Digital Communications Systems

BY
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Course Objectives

The students to be able to understand, analyze, and design fundamental digital communication systems.

To know various coding techniques such as source coding, line coding, and channel coding.

To understand various digital modulation techniques and their applications.

The course focuses on developing a thorough understanding of digital Communication systems by using a series of specific examples and problems.

Course outcomes

After the successful completion of the course, the student will be able to

CO1: Understand the basic concepts of digital communications with clear perception into practical applications and the importance of analog system over digital system.

CO2: Analyze SNR for different modulation techniques like PCM, DM, ADM and DPCM with their generation and detection methods.

CO3: Describe and determine the performance of line codes and methods to mitigate inter symbol interference.

CO4: Understand the generation, detection signal space diagram, spectrum, bandwidth efficiency, and probability of error analysis of different band pass modulation techniques.

CO5: Compare and contrast ASK, FSK, PSK digital carrier modulation schemes in terms of occupied bandwidth, complexity etc., and extend these into QPSK, QAM for improved spectral efficiency.

CO6: Apply the basics of information theory to calculate channel capacity and interpret the differences between the usage of systematic linear block codes (LBC) and convolutional codes for systematic and non-systematic channels.

Textbook and References

Textbooks/References

Text Books:

1. *Simon Hakin, "Communication Systems," Wiley India Edition, 4th Edition, 2011.*

2. *A. Bruce Carlson, & Paul B. Crilly, "Communication Systems – An Introduction to Signals & Noise in Electrical Communication", McGraw-Hill International Edition, 5th Edition, 2010.*

Reference Books:

1. *Sam Shanmugam, "Digital and Analog Communication Systems", John Wiley, 2005.*

2. *B.P. Lathi, &Zhi Ding, "Modern Digital & Analog Communication Systems", Oxford University Press, International 4th edition, 2010.*

3. *Bernard Sklar, "Digital Communications", Prentice-Hall PTR, 2nd edition, 2001.*

4. *Herbert Taub& Donald L Schilling, "Principles of Communication Systems", Tata McGraw-Hill, 3rd Edition, 2009.*

5. *J. G. Proakis, M Salehi, Gerhard Bauch, "Modern Communication Systems Using MATLAB," CENGAGE, 3rd Edition, 2013*

PREREQUISITES

- Signal and Systems
- Probability Theory and Stochastic Processes
  - Random Variables
  - Random Processes
  - Correlation Functions and Power Spectra
- Linear algebra and matrix operation

III B.Tech I-Sem (E.C.E)	**T**	**Tu**	**C**
SYLLABUS **JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY ANANTAPUR**	**3**	**1**	**3**
(15A04502) DIGITAL COMMUNICATION SYSTEMS			

UNIT I

Source Coding Systems: Introduction, sampling process, quantization, quantization noise, conditionsfor optimality of quantizers, encoding, Pulse-Code Modulation (PCM), Line codes, Differential encoding, Regeneration, Decoding & Filtering, Noise considerations in PCM systems, Time-Division Multiplexing (TDM), Synchronization, Delta modulation (DM), Differential PCM (DPCM), Processing gain, Adaptive DPCM (ADPCM), Comparison of the above systems.

UNIT II

Baseband Pulse Transmission: Introduction, Matched filter, Properties of Matched filter, Matchedfilter for rectangular pulse, Error rate due to noise, Inter-symbol Interference (ISI), Nyquist's criterion for distortion less baseband binary transmission, ideal Nyquist channel, Raised cosine filter & its spectrum, Correlative coding – Duo binary & Modified duo binary signaling schemes, Partial response signaling, Baseband M-array PAM transmission, Eye diagrams.

UNIT III

Signal Space Analysis: Introduction, Geometric representation of signals, Gram-Schmidt orthogonalization procedure, Conversion of the Continuous AWGN channel into a vector channel, Coherent detection of signals in noise, Correlation receiver, Equivalence of correlation and Matched filter receivers, Probability of error, Signal constellation diagram.

UNIT IV

Passband Data Transmission: Introduction, Passband transmission model, Coherent phase-shiftkeying – binary phase shift keying (BPSK), Quadrature shift keying (QPSK), Binary Frequency shift keying (BFSK), Error probabilities of BPSK, QPSK, BFSK, Generation and detection of Coherent BPSK, QPSK, & BFSK, Power spectra of above mentioned modulated signals,M-array PSK, M-array quadrature amplitude modulation (M-array QAM), Non-coherent orthogonal modulation schemes -Differential PSK, Binary FSK, Generation and detection of non-coherent BFSK, DPSK, Comparison of power bandwidth requirements for all the above schemes.

UNIT V

Channel Coding: Error Detection & Correction - Repetition & Parity Check Codes, Interleaving,Code Vectors and Hamming Distance, Forward Error Correction (FEC) Systems, Automatic Retransmission Query (ARQ) Systems, Linear Block Codes – Matrix Representation of Block Codes, Convolutional Codes – Convolutional Encoding, Decoding Methods.

SIGNAL

1.
a gesture, action, or sound that is used to convey information or instructions, typically by prearrangement between the parties concerned.

2.
an electrical impulse or radio wave transmitted or received.

3.convey information or instructions by means of a gesture, action, or sound.

SYSTEM

1.
a set of things working together as parts of a mechanism or an interconnecting network; a complex whole.

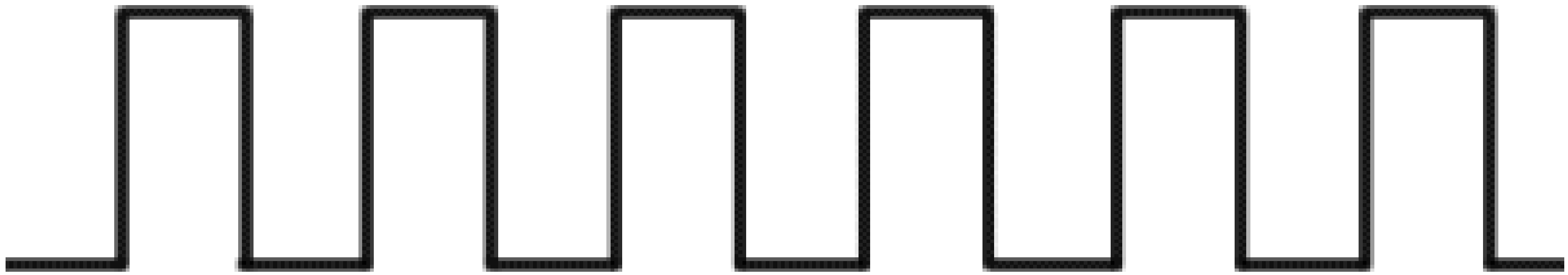
2.
a set of principles or procedures according to which something is done; an organized scheme or method.

Difference between analog and digital signals

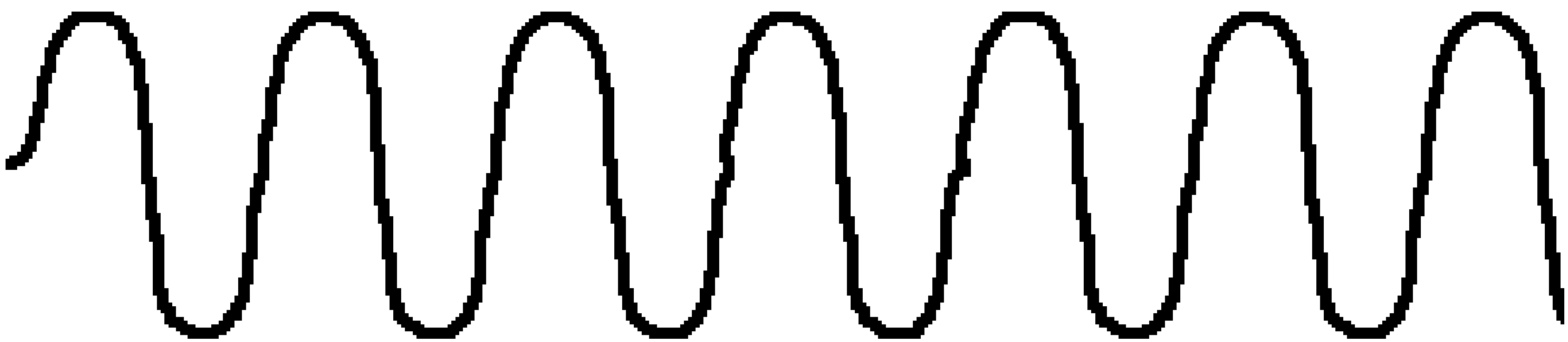
S. No.	Analog signal	Digital Signal
1	Analog signals are continuous signals	Digital signals are discrete signals.
2	Analog signal uses continuous values for representing the information.	A digital signal uses discrete values for representing the information.
3	Analog signals can be affected by the noise during the transmission.	Digital signals cannot be affected by the noise during transmission.
4	Accuracy of Analog signal is affected by the noise.	Digital signals are noise-immune hence there accuracy is less affected
5	Devices which are using analog signals are less flexible	Device using digital signals are very flexible
6	Analog signals consumes less bandwidth	Digital signals consume more bandwidth.
7	Analog signal are stored in the form of continuous wave form.	Digital signals are stored in the form of binary bits "0", "1".
8	Analog signals have low cost.	Digital signals have high cost.
9	Analog signals are portable.	Digital signals are not Portable.
10	Analog signals give observation error	Digital Signals doesn't give observation error.

Analog Communication	Digital Communication
Transmitted modulated signal is analog in nature.	Transmitted signal is digital i.e. train of digital pulses.
Amplitude, frequency or phase variations in the transmitted signal represent the information or message.	Amplitude, width or position of the transmitted pulses is constant. The message is transmitted in the form of code words.
Noise immunity is poor for AM, but improved for FM and PM.	Noise immunity is excellent.
It is not possible to separate out noise and signal. Therefore, repeaters cannot be used.	It is possible to separate signal from noise. Therefore, repeaters can be used.
Coding is not possible.	Coding techniques can be used to detect and correct the errors.
Bandwidth required is lower than that for the digital modulation method.	Due to higher bit rates, higher channel bandwidth is required.
FDM is used for multiplexing.	TDM is used for multiplexing.
Not suitable for transmission of secret information in military applications.	Due to coding techniques, it is suitable for military applications.
Analog modulation systems are AM, FM, PM, PAM, AWM, etc.	Digital modulation systems are PCM, DM, ADM, DPCM, etc.

