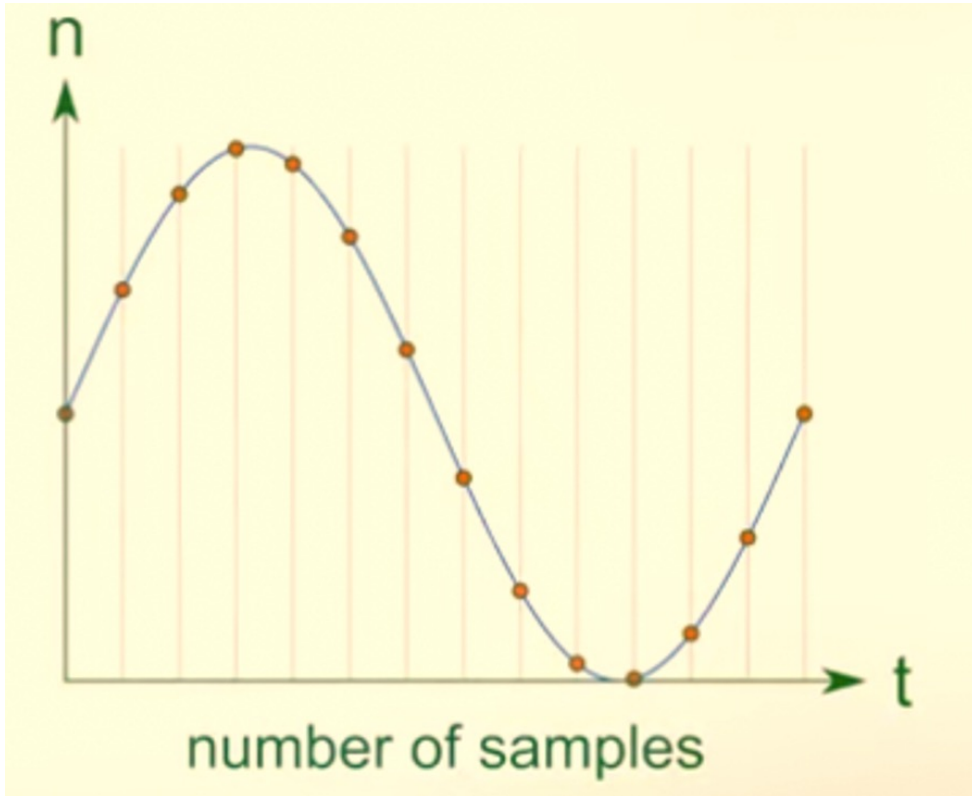
reconstruction OK



aliasing

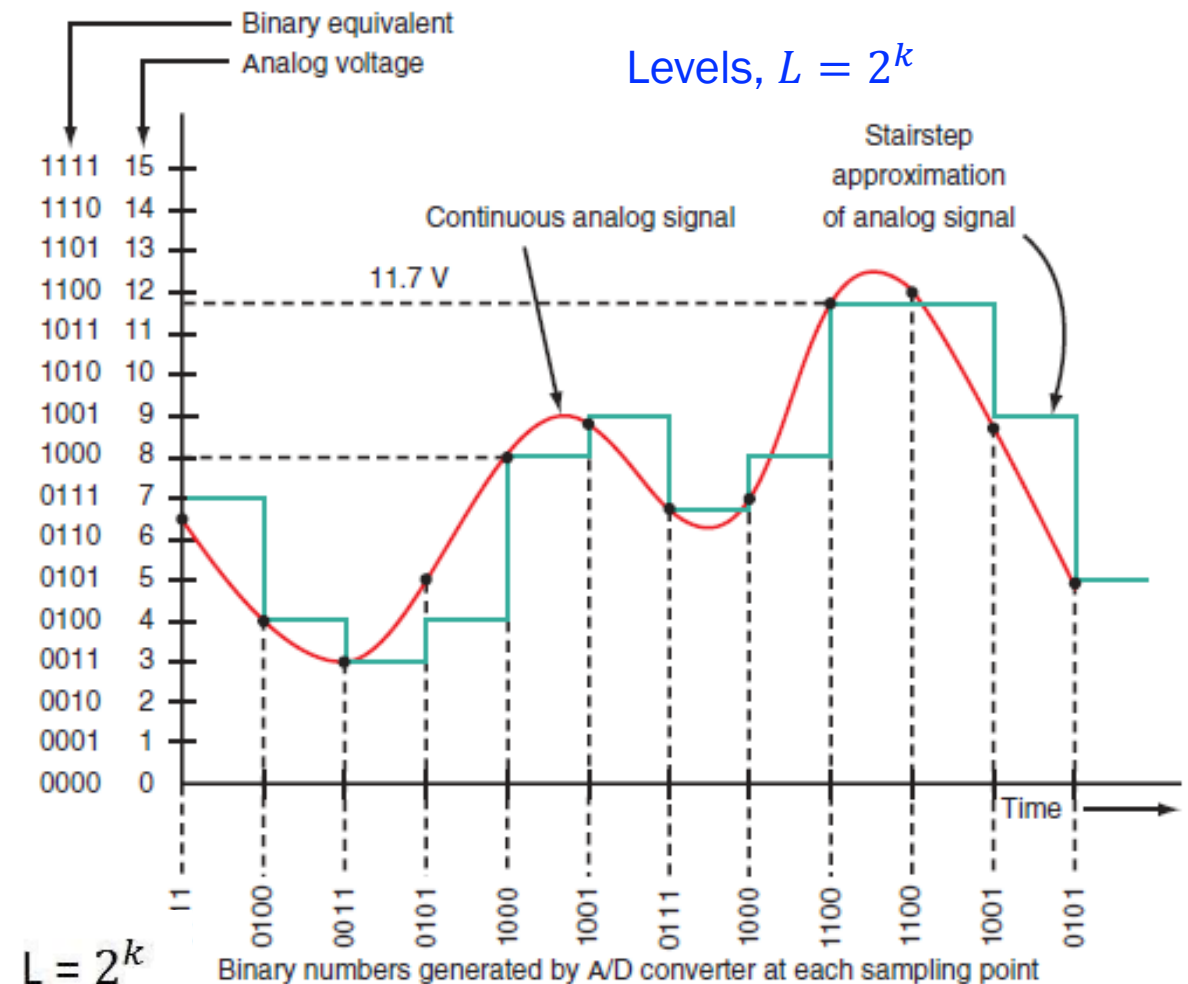


# Sampling Rate vs Sampling Depth



# Quantization and Encoding.

- Quantization is the process of converting an infinite number of possibilities to a finite number of levels.
- Analog signal is smooth and continuous, it represents infinite number of actual voltage levels and practically it is not possible to convert all analog samples to a precise proportional binary number.
- Figure shows the voltage range of 0-15V so there are total 16 levels. Here if analog input is 8V, its binary equivalent is 1000. But if analog input is 11.7 as shown in figure, then the approximate value i.e. 12 V will produce binary value 1100.
- Coding is the process by which a quantized value is converted to binary notation



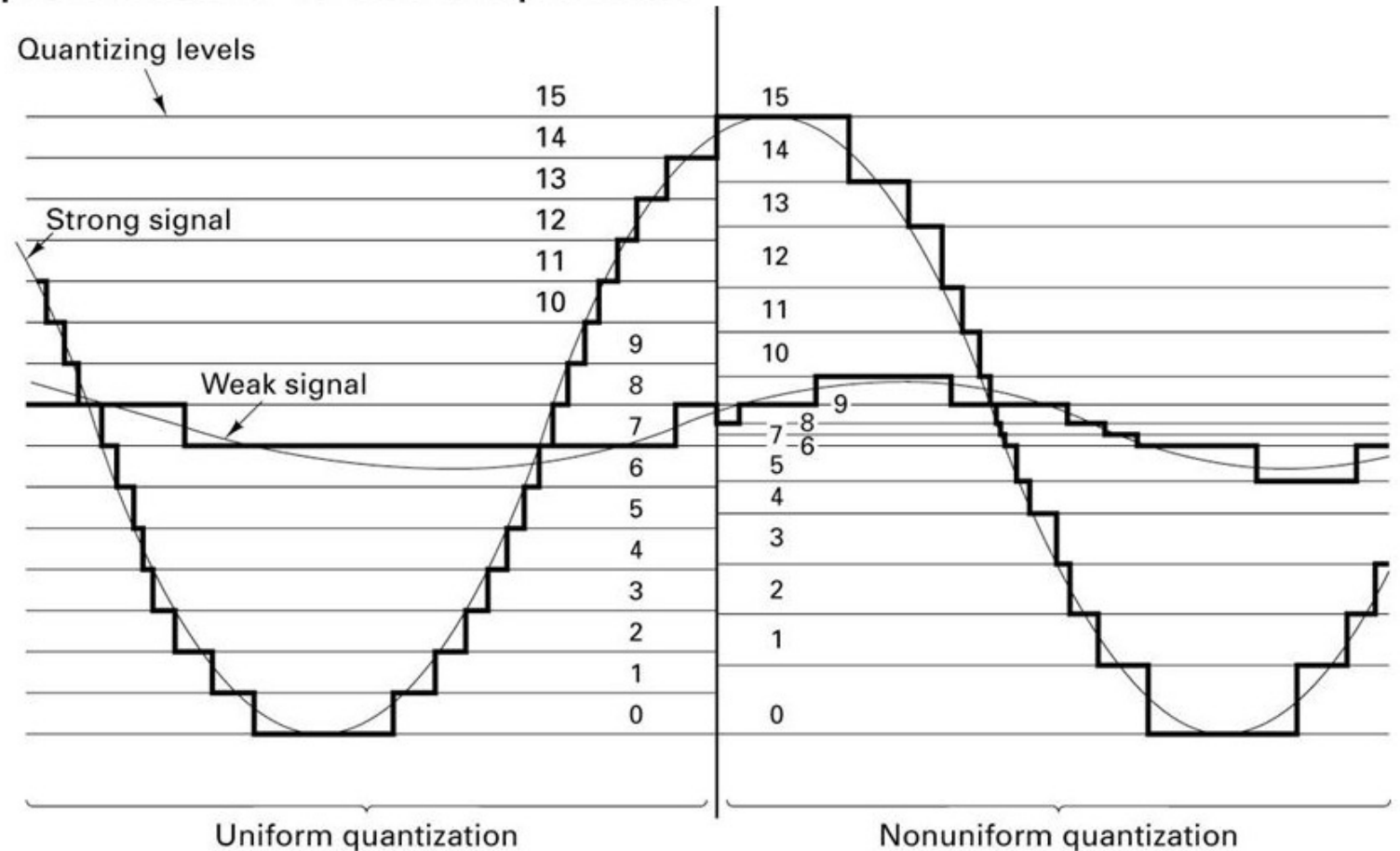


# TYPES OF QUANTIZATION

## ■ Uniform Quantization

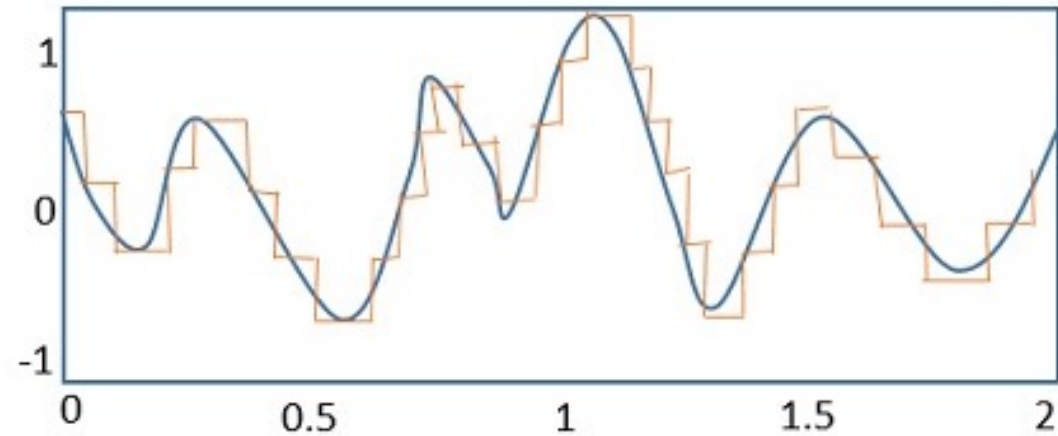
## ■ Non-uniform Quantization

Using a uniform quantizer for speech signals provides coarse quantization at low amplitudes



# Quantization Error and Noise

- The difference between an input value and its quantized value is called a **Quantization Error**.
- It is a type of quantization error, which usually occurs in analog audio signal, while quantizing it to digital. For example, in music, the signals keep changing continuously, where a regularity is not found in errors. Such errors create a wideband noise called as **Quantization Noise**.