Digital Signals

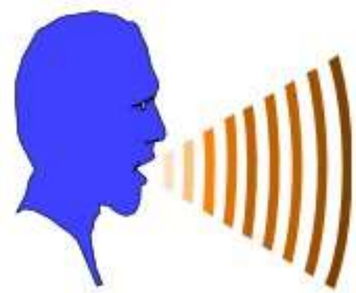


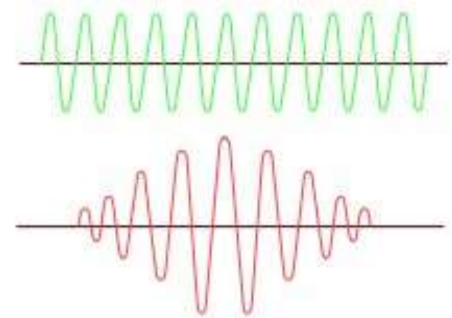
Analog Signals



Transmission Systems

- Analog Communications
  - Continuous modulation
  - Fidelity is usually defined in terms of SNR.





- Digital Communications
  - Signals made up of discrete symbols selected from a finite set (e.g., binary data).
  - Fidelity or Accuracy is specified in terms of bit error rate (Probability of making a bit error).

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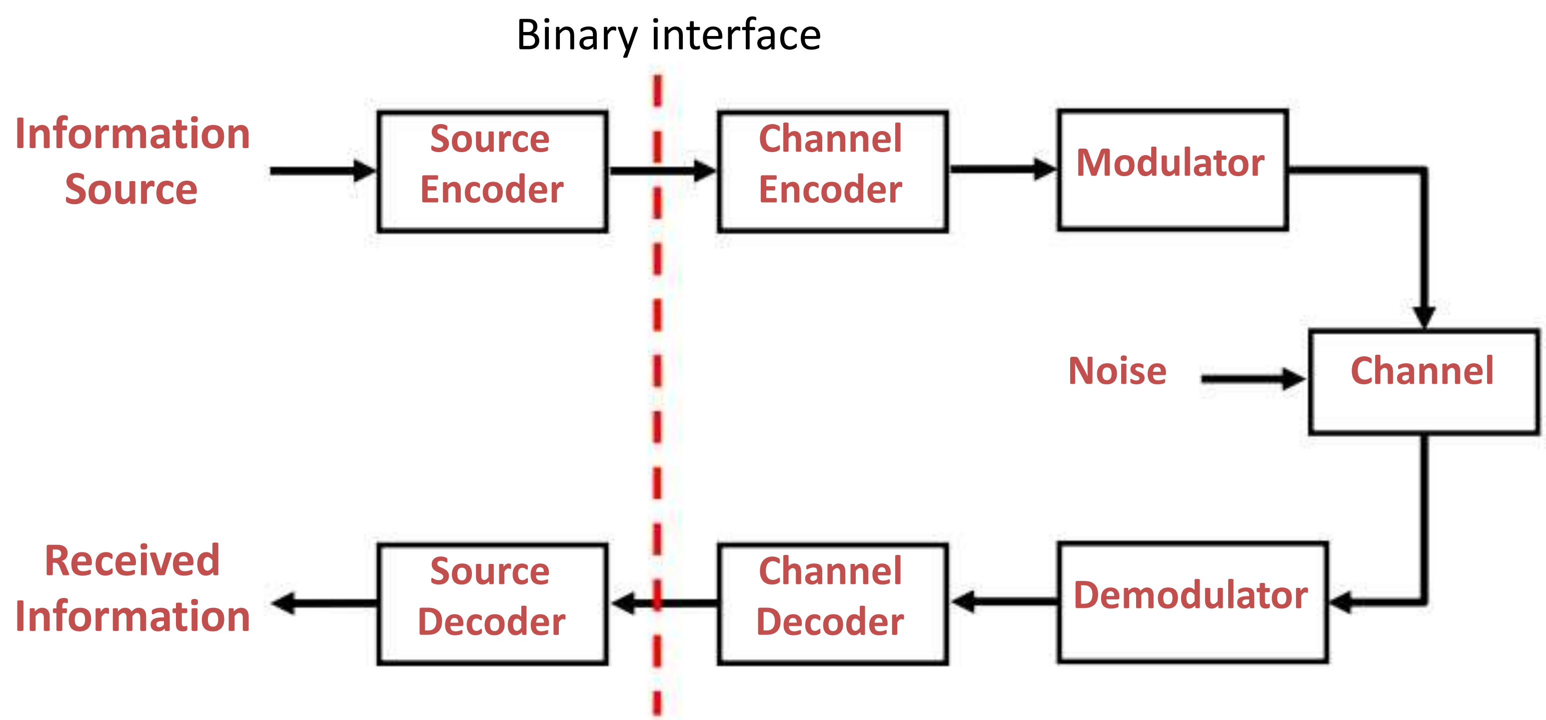
Simplified block diagram of a digital communication system



Elements of digital communications system

Discrete information source

Source Encoder

Channel Encoder

Modulator

Electrical Communication Channel

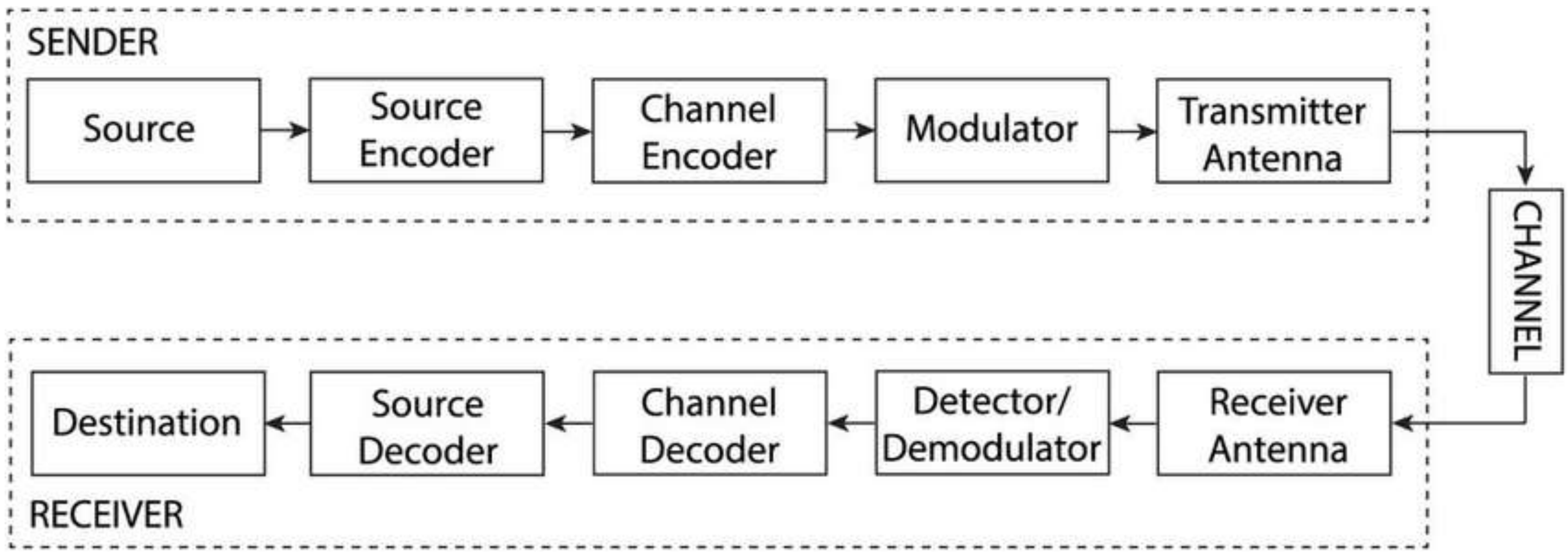
Noise

Demodulator

Channel Decoder

Source Decoder

Destination


SENDER
Source
Source Encoder
Channel Encoder
Modulator
Transmitter Antenna
CHANNEL
Receiver Antenna
Detector/ Demodulator
Channel Decoder
Source Decoder
Destination
RECEIVER

Source Encoder

Source encoding aims to convert information waveforms (text, audio, image, video, etc.) into bits, the universal currency of information in the digital world.

The three major steps are:

- Sampling: convert the continuous-time analog waveform to discrete-time sequence (but still continuous-valued).

- Quantization: convert each continuous-valued symbol to discrete-valued representatives.

- Data compression: remove the redundancy in the data and generate roughly uniformly distributed bits. Source decoding does the reverse of encoding.

Source Encoder and Decoder

In source coding, the encoder maps the digital signal generated at the source output into another signal in digital form. The mapping is one to one and the objective is to eliminate or reduce the redundancy so as to provide an efficient representation of the source output.

The source decoder simply performs the inverse mapping and thereby delivers to the user destination, a reproduction of the digital source output. The advantage of source coding is to reduce the bandwidth of transmission.

Channel Encoder and Decoder

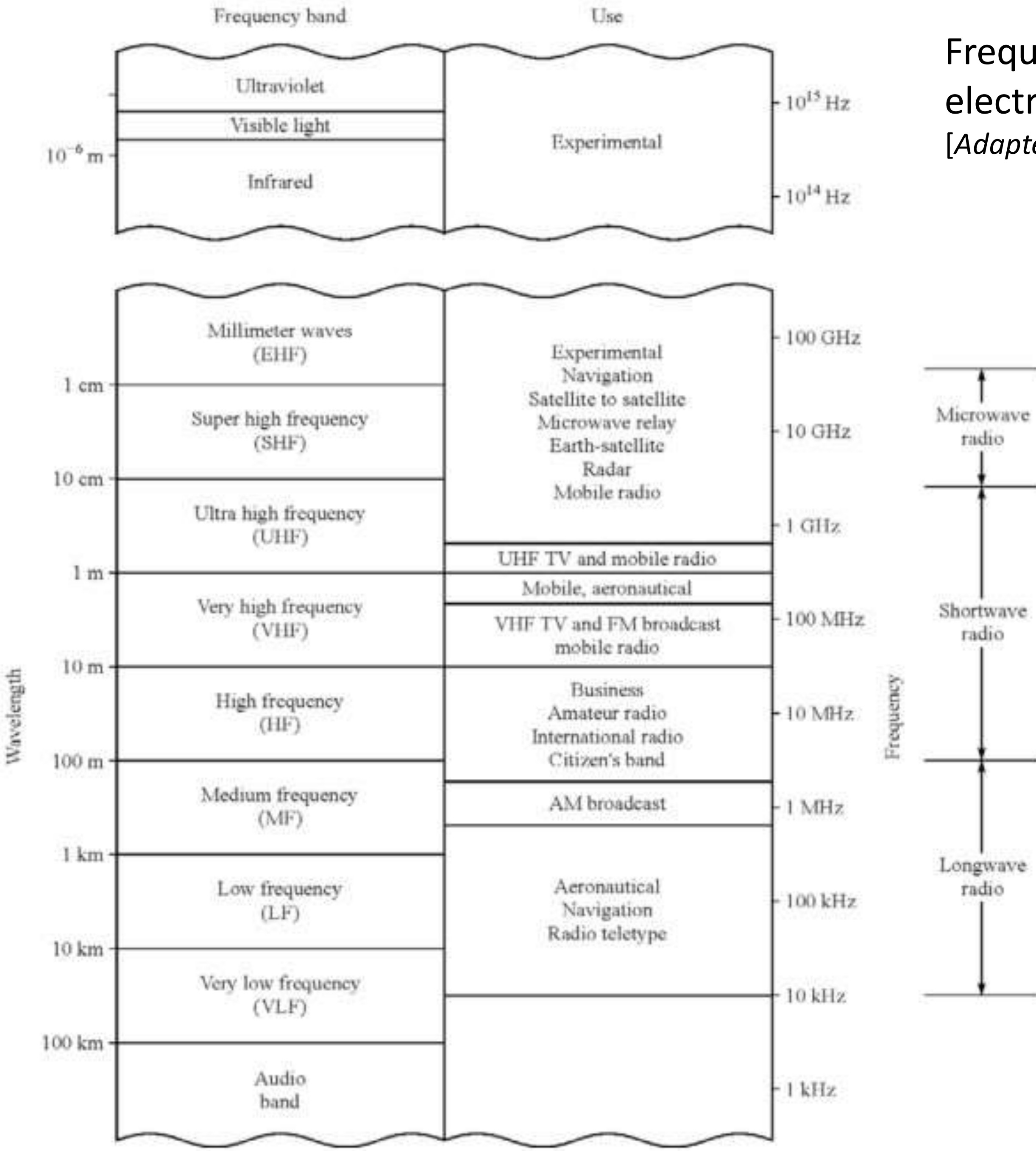
The purpose of channel encoder is to map the incoming digital signal into a channel input and for the decoder to map the channel output into an output digital signal in such a way that the effect of channel noise is minimized. That is the combined roll of channel encoder and decoder is to provide reliable communication. This provision is satisfied by introducing redundancy in a prescribed fashion.

In the channel encoder and exploiting it in the decoder, to reconstruct the original encoder input as accurately as possible.

The overall purpose of the system is to transmit the message or sequence of symbols coming out of source to a destination point as a high rate and accuracy as possible.

Communication Channels

- Wireline channels
  - Telephone network
  - Twisted-pair wire lines and coaxial cable
- Fiber-optic channels
  - Higher bandwidth, > GHz
- Underwater acoustic channels
  - With increasing interest, but very challenging to design
- Storage channels
  - Magnetic tape, magnetic disks, optical disks, compact disks
- Wireless channels



Frequency range for wireless electromagnetic channels.
[*Adapted from Carlson* (1975), 2nd edition]

What makes Communications Systems Challenging?

- Transmission in a particular application depends on many factors. This includes:
  - information rate (bit rate)
  - cost
  - number of users
  - quality of service (BER, Delay, SNR)
  - medium over which the information is to be sent - Channel.

- Example: wireless systems requires a different design from an optical fibre communications link.

What are the Features of a Good Communication System?

- Small signal power (measured in Watts or dBm)
- Large data rate (measured in bits/sec)
- Small bandwidth (measured in Hertz)
- Low distortion (measured in SNR or bit error rate)
- Low cost - with digital communications, large complexity does not always result in large cost

In practice, there must be tradeoffs made in achieving these goals

System Design Tradeoffs

Data Rate vs. Bandwidth

- Bandwidth Efficiency

  - defined as the ratio of data rate *R* to bandwidth *W* (bits/sec/Hz)

Want large bandwidth efficiency

  - Typical current wireless systems provide < 1bit/sec/Hz
  - Newly researched systems can provide > 10bits/sec/Hz

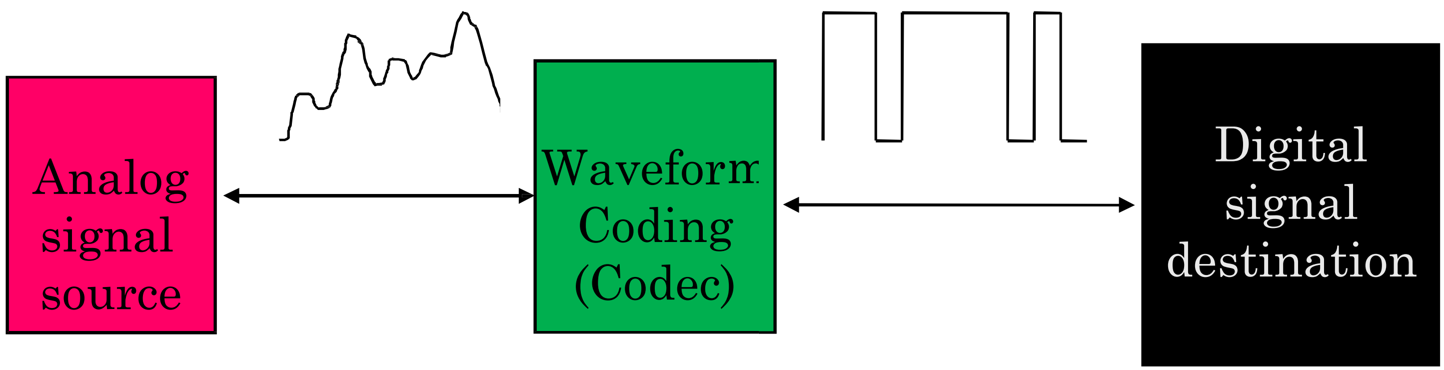
- Increased data rate leads to shorter data pulses which leads to larger bandwidth

  - This tradeoff (*Data Rate vs. Bandwidth)* cannot be avoided.

- Some modulation schemes use bandwidth more efficiently than others.

Sampling
and
Quantization

- Digital representation of analog signals

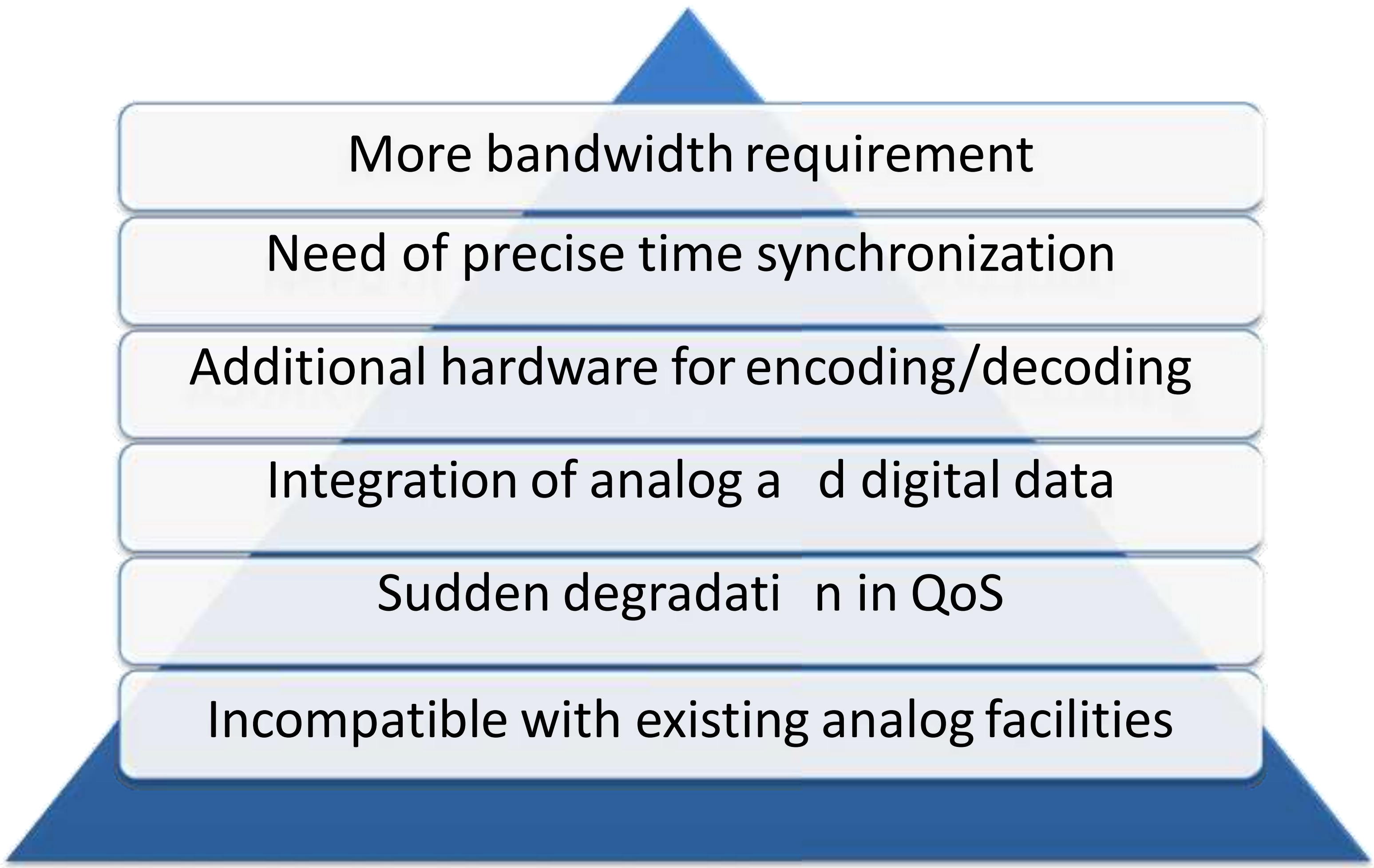


Analog-to-Digital Encoding

Advantages of Digital Transmissions

- Noise immunity
- Error detection and correction
- Ease of multiplexing
- Integration of analog and digital data
- Use of signal regenerators
- Data integrity and security
- Ease of evaluation and measurements
- More suitable for processing ....