Original  
and  
Quantized  
Signal

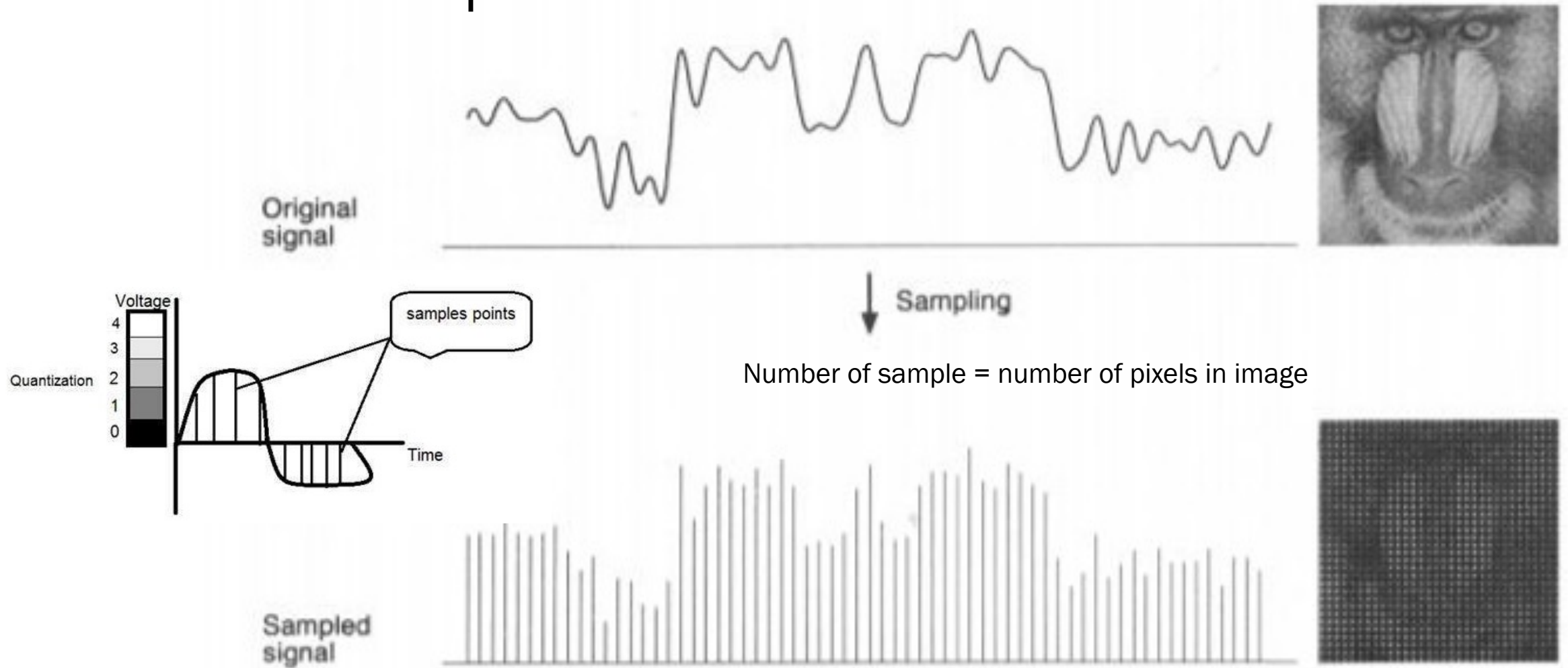


Quantization  
Error

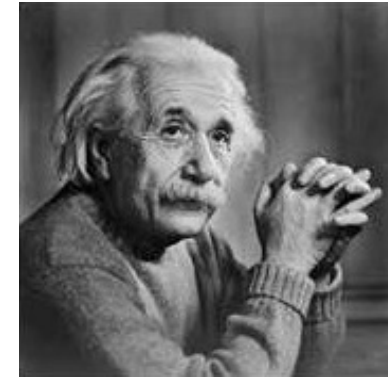
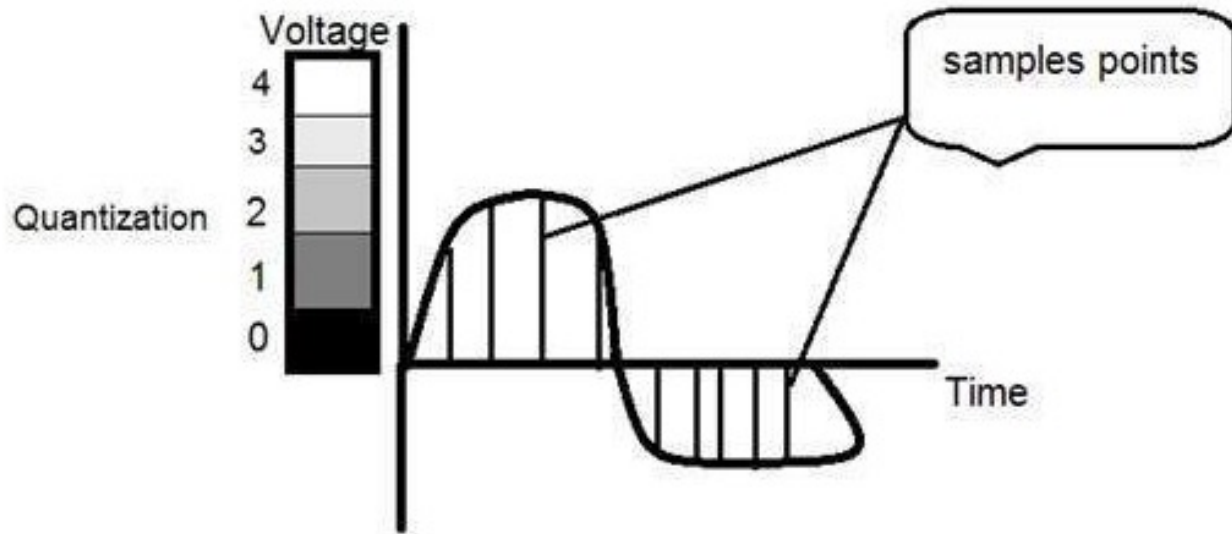
# Sampling Vs Quantization

Sampling	Quantization
Digitization of co-ordinate values.	Digitization of amplitude values.
X – axis (time) – discretized.	X – axis (time) – continuous.
Y – axis (amplitude) – continuous.	Y – axis (amplitude) – discretized.
Sampling is done prior to the quantization process.	Quantization is done after the sampling process.
It determines the spatial resolution of the digitized images.	It determines the number of grey levels in the digitized images.
A single amplitude value is selected from different values of the time interval to represent it.	Values representing the time intervals are rounded off to create a defined set of possible amplitude values.

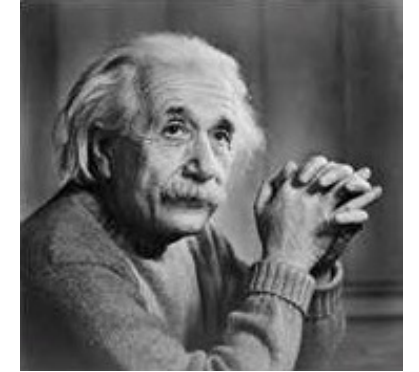
# Real world example



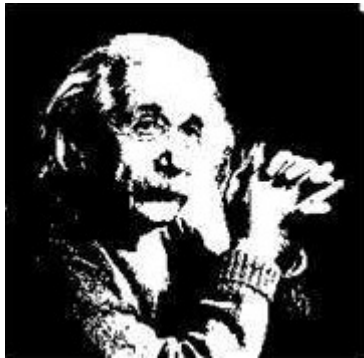
# Real world example



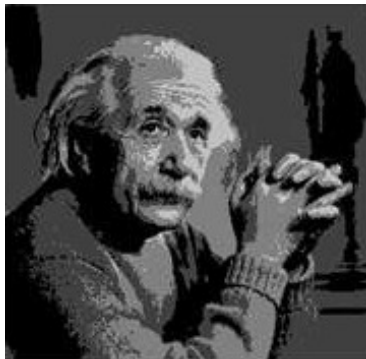
7bpp – 128 Gray levels



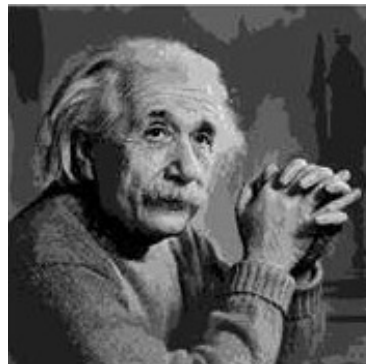
8bpp – 256 Gray levels



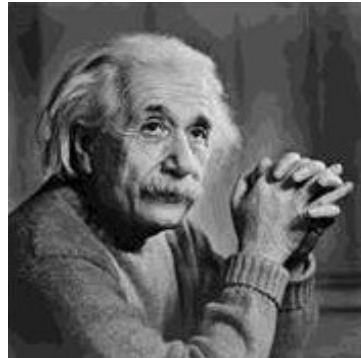
1bpp – 2 Gray levels



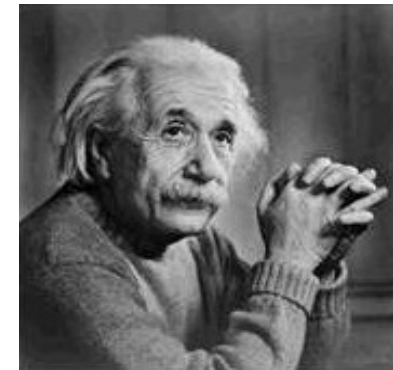
2bpp – 4 Gray levels



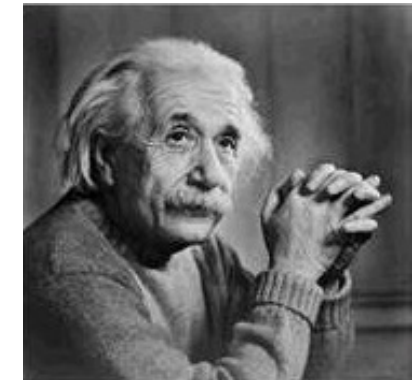
3bpp – 8 Gray levels



4bpp – 16 Gray levels

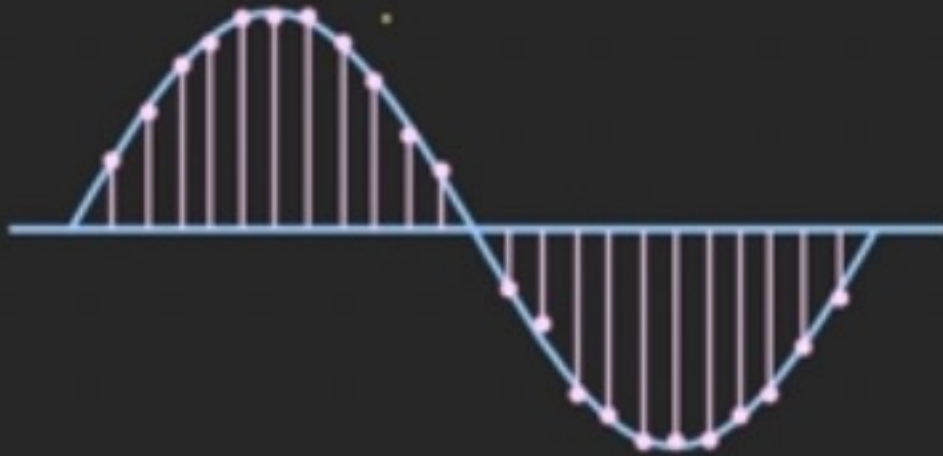


5bpp – 32 Gray levels

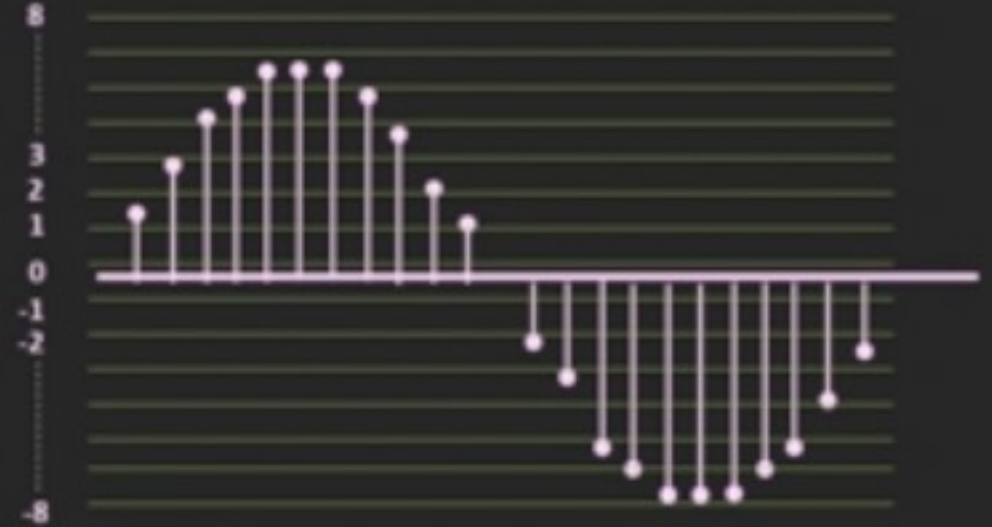


6bpp – 64 Gray levels

# Summary of Pulse Code Modulation



**Sampling**



**Quantization**



**Encoding**

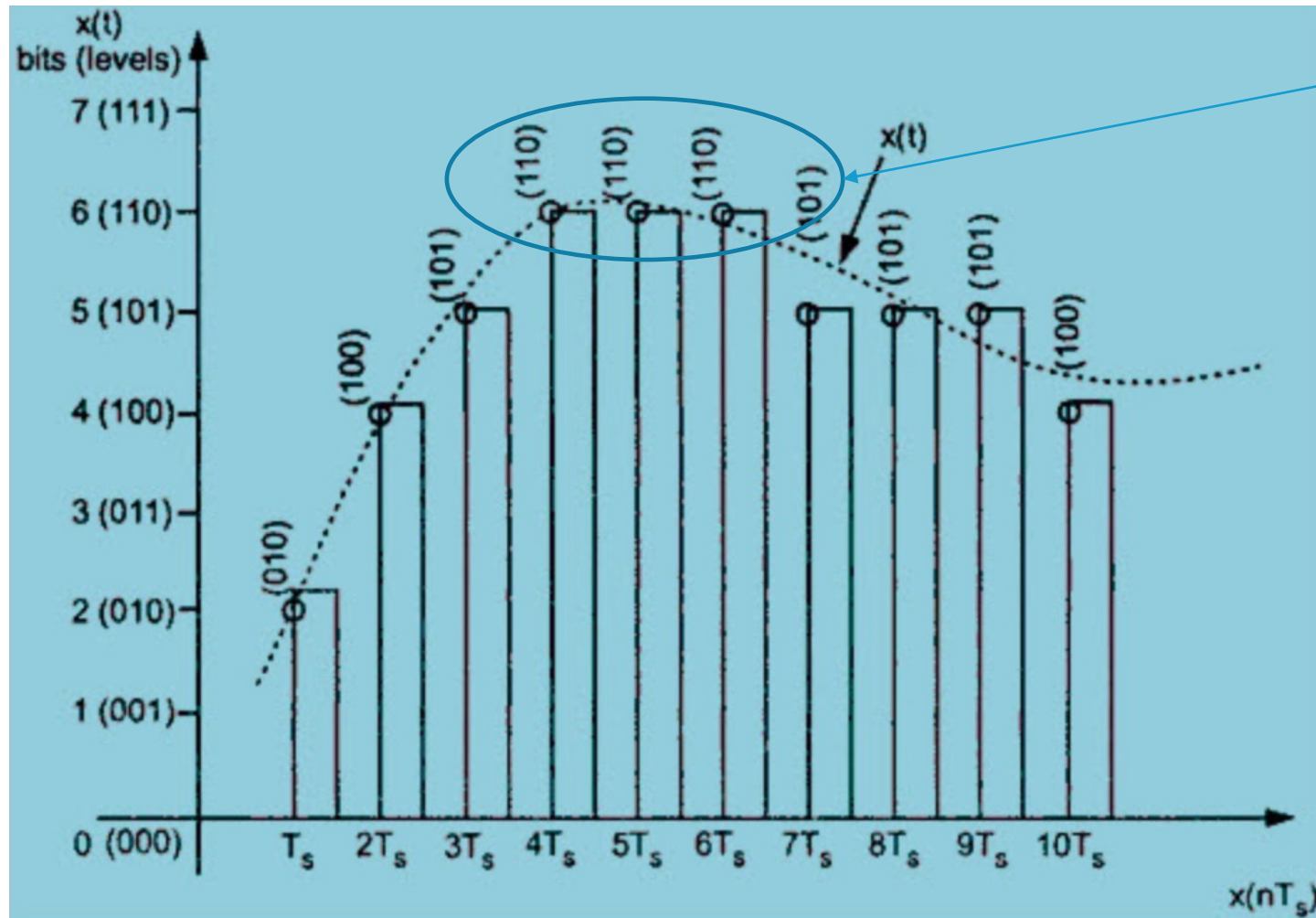


# DIFFERENTIAL PULSE CODE MODULATION (DPCM)

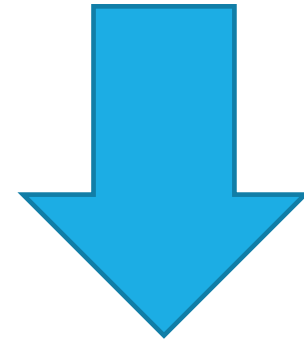




# Issues and inefficiencies of PCM



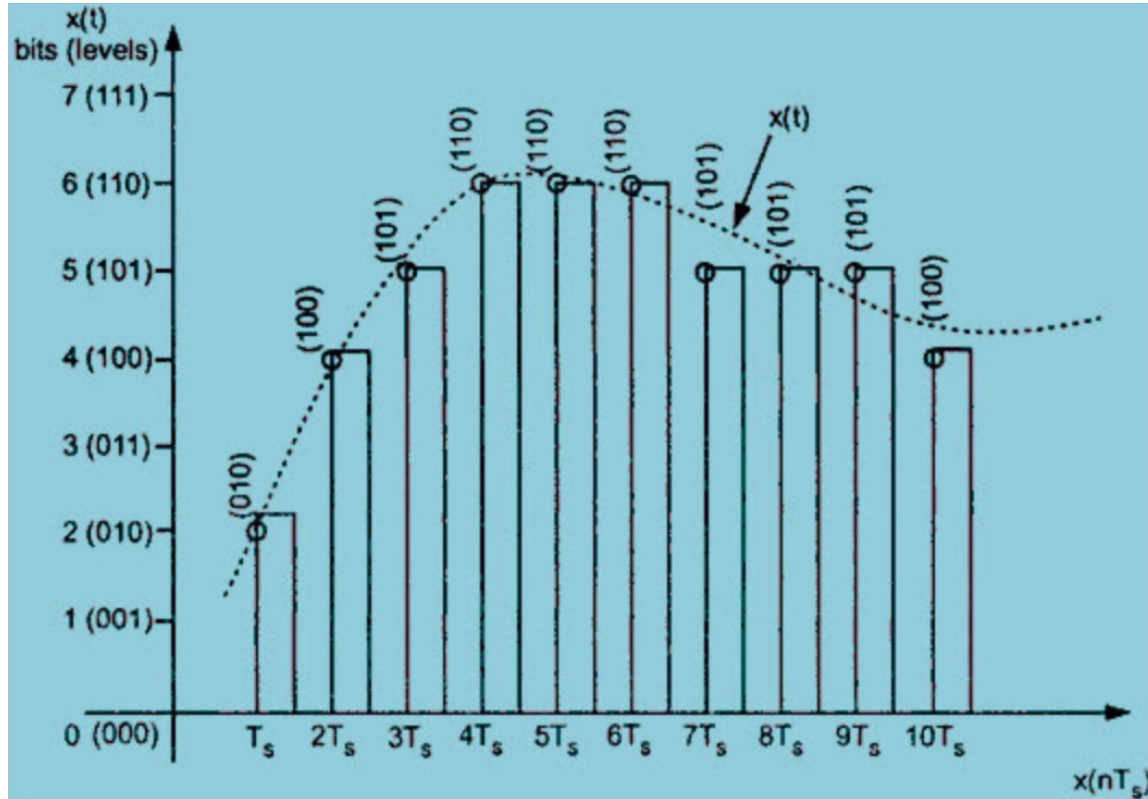
Illustrate the redundant information



Need to reduce sending redundant information in order to minimize BW usage



# Differential PCM (DPCM) technique



- Samples are encoded by using 3-bit (7 levels) PCM
- Samples are quantized to the nearest digital level

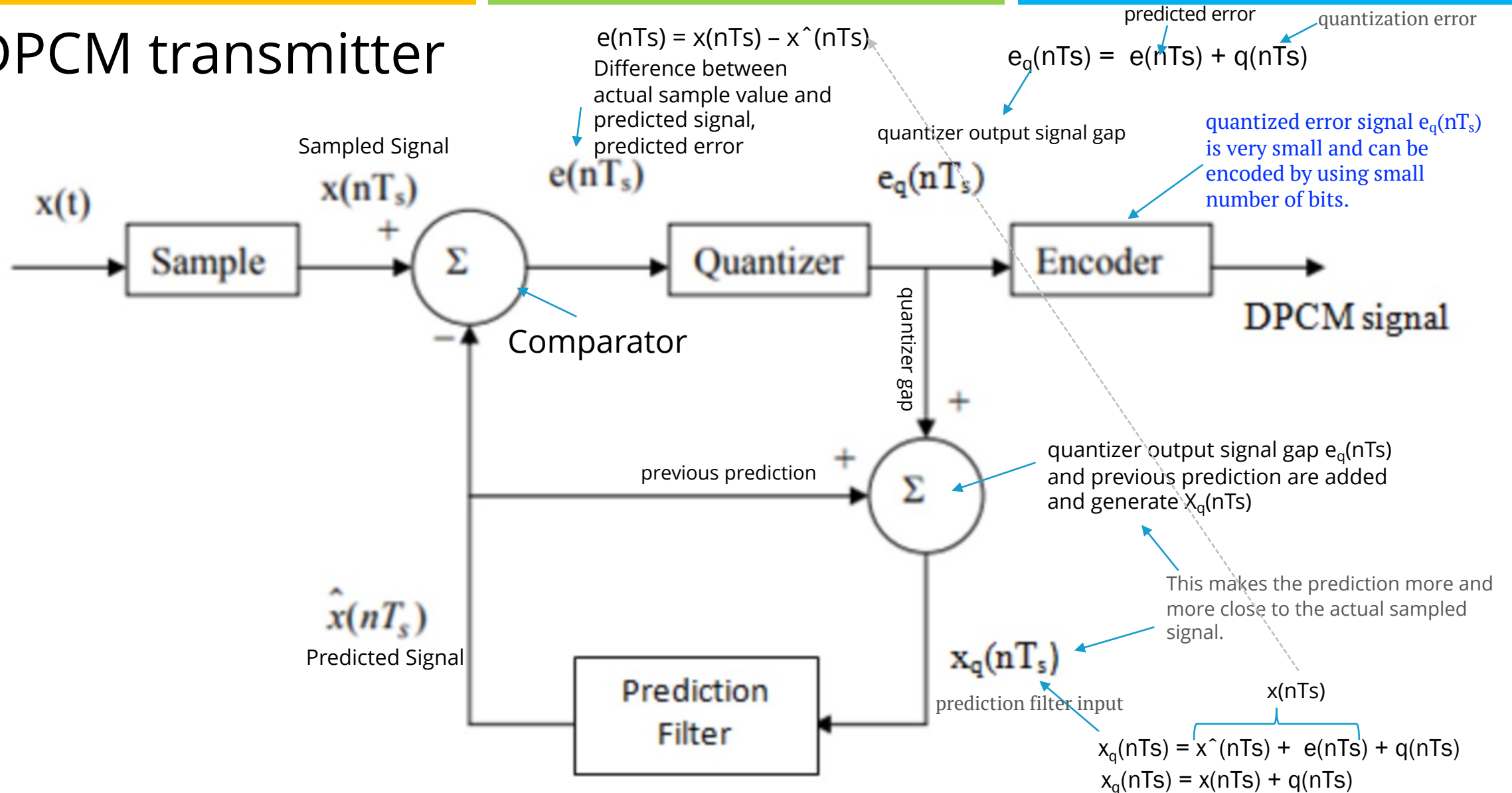
- At points  $4T_s$ ,  $5T_s$ , and  $6T_s$ , three samples are carrying the same information means redundant
- At  $9T_s$  and  $10T_s$ , the difference only due to the last bit and first two bits are same
- To avoid this redundant information sending an intelligent decision can be take to the next sampled value based on its previous quantised value.
- Which is called a Differential PCM (DPCM) technique.