Disadvantages of Digital Transmissions

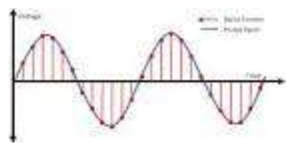


SAMPLING PROCESS

Sampling is the process of converting analog signal into a discrete signal or making an analog or continuous signal to occur at a particular interval of time, this phenomena is known as sampling.

SAMPLING THEOREM:-

Sampling theorem states that a band limited signal having no frequency components higher than fm hertz can be sampled if its sampling freq is equal to or greater than Nyquist rate.



Analog Signal Representation

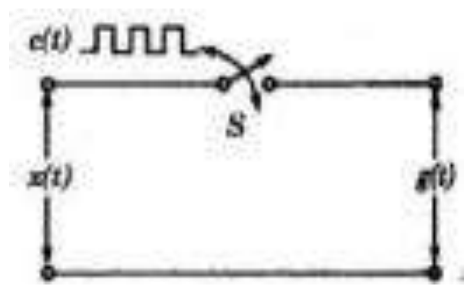
Sampling Techniques

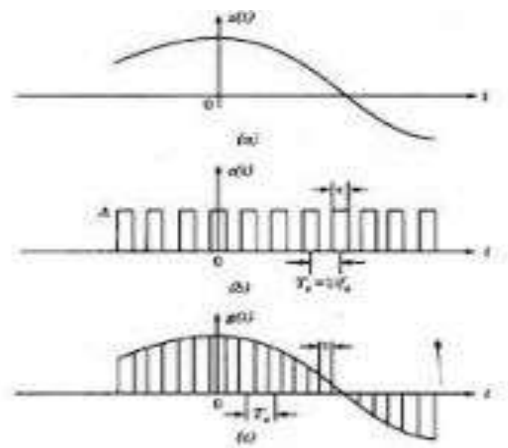
Their are basically three types of Sampling techniques, namely:

1. Natural Sampling
2. Flat top Sampling
3. Ideal Sampling

1. Natural Sampling:

Natural Sampling is a practical method of sampling in which pulse have finite width equal to τ. Sampling is done in accordance with the carrier signal which is digital in nature.





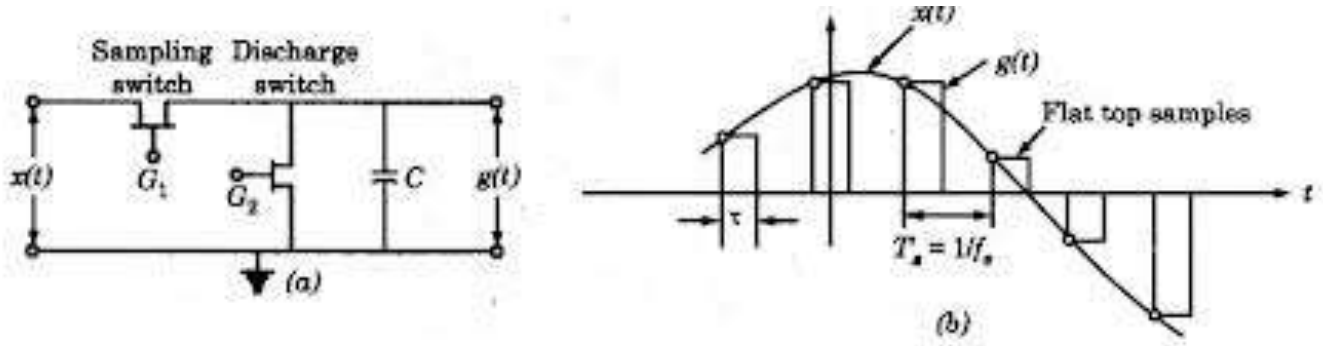
Functional Diagram of Natural Sampler   Natural Sampled Waveform

With the help of functional diagram of a Natural sampler, a sampled signal g(t) is obtained by multiplication of sampling function c(t) and the input signal x(t). Spectrum of Natural Sampled Signal is given by:

$$G(f) = A\tau / T_s . [\, \Sigma \sin c(n\, f_s . \tau)\, X(f - n\, f_s)]$$

2. Flat Top Sampling:

Flat top sampling is like natural sampling i.e; practical in nature. In comparison to natural sampling flat top sampling can be easily obtained. In this sampling techniques, the top of the samples remains constant and is equal to the instantaneous value of the message signal x(t) at the start of sampling process. Sample and hold circuit are used in this type of sampling.



•**Figure(a),** shows functional diagram of a sample hold circuit which is used to generate fat top samples.

•**Figure(b),** shows the general waveform of the flat top samples. It can be observed that only starting edge of the pulse represent the instantaneous value of the message signal x(t).

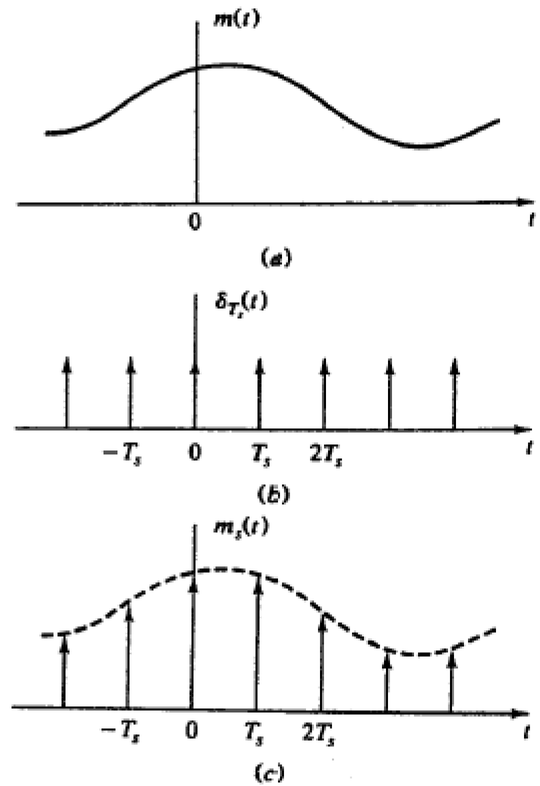
Spectrum of Flat top Sampled Signal is given by: $G(f) = f_s .[\ \Sigma\ X(f-n f_s). H(f)]$

3.Ideal Sampling Technique:

Ideal sampling is also known as Instantaneous sampling or Impulse Sampling. Train of impulse is used as a carrier signal for ideal sampling. In this sampling technique the sampling function is a train of impulses and the principle used is known as multiplication principle.

Here,

Figure (a), represent message signal or input signal or signal to be sampled.
Figure (b), represent the sampling function.
Figure (c), represent the resultant signal.



Spectrum of IdealSampled Signal is given by: G(f) = f_s .[Σ X(f-n f_s)]

NYQUIST RATE:

Nyquist rate is the rate at which sampling of a signal is done so that overlapping of frequency does not take place. When the sampling rate become exactly equal to $2f_m$ samples per second, then the specific rate is known as Nyquist rate. It is also know aas the minimum sampling rate and given by: $\mathbf{f_s = 2f_m}$

Effect of Under sampling: ALIASING

It is the effect in which overlapping of a frequency components takes place at the frequency higher than Nyquist rate. Signal loss may occur due to aliasing effect. We can say that aliasing is the phenomena in which a high frequency component in the frequency spectrum of a signal takes identity of a lower frequency component in the same spectrum of the sampled signal.

Because of overlapping due to process of aliasing, sometimes it is not possible to overcome the sampled signal x(t) from the sampled signal g(t) by applying the process of low pass filtering since the spectral components in the overlap regions . hence this causes the signal to destroy.

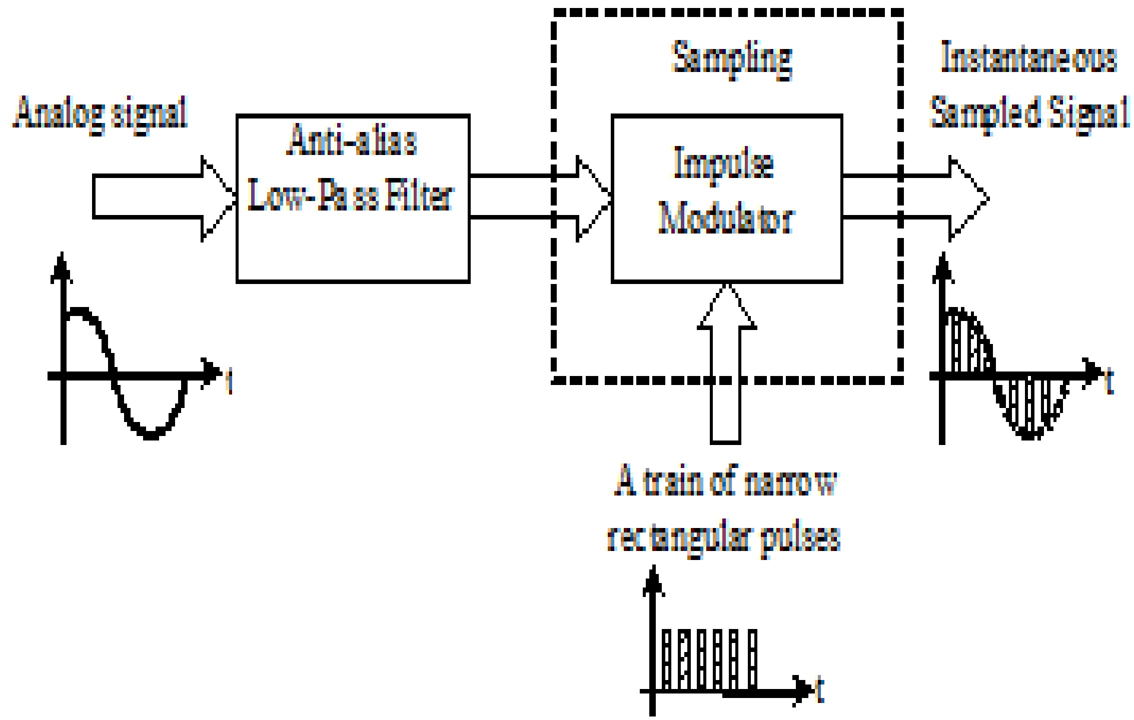
The Effect of Aliasing can be reduced:

1) Pre alias filter must be used to limit band of frequency of the required signal $\mathbf{f_m}$ Hz.