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# DPCM

- Differential pulse code modulation (digital) works on the principle of prediction (The value of the present sample is predicted from the previous samples)
  - If the redundancy is reduced, then the overall bitrate will decrease, and the number of bits required to transmit one sample will also reduce.
-

# DPCM transmitter



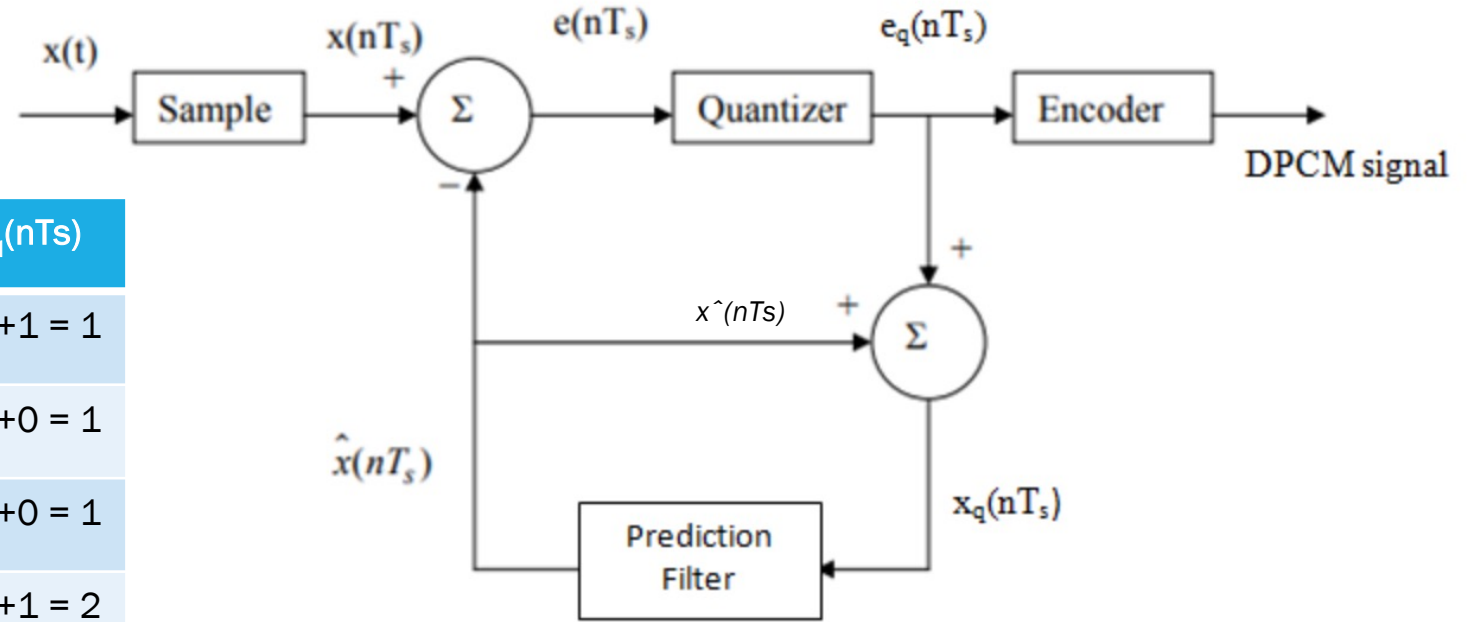
# DPCM transmitter

$x(nT_s)$	$\hat{x}(nT_s)$	$e(nT_s)$	$e_q(nT_s)$	$x_q(nT_s)$
1.1	0	1.1	1	$0+1 = 1$
1.2	1	0.2	0	$1+0 = 1$
1.3	1	0.3	0	$1+0 = 1$
1.6	1	0.6	1	$1+1 = 2$
1.7	2	0.3	0	$2+0 = 2$

$$x_q(nT_s) = \hat{x}(nT_s) + e_q(nT_s)$$

Transmitted signal,  $e_q(nT_s) \rightarrow 10010$

Encoded Txn DPCM Signal  $\rightarrow 001, 000, 000, 001, 000$



$$e(nT_s) = x(nT_s) - \hat{x}(nT_s)$$

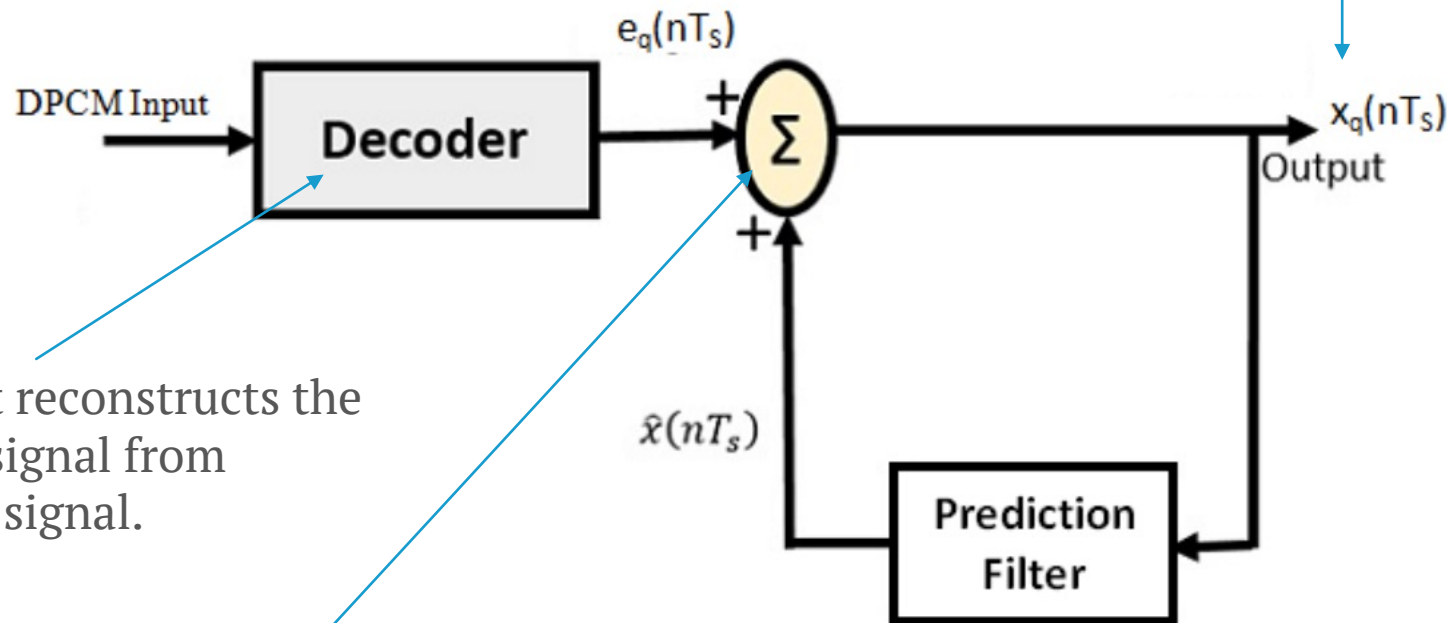
$$e_q(nT_s) = e(nT_s) + q(nT_s)$$

$$x_q(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s)$$

$$x_q(nT_s) = x(nT_s) + q(nT_s)$$

# DPCM Receiver

Thus, the signal at the receiver differs from actual signal by quantization error  $q(nT_s)$ , which is introduced permanently in the reconstructed signal.



The decoder first reconstructs the quantized error signal from incoming binary signal.

The prediction filter output and quantized error signals are summed up to give the quantized version of the original signal.

$e_q(nT_s)$	$\hat{x}(nT_s)$	$x_q(nT_s)$
1	0	$0+1 = 1$
0	1	$1+0 = 1$
0	1	$1+0 = 1$
1	1	$1+1 = 2$
0	2	$2+0 = 2$

# Advantages of DPCM

- As the difference between  $x(nT_s)$  and  $\hat{x}(nT_s)$  is being encoded and transmitted by the DPCM technique, a small difference voltage is to be quantized and encoded.
- This will require less number of quantization levels and hence less number of bits to represent them.
- Thus signalling rate and bandwidth of a DPCM system will be less than that of PCM.

## Applications

- DPCM is mainly used for Speech, Image and audio signal compressions
- In Images, there is a correlation between the neighboring pixels,
- in video signals, the correlation is between the same pixels in consecutive frames and inside frames.
- Further This methods is suitable for real time applications