2) Sampling frequency $\mathbf{f_s}$ must be selected such that $\mathbf{f_s > 2f_m}$.

Antialiasing Filter

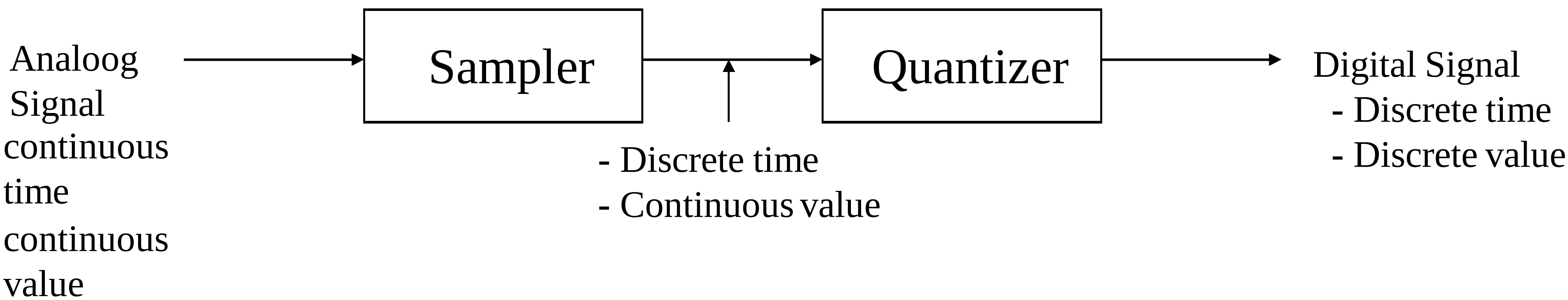
An anti-aliasing filter is a *low-pass filter* of sufficient higher order which is recommended to be used prior to sampling.



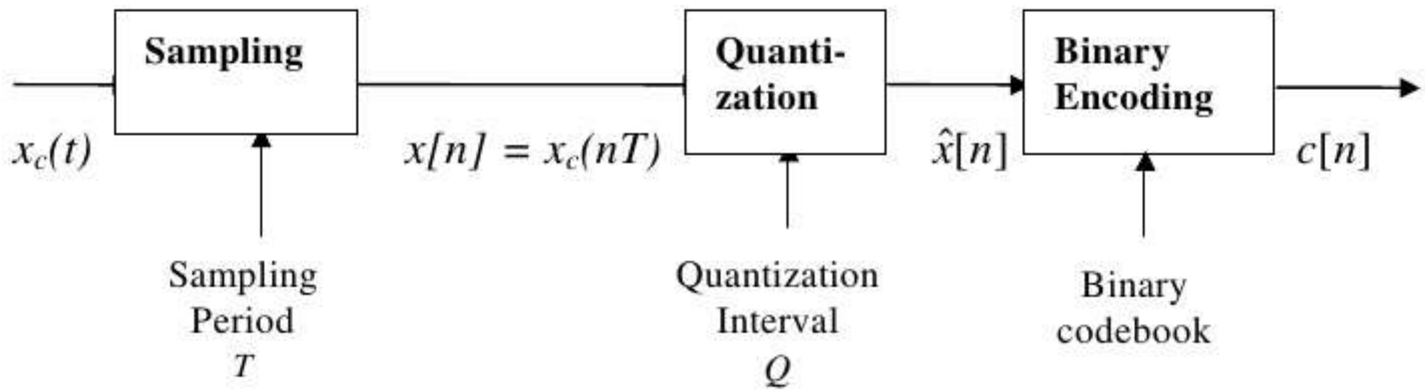
Minimizing Aliasing

Quantization

Quantization is a non linear transformation which maps elements from a continuous a finite set.It is also the second step required by A/D conversion.



Three Processes in A/D Conversion



- Sampling: take samples at time *nT*
  - T: sampling period;
  - $f_s = 1/T$: sampling frequency
- Quantization: map amplitude values into a set of discrete values *kQ*
  - Q: quantization interval or stepsize
- Binary Encoding
  - Convert each quantized value into a binary codeword

Types of Quantization

There are two types of Quantization

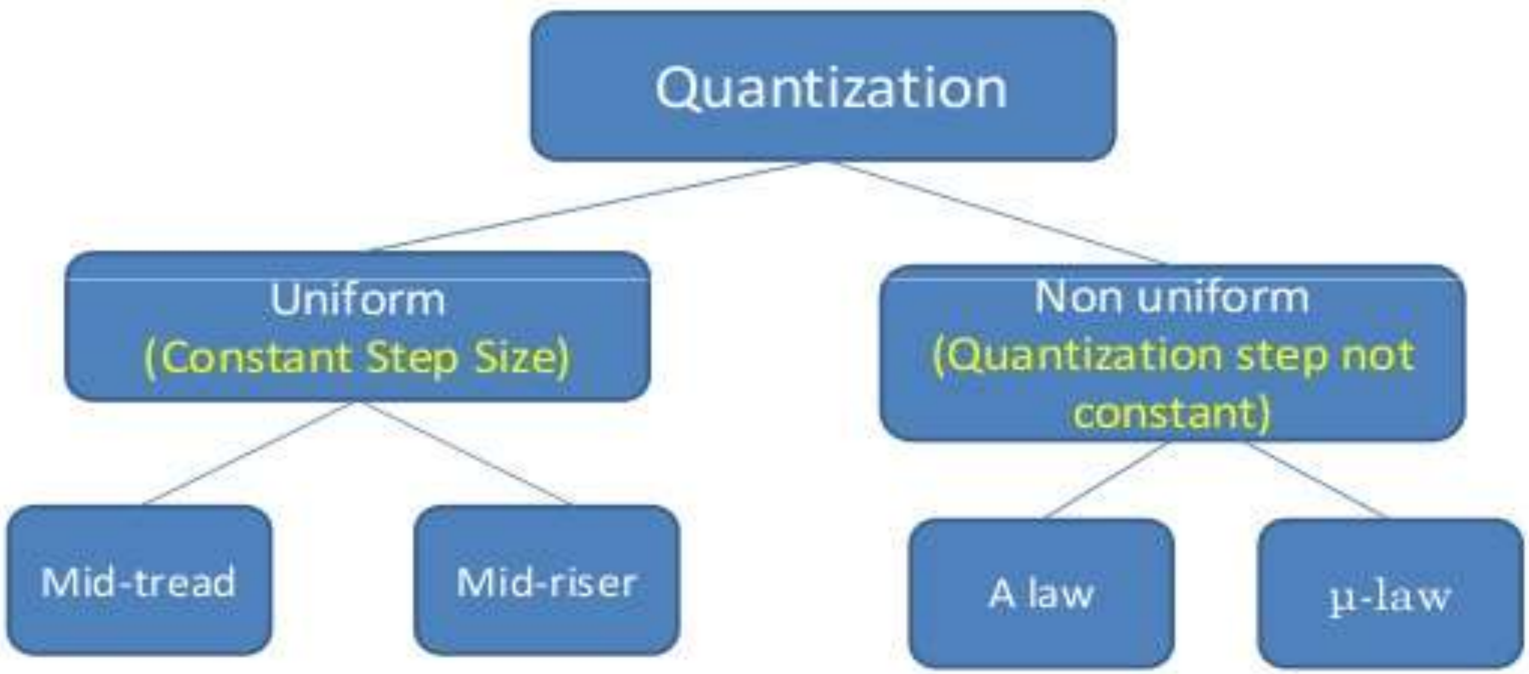
Uniform Quantization

The type of quantization in which the quantization levels are uniformly spaced is termed as a **Uniform Quantization**.

Non-uniform Quantization

The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a **Non-uniform Quantization**.

QUANTIZER CLASSIFIED

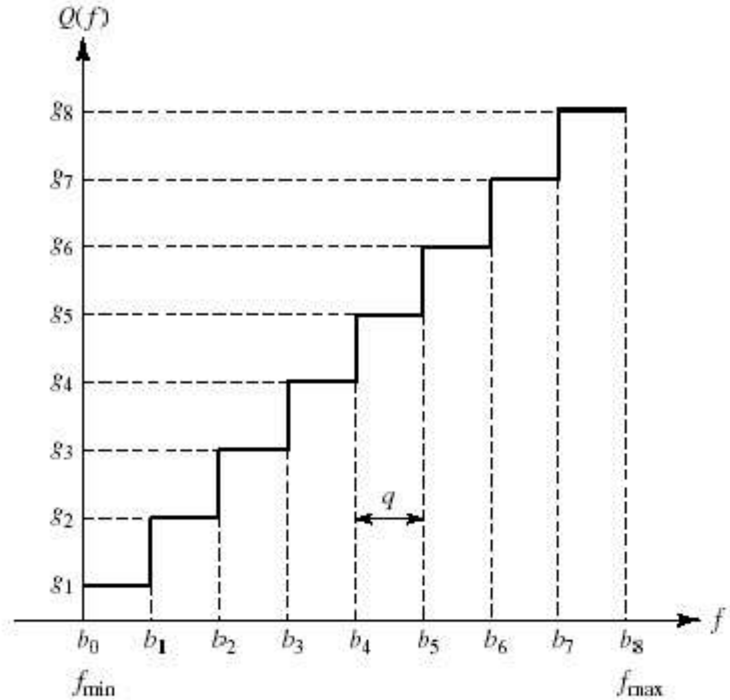


Uniform Quantization

- Applicable when the signal is in a finite range (f_{min}, f_{max})
- The entire data range is divided into L equal intervals of length Q *(known as quantization interval or quantization step-size)*

$$Q=(f_{max}-f_{min})/L$$

- Interval i is mapped to the middle value of this interval
- We store/send only the index of quantized value



$$\text{Index of quantized value} = Q_i(f) = \left\lfloor \frac{f - f_{\min}}{Q} \right\rfloor$$

$$\text{Quantized value} = Q(f) = Q_i(f)Q + Q/2 + f_{\min}$$

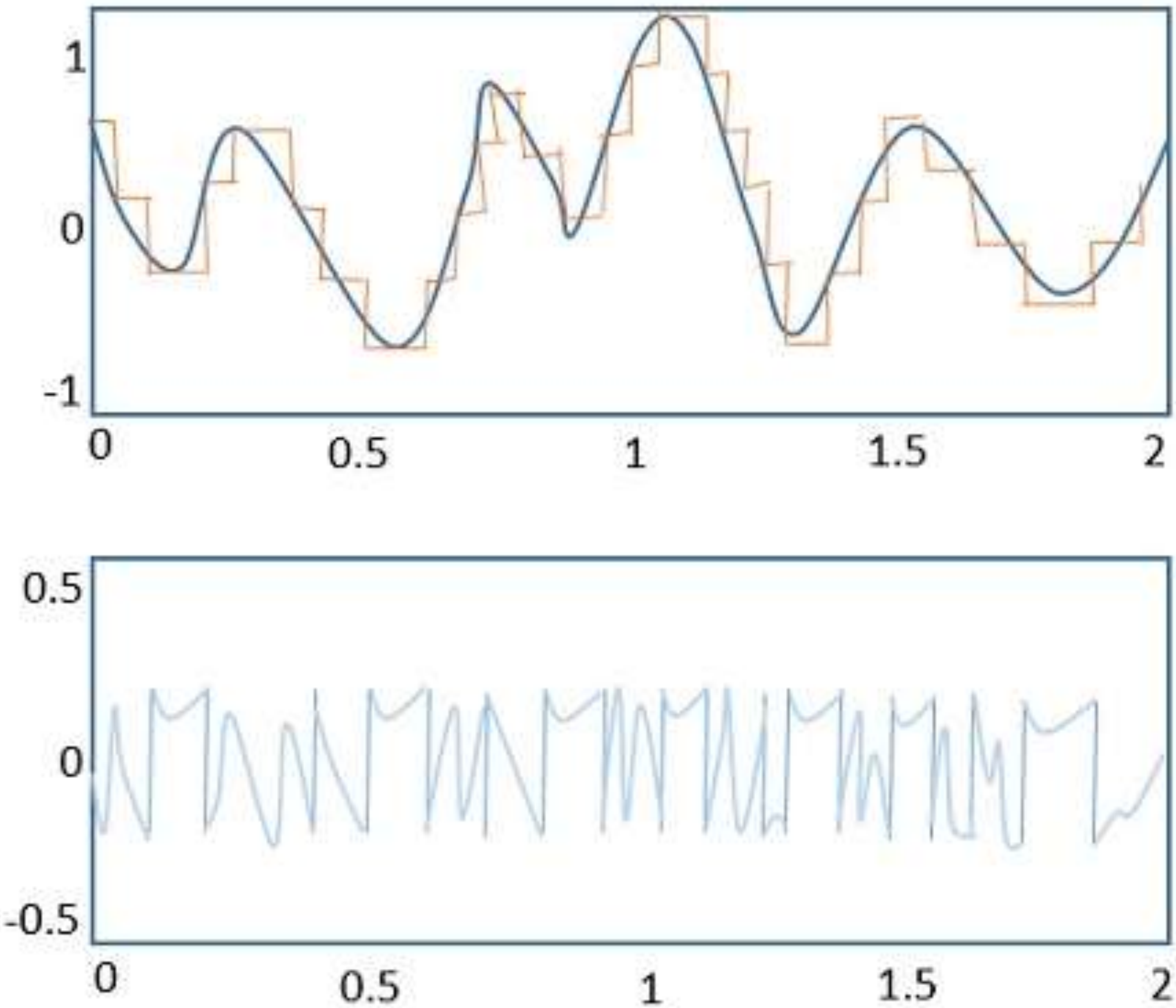
Quantization error

For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values.

The difference between an input value and its quantized value is called a **Quantization Error**. A **Quantizer** is a logarithmic function that performs

Quantization roundingoffthevalueroundingoffthevalue. An analog-to-digital converter (**ADC**) works as a quantizer.

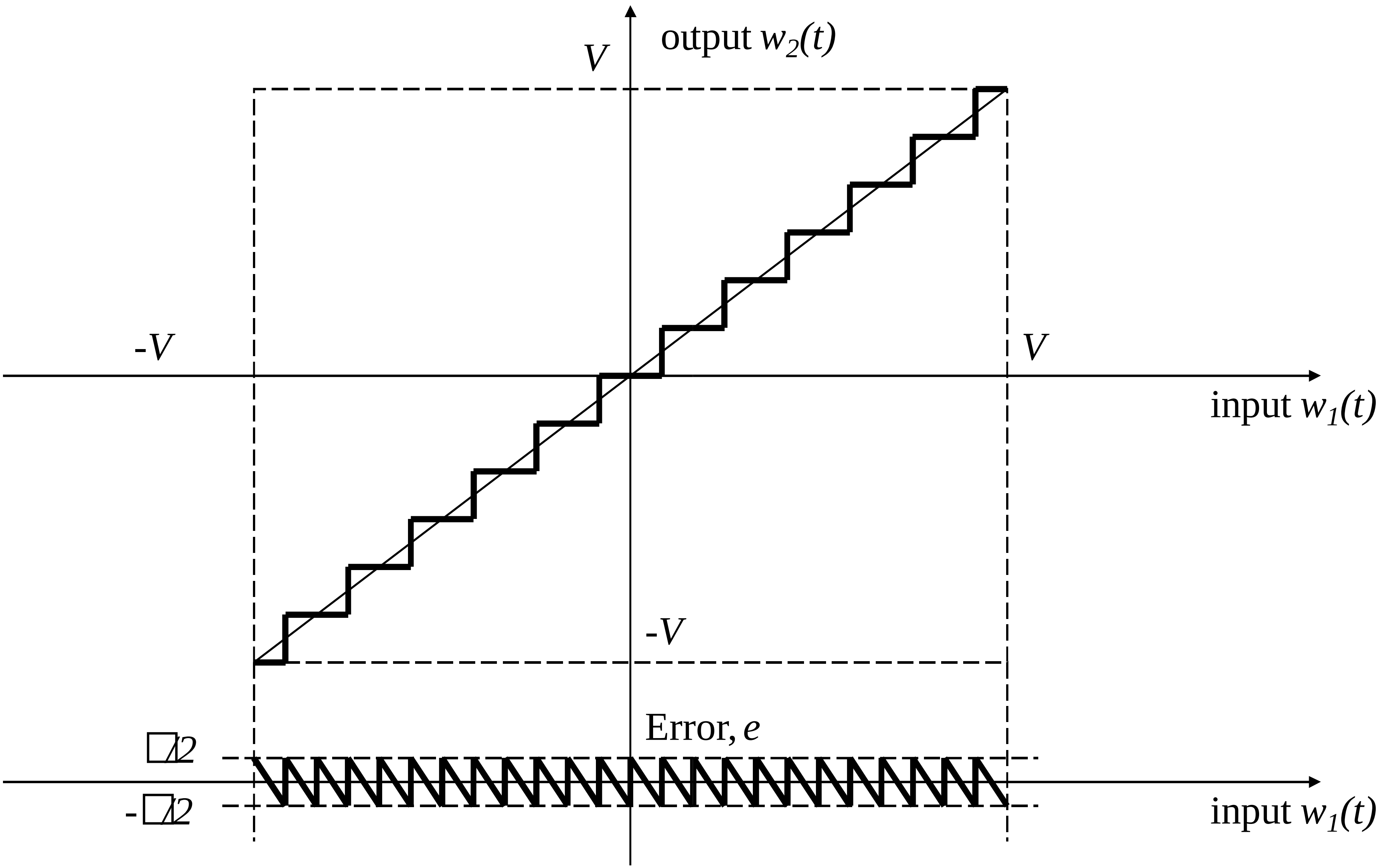
The following figure illustrates an example for a quantization error, indicating the difference between the original signal and the quantized signal.



Original and Quantized Signal

Quantization Error

Quantization Error, *e*



Types of uniform quantization

There are two types of uniform quantization.
They are :-

1. Mid-Rise type
2. Mid-Tread type.

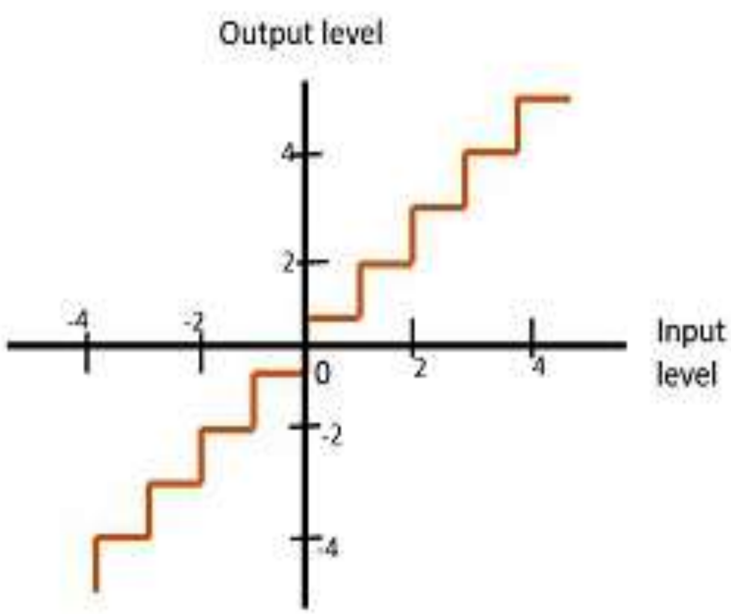


Fig 1 : Mid-Rise type Uniform Quantization

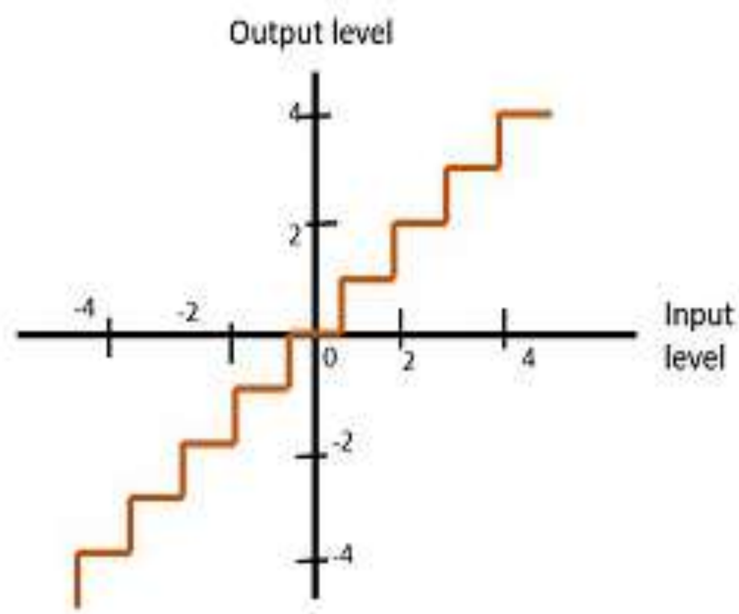


Fig 2 : Mid-Tread type Uniform Quantization

QUANTIZING PROCESS

- In midtread type, the decision levels are located at $\pm\Delta/2$, $\pm 3\Delta/2$,... and the representation levels are located at $\pm\Delta$, $\pm 2\Delta$,... where Δ is the step size.
- In midriser type, the decision levels are located at $\pm\Delta$, $\pm 2\Delta$,... and the representation levels are located at $\pm\Delta/2$, $\pm 3\Delta/2$, ...where Δ is the step size.
- The overload level is defined as the absolute value and which is one half of the peak to peak range of the input sample values.