# DELTA MODULATION

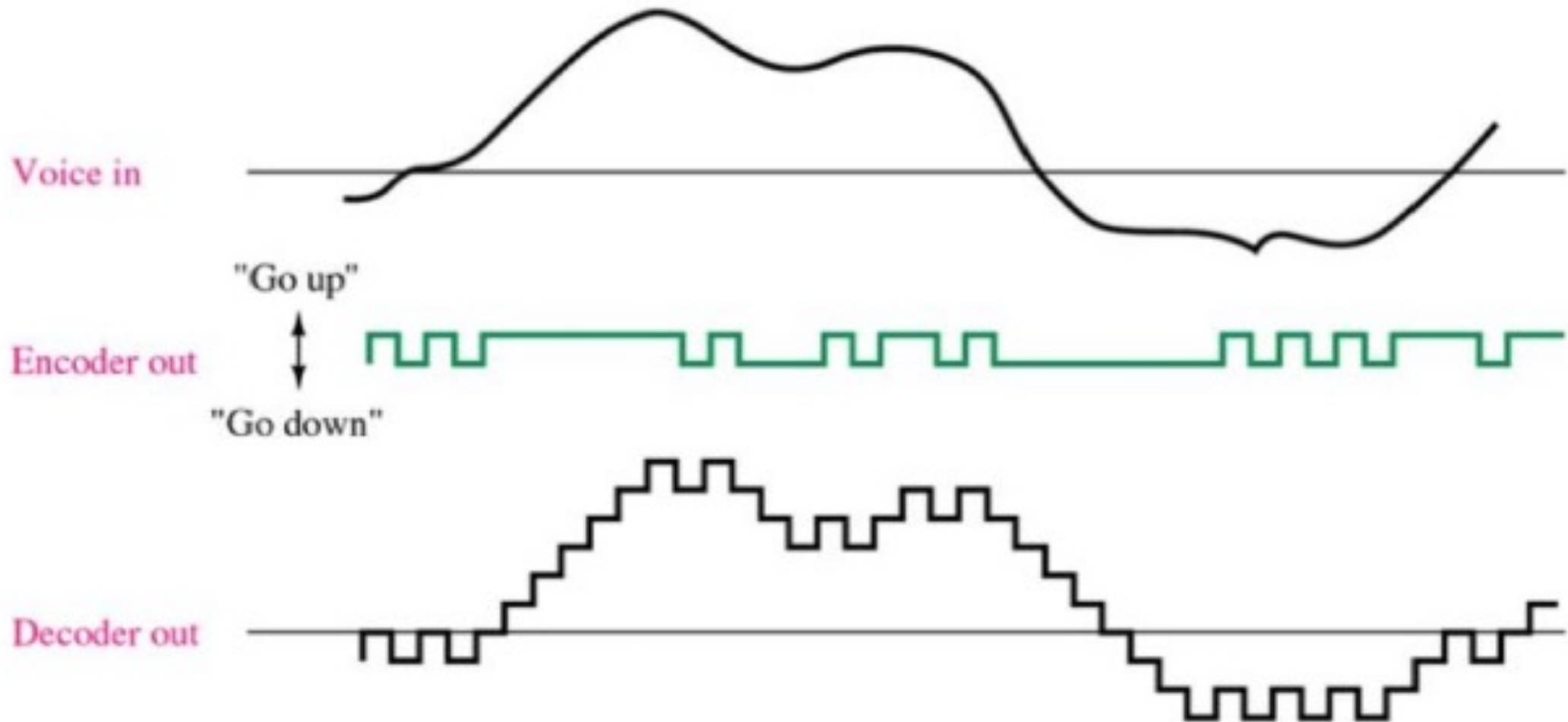


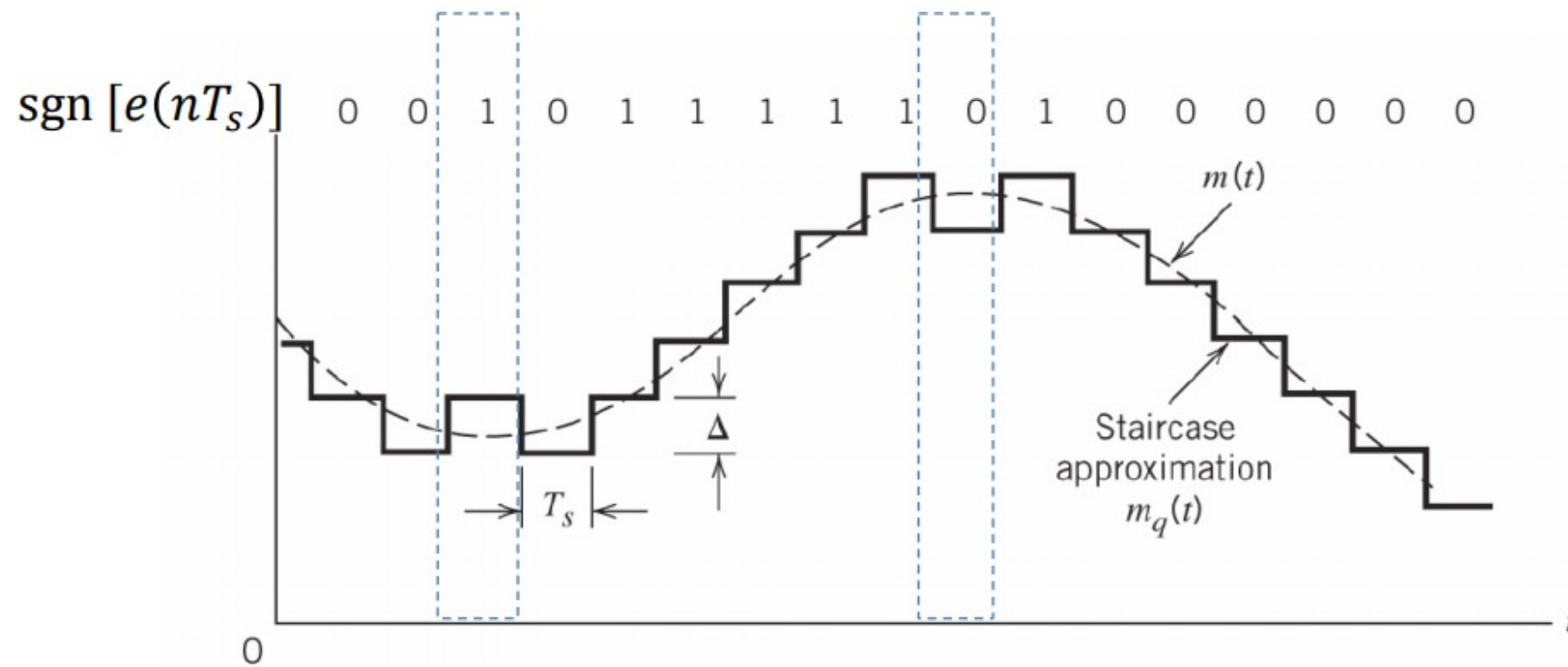
# Delta Modulation

- A delta modulation (DM or  $\Delta$ -modulation) is an analog-to-digital and digital-to-analog signal conversion technique used for transmission of voice information where quality is not of primary importance.
- Delta modulation transmits only one bit per sample. Here, the present sample value is compared with the previous sample value and this result whether the amplitude is increased or decreased is transmitted.
- Input signal  $x(t)$  is approximated to step signal by the delta modulator. This step size is kept fixed.
- The difference between the input signal  $x(t)$  and staircase approximated signal is confined to two levels, i.e.,  $+\Delta$  and  $-\Delta$ .
- Now, if the difference is positive, then approximated signal is increased by one step, i.e., ' $\Delta$ '. If the difference is negative, then approximated signal is reduced by ' $\Delta$ '.
- When the step is reduced, '0' is transmitted and if the step is increased, '1' is transmitted. Hence, for each sample, only one binary bit is transmitted.



# Delta Modulation



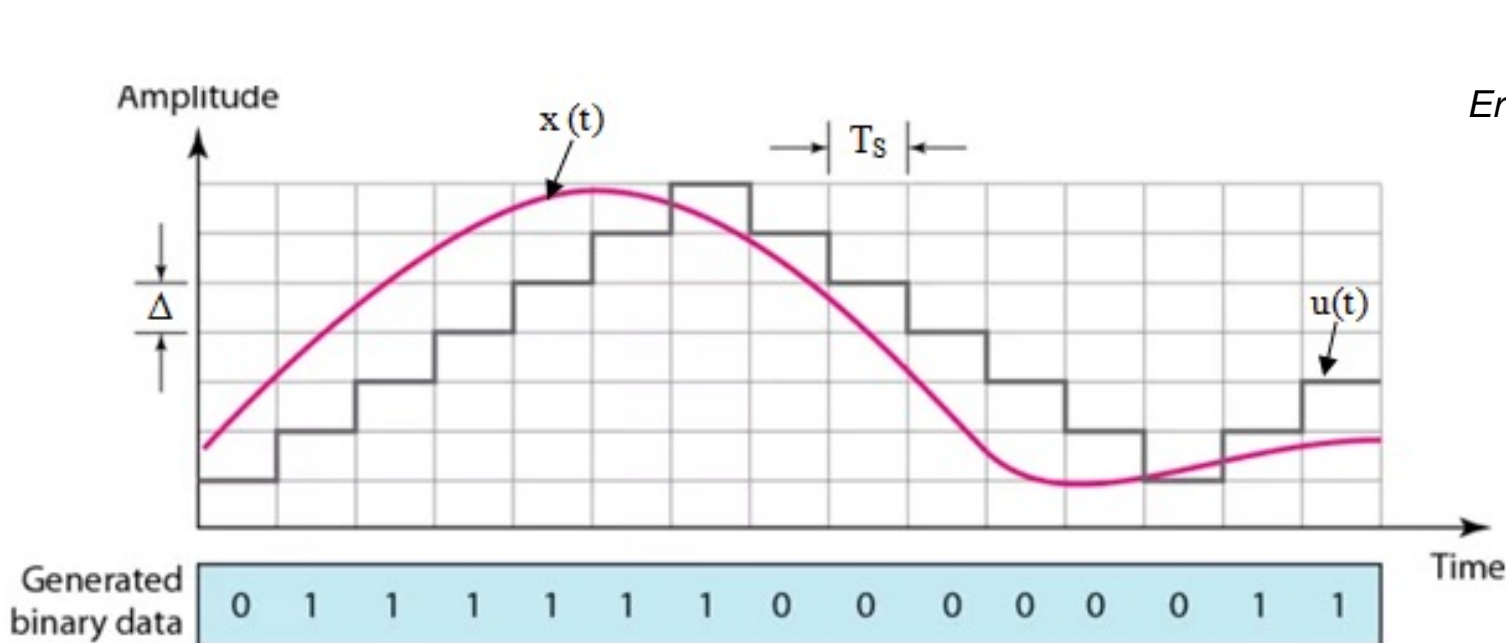


$$e(nT_s) = m(nT_s) - m_q(nT_s - T_s)$$

$$e_q(nT_s) = \Delta \text{sgn}[e(nT_s)]$$

$$m_q(nT_s) = m_q(nT_s - T_s) + e_q(nT_s)$$

# Delta Modulation Waveform



$T_s$  = sampling interval.

Error at present sample  
 $e(nT_s) = x(nT_s) - \hat{x}(nT_s)$

Sample signal of  $x(t)$   
 $b(nT_s) = \begin{cases} +\Delta & \text{if } x(nT_s) \geq \hat{x}(nT_s) \\ -\Delta & \text{if } x(nT_s) < \hat{x}(nT_s) \end{cases}$

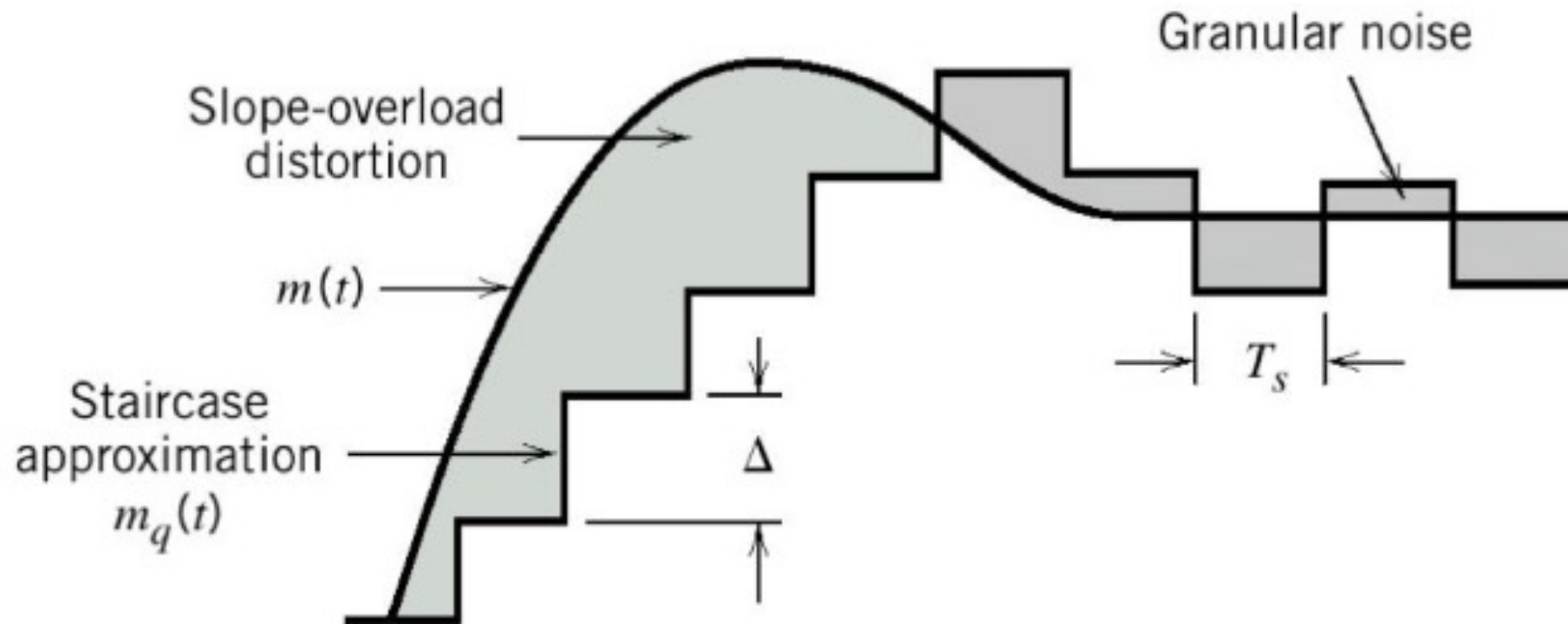
Last sample Approx. of staircase  
 $\hat{x}(nT_s)$

Where

$$b(nT_s) = \Delta \operatorname{sgn}[e(nT_s)]$$

If  $b(nT_s) = +\Delta$  then a binary '1' is transmitted and  
 if  $b(nT_s) = -\Delta$  then a binary '0' is transmitted