QUANTIZING PROCESS

Number of representation levels is equal to twice the absolute value of the overload level divided by the step size

$$L = \frac{2x_{max}}{\Delta}$$

When the quantizer produces discrete output for a sample a **quantization error occurs** which is the difference between the input and output values of quantizer.

•The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.

•The **Mid-tread** type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.

•Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

- Problems with uniform quantization
  - Only optimal for uniformly distributed signal
  - Real audio signals (speech and music) are more concentrated near zeros
  - Human ear is more sensitive to quantization errors at small values
- Solution
  - Using non-uniform quantization
    - quantization interval is smaller near zero

Nonuniform Quantizer

The type of **quantization** in which the **quantization** levels are uniformly spaced is termed as a **Uniform Quantization**. The type of **quantization** in which the **quantization** levels are unequal and mostly the relation between them is logarithmic, is termed as a **Non-uniform Quantization**.

It is used to reduce quantization error and increase the dynamic range when an input signal is not uniformly distributed over its allowed range of values.

Need for non uniform quantization:

A **non-uniform quantizer** can be designed so that the **quantization** levels are spaced more closely for smaller amplitudes and spaced more far apart for larger amplitudes. as a result the signal-to-noise ratio can be made constant for both small signals and for large signals

In Non - Uniform Quantizer the step size varies. The use of a non – uniform quantizer is equivalent to passing the baseband signal through a compressor and then applying the compressed signal to a uniform quantizer. The resultant signal is then transmitted.

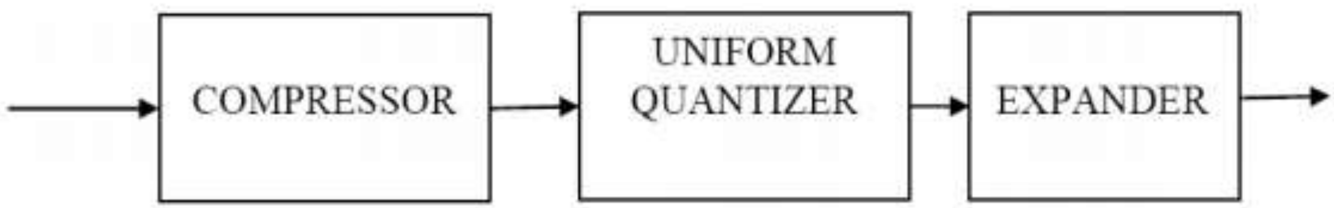


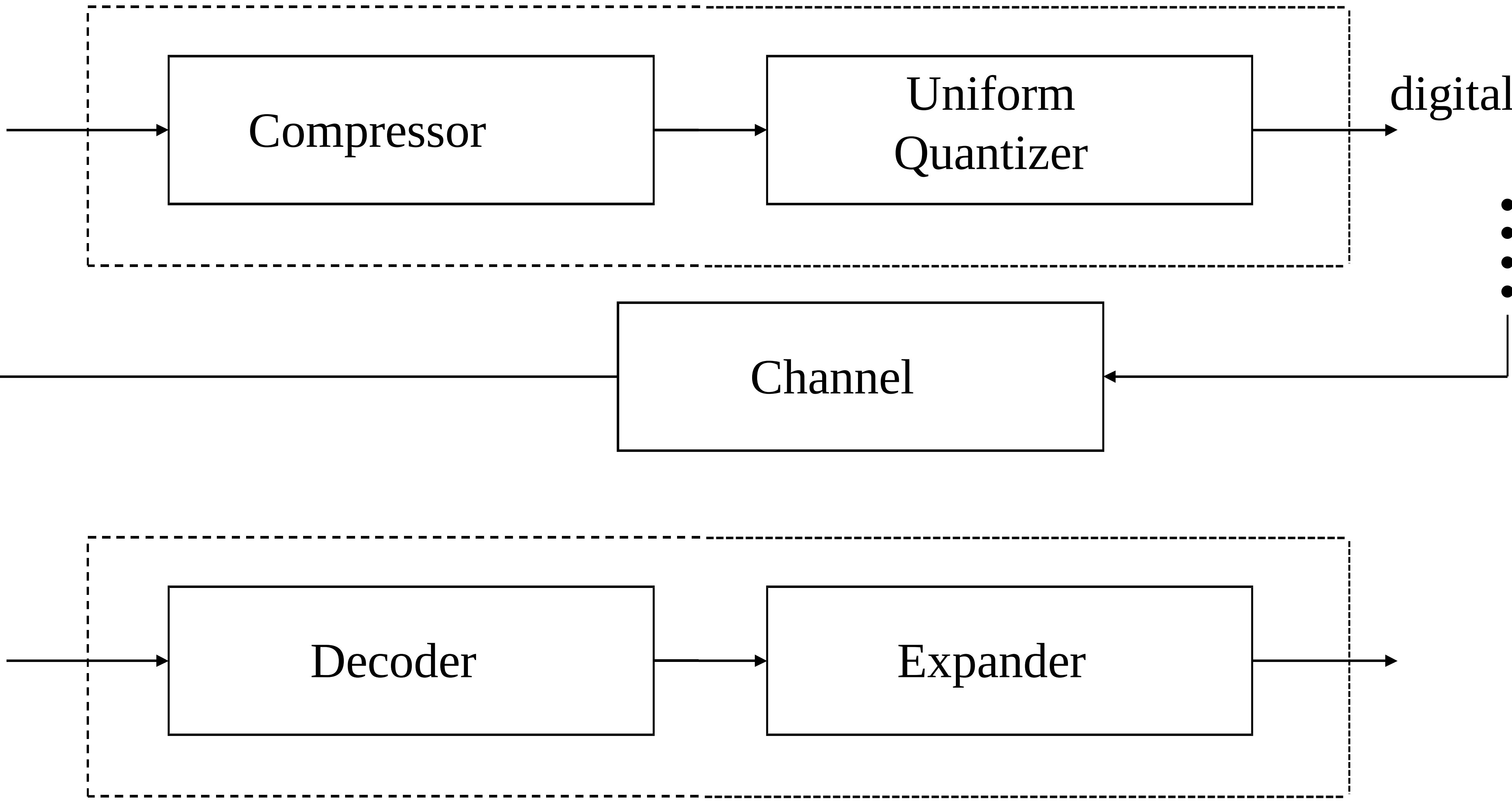
Fig: 2.14 MODEL OF NON UNIFORM QUANTIZER

At the receiver, a device with a characteristic complementary to the compressor called Expander is used to restore the signal samples to their correct relative level. The Compressor and expander take together constitute a Compander.

Compander = Compressor + Expander

compressing-and-expanding" is called "companding."

Nonuniform quantizer



Advantages of Non- Uniform Quantization:

1. Higher average signal to quantization noise power ratio than the uniform quantizer when the signal pdf is non uniform which is the case in many practical situation.

2. RMS value of the quantizer noise power of a non – uniform quantizer is substantially proportional to the sampled value and hence the effect of the quantizer noise is reduced.

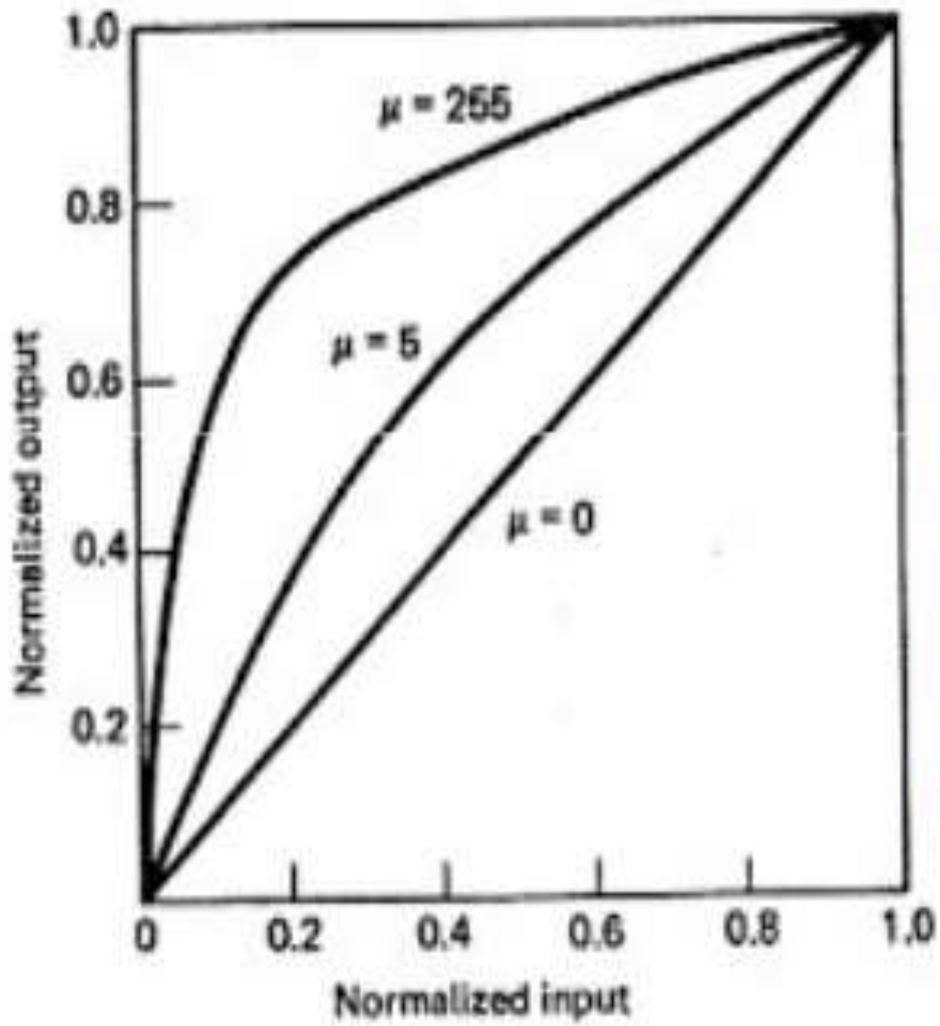
companding

- The word **Companding** is a combination of Compressing and Expanding, which means that it does both. The effects of noise and crosstalk are reduced by using this technique.