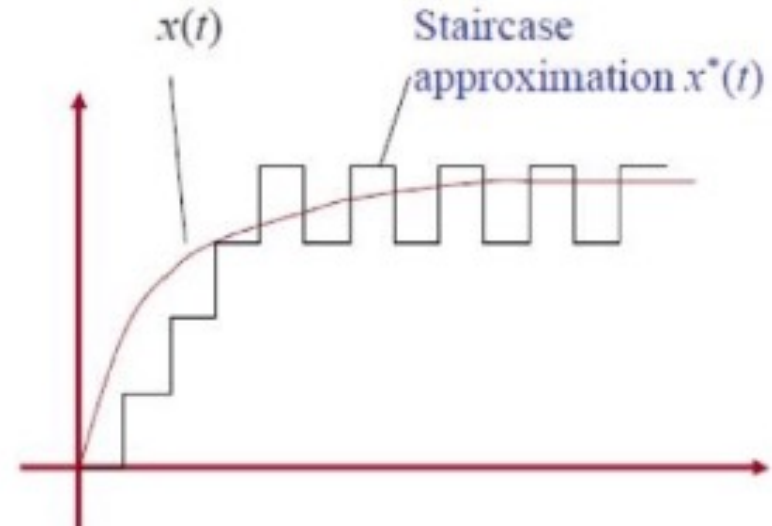
# Noise In Delta modulation



# Slope-overload distortion Vs Granular Noise



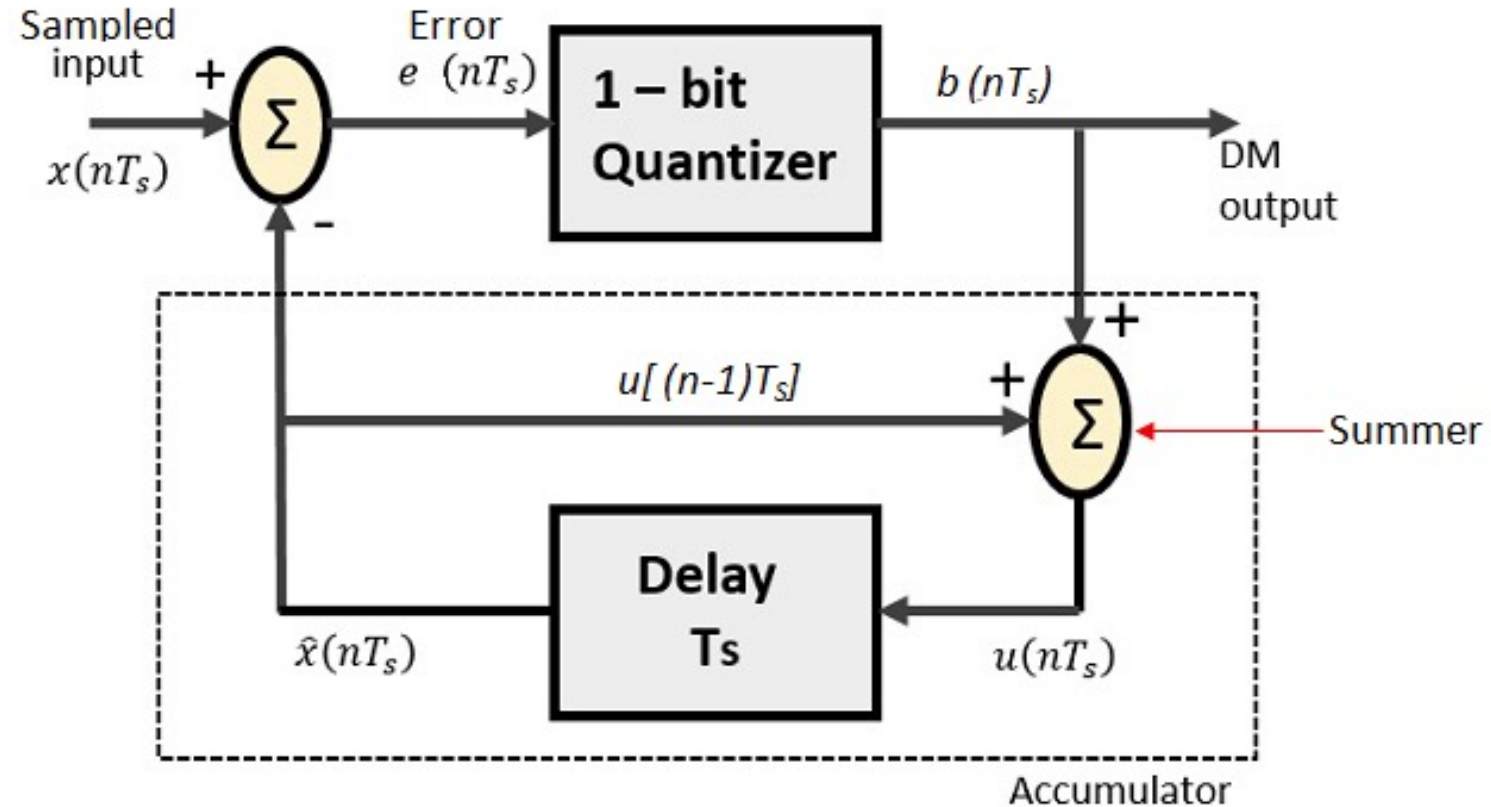
- Slope overload distortion occurs when  $\Delta$  is too small, the staircase approximation can't follow closely to the actual curve of message signal
- Large  $\Delta$  is needed for rapid variation of staircase signal to reduce the slope distortion



- Granular noise occurs when  $\Delta$  is too large relative to the local slope characteristics of signal
- Granular noise is smaller to quantization noise in PCM
- Small  $\Delta$  is needed for slowly varying signal to reduce the granular noise

# Delta Transmitter

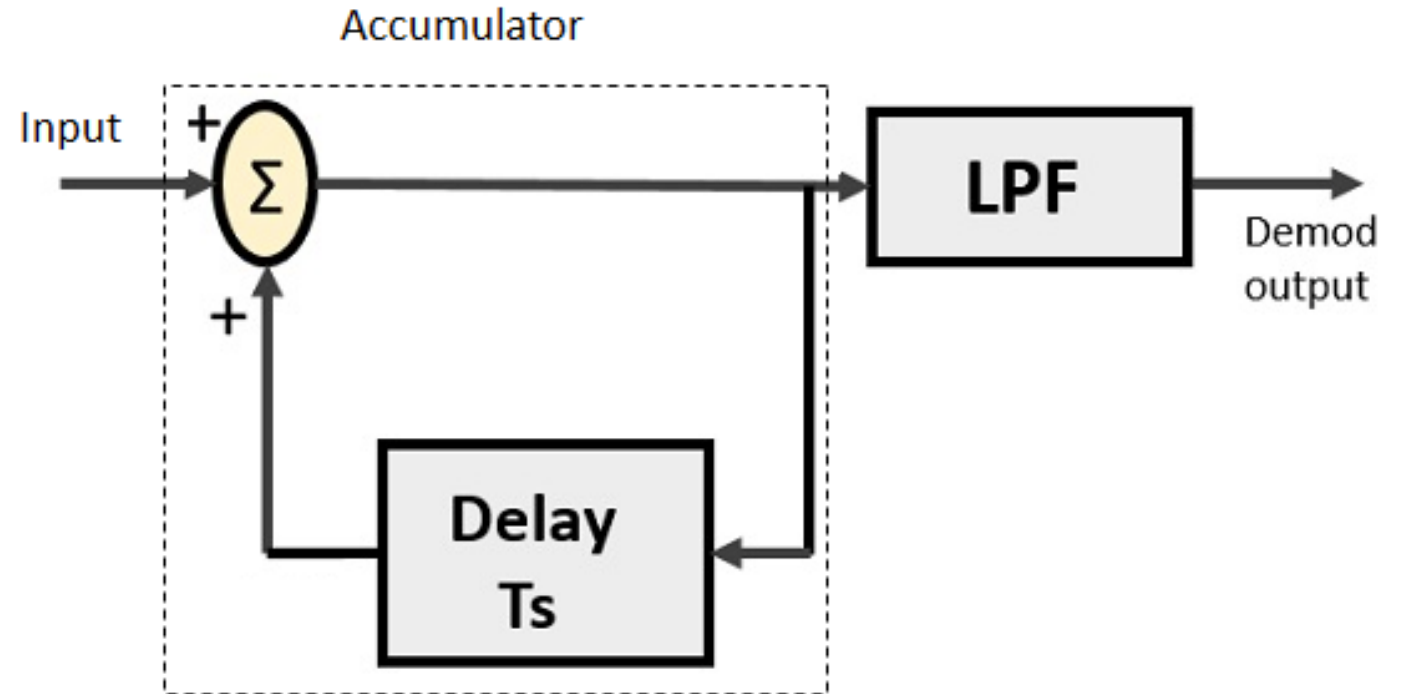
- It consists of a 1-bit quantizer and a delay circuit along with two summer circuits.
- The summer in the accumulator adds quantizer output ( $\pm\Delta$ ) with the previous sample approximation.
- The previous sample approximation  $u[(n-1)T_s]$  is restored by delaying one sample period  $T_s$ .
- The samples input signal  $x(nT_s)$  and staircase approximated signal  $\hat{x}(nT_s)$  are subtracted to get error signal  $e(nT_s)$ .
- Thus, depending on the sign of  $e(nT_s)$ , one bit quantizer generates an output of  $+\Delta$  or  $-\Delta$ .



If the step size is  $+\Delta$ , then binary '1' is transmitted and if it is  $-\Delta$ , then binary '0' is transmitted

# Delta Modulation Receiver

- The accumulator generates the staircase approximated signal output and is delayed by one sampling period  $T_s$ .
- It is then added to the input signal.
- If the input is binary '1' then it adds  $+\Delta$  step to the previous output (which is delayed).
- If the input is binary '0' then one step ' $\Delta$ ' is subtracted from the delayed signal.
- Also, the low pass filter smoothens the staircase signal to reconstruct the original message signal  $x(t)$ .



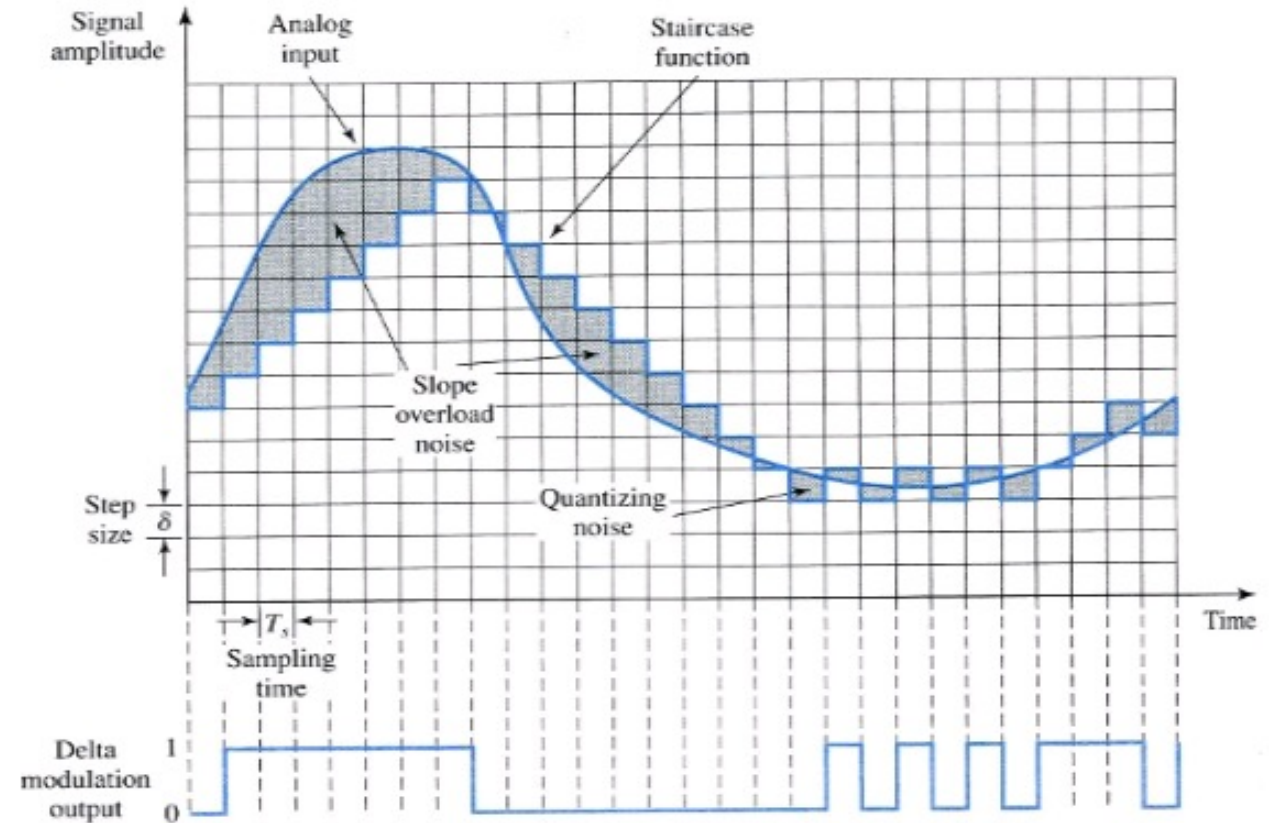
# Advantages and Disadvantages of DM

## Advantages

- Low signalling rate and low transmission channel bandwidth because in DM only one bit is transmitted per sample
- The delta modulator transmitter and receiver are less complicated to implement as compared to the PCM

## Disadvantages

- The two distortions slope-overhead error and granular noise are present.
- Practically the signalling rate with no slope-overhead error will be much higher than that of PCM.





# PCM vs DM

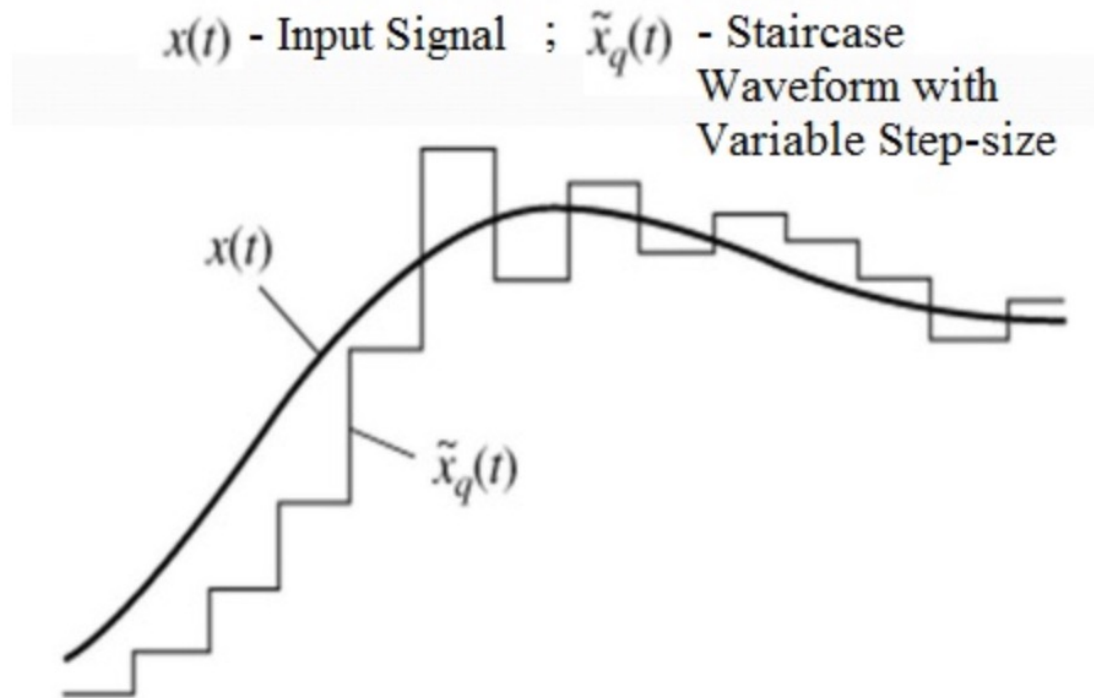
PCM	DM
In PCM, feedback does not exist in transmitter or receiver.	While in DM, feedback exists in transmitter.
Per sample 4, 8, or 16 bits are used.	Here, only one bit is used per sample.
PCM requires highest transmitter bandwidth.	DM requires lowest transmitter bandwidth.
PCM is complex in terms of complexity of implementation.	While DM is simple in terms of complexity of implementation.
PCM has good signal to noise ratio.	While DM has poor signal to noise ratio.
PCM is costly.	DM is cheap.
PCM may be a technique wont to digitally represent sampled analog signals.	Digital to analog and analog to digital converter.
In PCM, signal requires encoder and decoder both sides.	In DM, signal can modulate and demodulate.
PM is mostly used in video telephony and audio telephony.	DM is mostly used in speeches as well as images.



# ADAPTIVE DELTA MODULATION

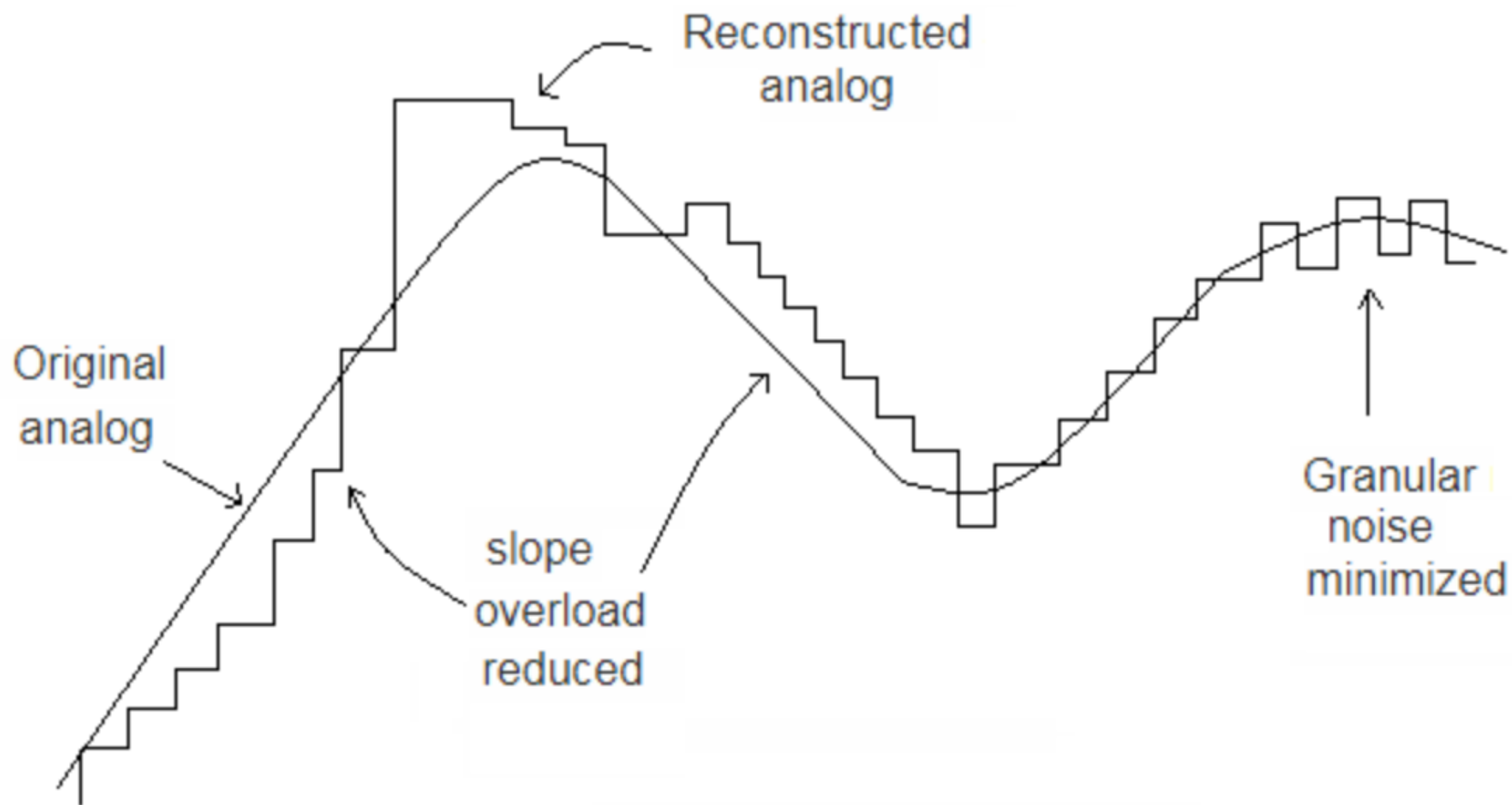


# Adaptive Delta Modulation



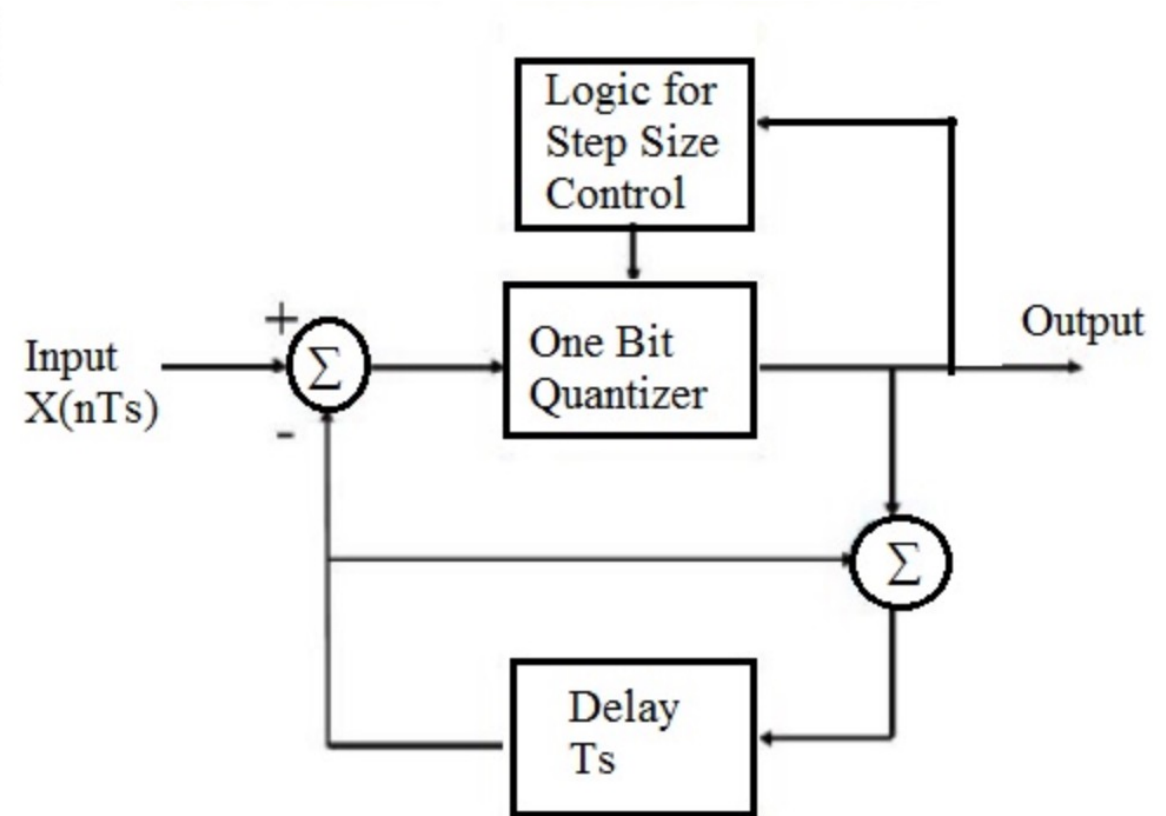
- This Modulation is the refined form of delta modulation.
- This method was introduced to solve the granular noise and slope-overload error caused during Delta modulation.
- This Modulation method is similar to Delta modulation except that the step size is variable according to the input signal in Adaptive Delta Modulation whereas it is a fixed value in delta modulation.

# ADM example



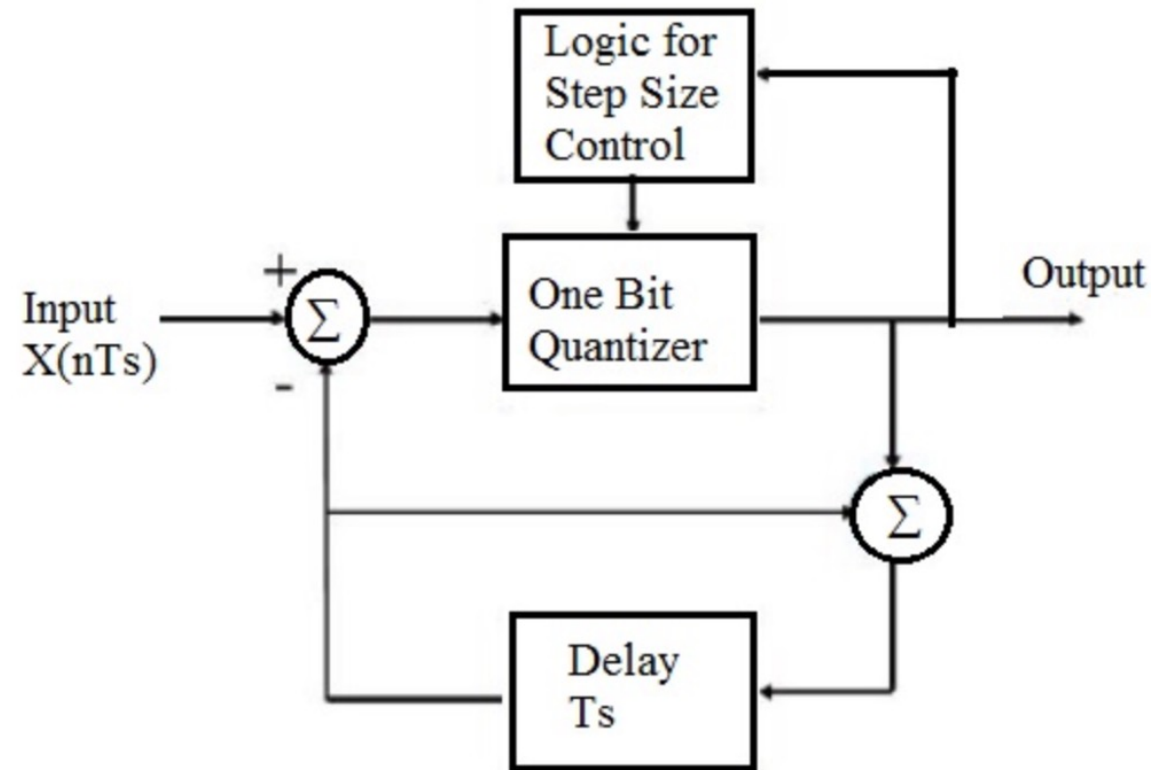
# Adaptive Delta Modulation Transmitter

- The transmitter circuit consists of a summer, quantizer, Delay circuit, and a logic circuit for step size control.
- The baseband signal  $X(nT_s)$  is given as input to the circuit.
- The feedback circuit present in the transmitter is an Integrator.
- The integrator generates the staircase approximation of the previous sample.

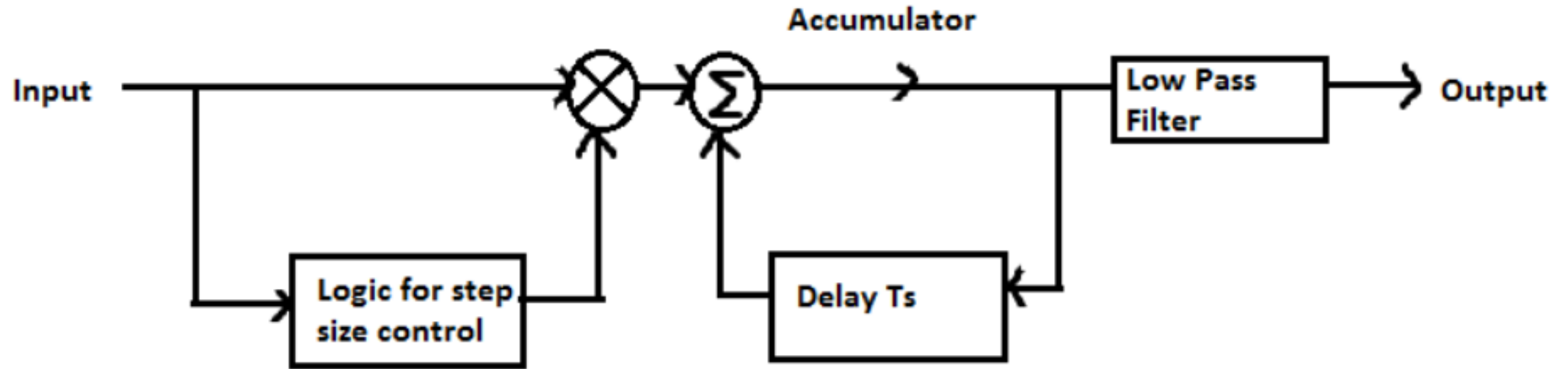


# ADM Transmitter Cont.

- At the summer circuit, the difference between the present sample and staircase approximation of previous sample  $e(nT_s)$  is calculated.
- This error signal is passed to the quantizer, where a quantized value is generated.
- The step size control block controls the step size of the next approximation based on either the quantized value is high or low.
- The quantized signal is given as output.



# ADM Receiver



- The receiver has two parts.
- First part is the step size control. Here the received signal is passed through a logic step size control block, where the step size is produced from each incoming bit. Step size is decided based on present and previous input.
- In the second part of the receiver, the accumulator circuit recreates the staircase signal. This waveform is then applied to a low pass filter which smoothens the waveform and recreates the original signal.

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# Advantages of ADM

- Adaptive delta modulation decreases slope error present in delta modulation.
  - During demodulation, it uses a low pass filter which removes the quantized noise.
  - The slope overload error and granular error present in delta modulation are solved using this modulation. Because of this, the signal to noise ratio of this modulation is better than delta modulation.
  - In the presence of bit errors, this modulation provides robust performance. This reduces the need for error detection and correction circuits in radio design.
  - The dynamic range of Adaptive delta modulation is large as the variable step size covers large range of values.
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# Differences between DM and ADM

- In Delta Modulation step size is fixed for the whole signal. Whereas in Adaptive delta modulation, the step size varies depending upon the input signal.
  - The slope overload and granular noise errors which are present in delta modulation are minimized in ADM.
  - The dynamic range of Adaptive delta modulation is wider than delta modulation.
  - This modulation utilizes bandwidth more effectively than delta modulation.
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# Applications of ADM

- This modulation is used for a system which requires improved wireless voice quality as well as speed transfer of bits.
  - In television signal transmission this modulation process is used.
  - This modulation method is used in voice coding.
  - This modulation is also used as a standard by NASA for all communications between mission control and spacecraft.
  - Motorola's SECURENET line of digital radio products uses 12kbits/sec Adaptive Delta Modulation.
  - To provide voice detection quality audio at deployed areas, military uses 16 to 32 kbit/sec modulation system in TRI-TAC digital telephones.
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# Applications of ADM

- US army forces use 16kbit/sec rates to conserve bandwidth over tactical links.
  - For improved voice quality US Air Forces uses 32kbits/sec rates.
  - In Bluetooth-services to encode voice signals, this modulation is used with 32bits/sec rates.
  - HC55516 decoder is used in various arcade games such as sinistar and smash tv and pinball machines such as gorgor or space shuttle, to play pre-recorded sounds.
  - Adaptive delta modulation is also known as continuously variable slope delta modulation.
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ANY QUESTION??

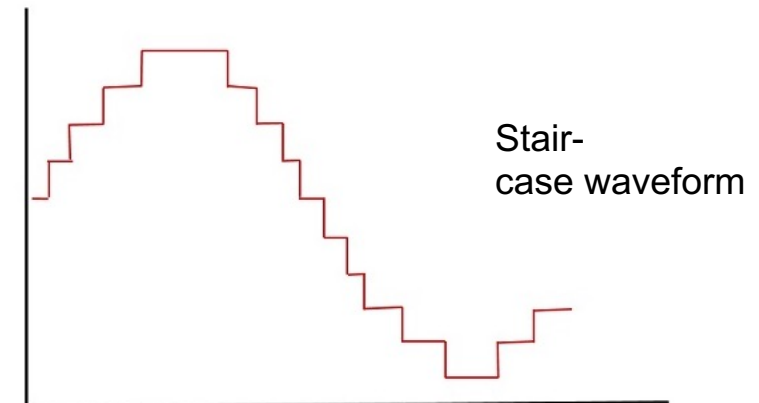
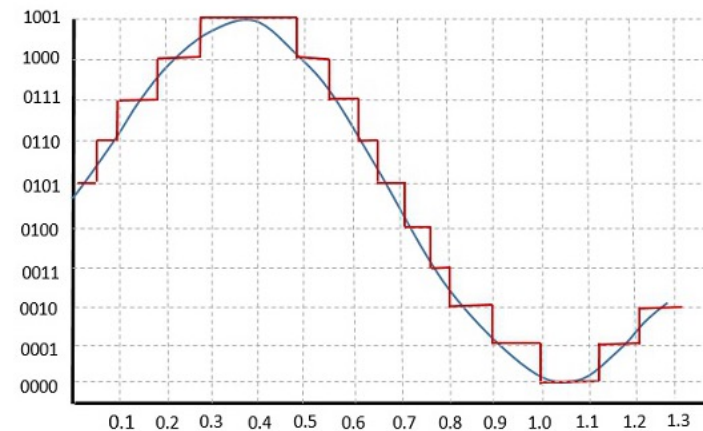
THANK YOU!!



# Quantization

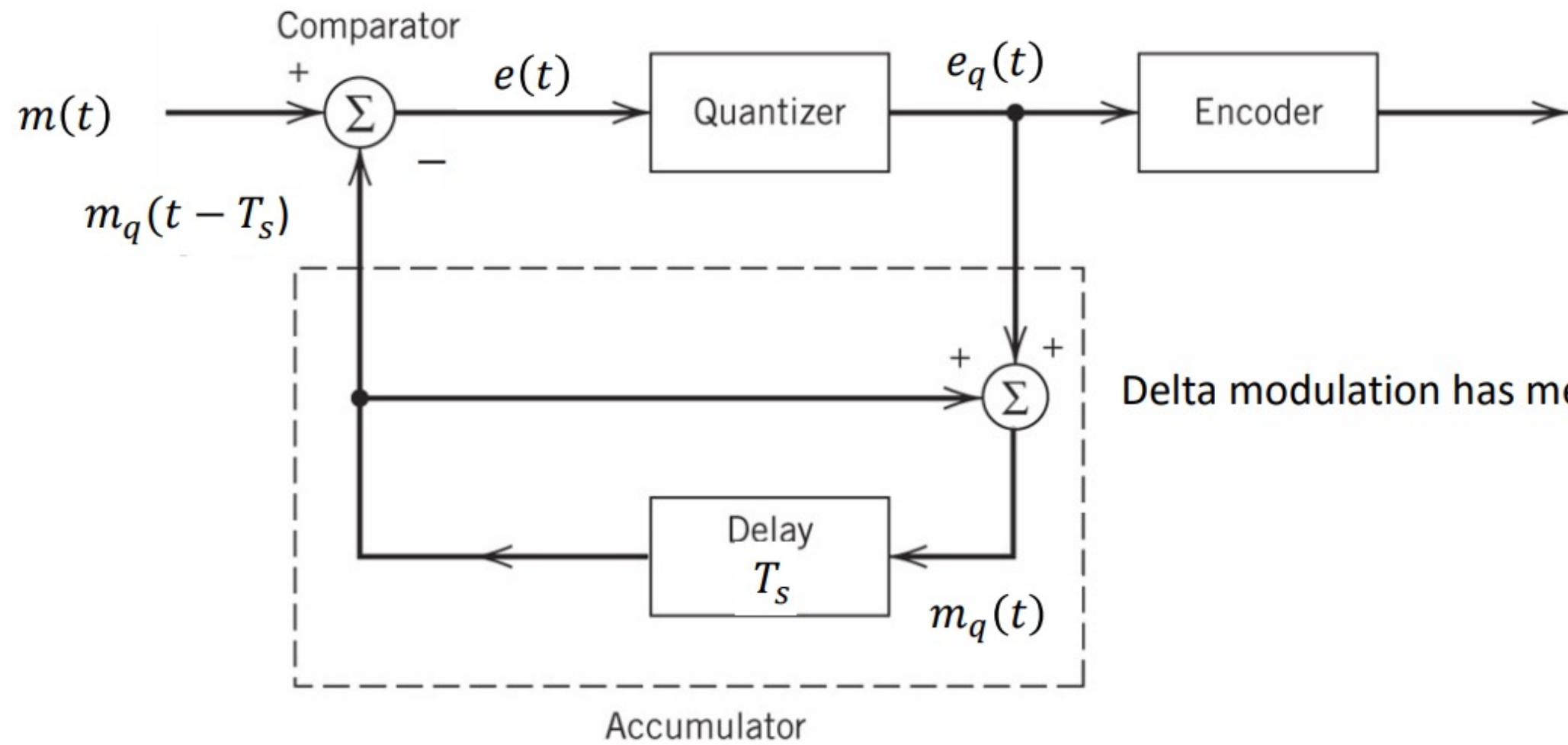
- **Quantization** is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.
- The quality of a Quantizer output depends upon the number of quantization levels used.
- The discrete amplitudes of the quantized output are called as **representation levels** or **reconstruction levels**.
- The spacing between the two adjacent representation levels is called a **quantum** or **step-size**.

Both sampling and quantization result in the loss of information.



# Delta Modulation

- A delta modulation (DM or  $\Delta$ -modulation) is an analog-to-digital and digital-to-analog signal conversion technique used for transmission of voice information where quality is not of primary importance.
- The difference between the input signal  $x(t)$  and staircase approximated signal is confined to two levels, i.e.,  $+\Delta$  and  $-\Delta$ .
- A single bit PCM code to achieve digital transmission of analog signal. Use only 1 bit either logic "1" or "0"
- Logic "0" is transmitted if current sample is smaller than the previous sample
- Logic "1" is transmitted if current sample is larger than the previous sample



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# Applications of Delta Modulations

- Delta modulation is employed to realize high signal to noise ratio.
  - This modulation is applied to ECG waveform for database reduction and real-time signal processing.
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