- There are two types of Companding techniques. They are −

- A-law Companding Technique
- Uniform quantization is achieved at **A = 1**, where the characteristic curve is linear and no compression is done.
- A-law has mid-rise at the origin. Hence, it contains a non-zero value.
- A-law companding is used for PCM telephone systems.

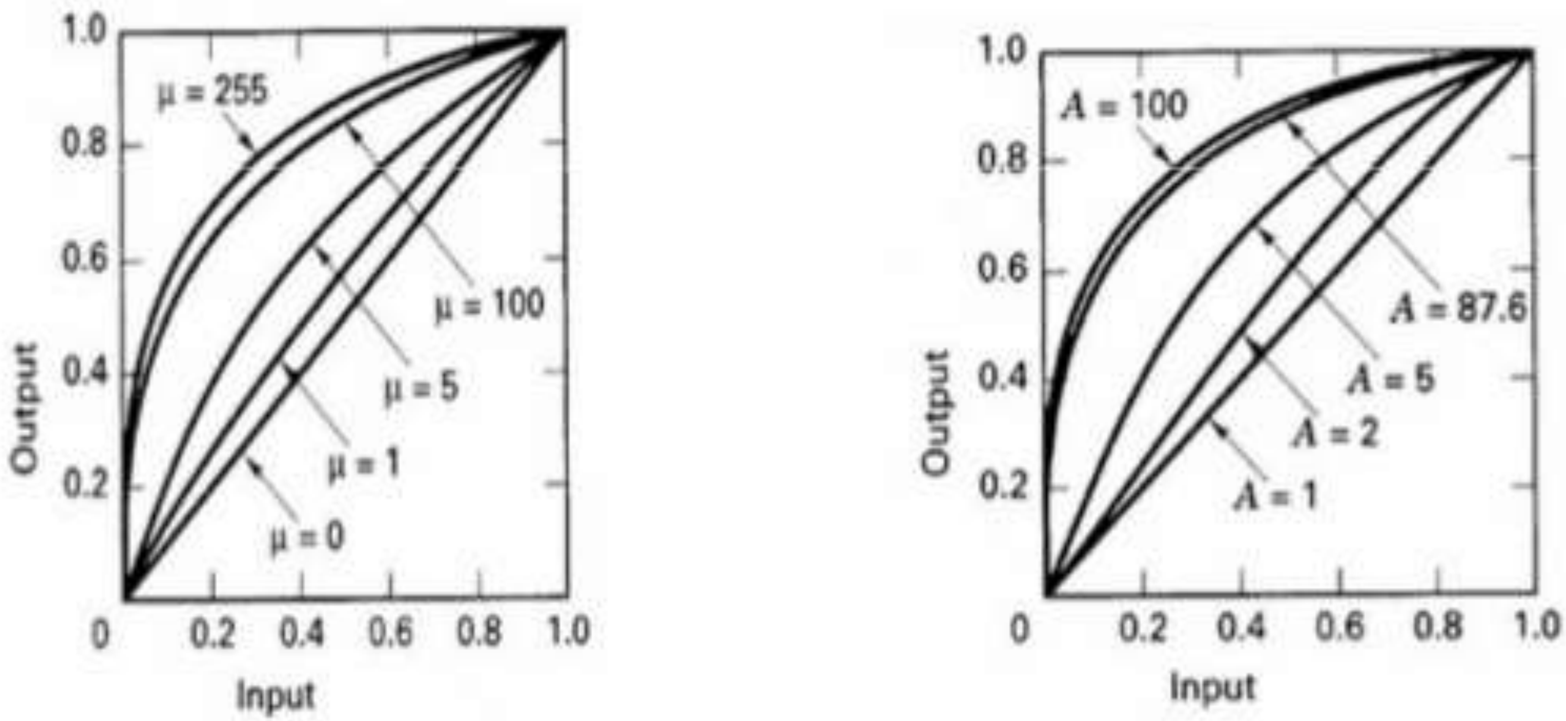
- μ-law Companding Technique
- Uniform quantization is achieved at **μ = 0**, where the characteristic curve is linear and no compression is done.
- μ-law has mid-tread at the origin. Hence, it contains a zero value.
- μ-law companding is used for speech and music signals.
- μ-law is used in North America and Japan.

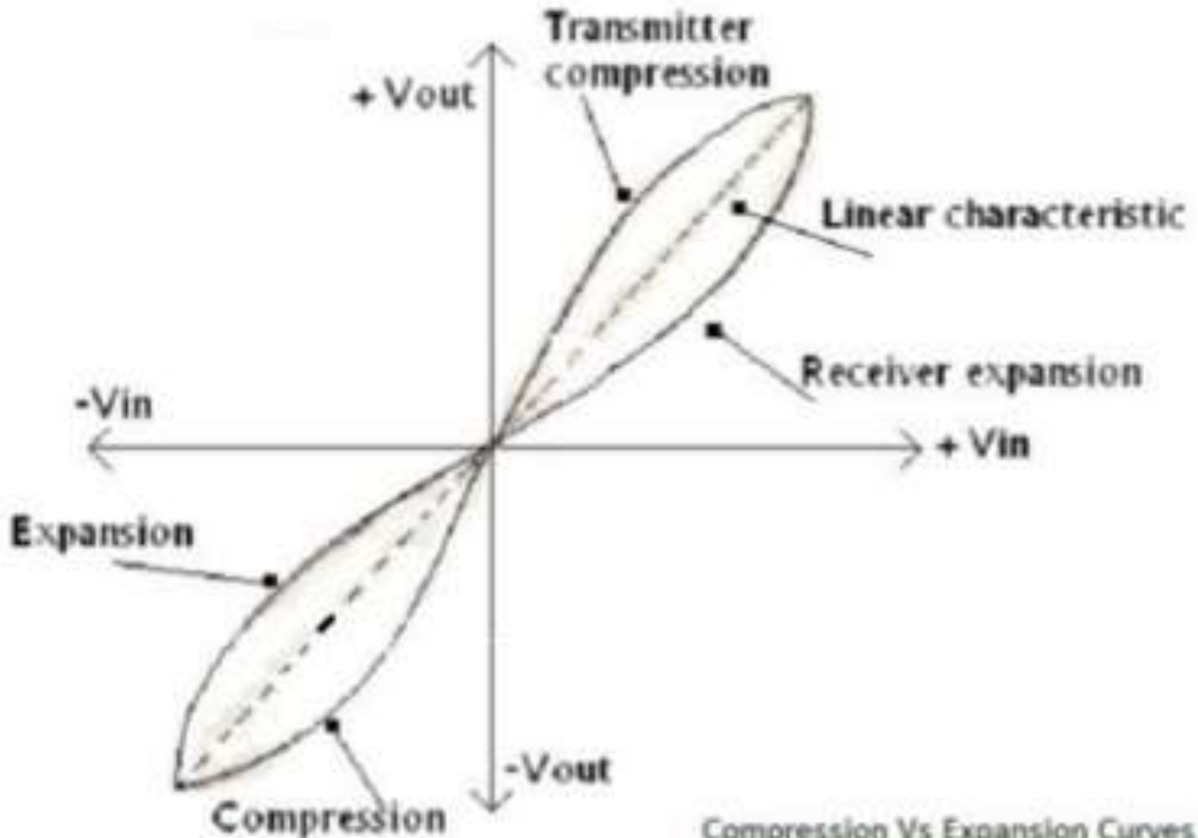


μ-law is used for PCM telephone systems in the US, Canada and Japan. A practical value for μ is 255.

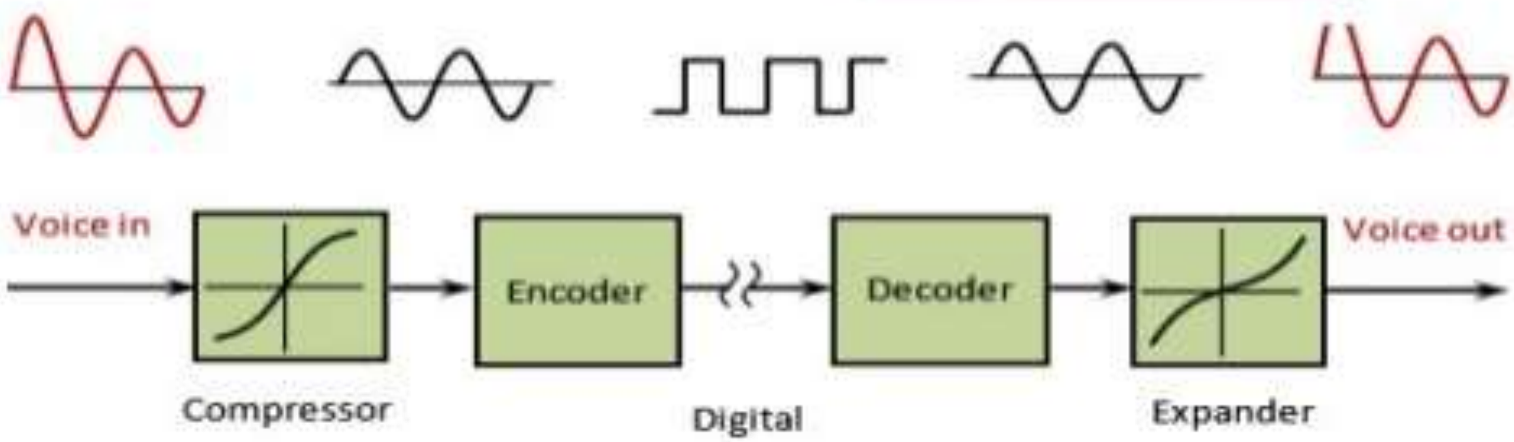
Compressor characteristic of μ-law

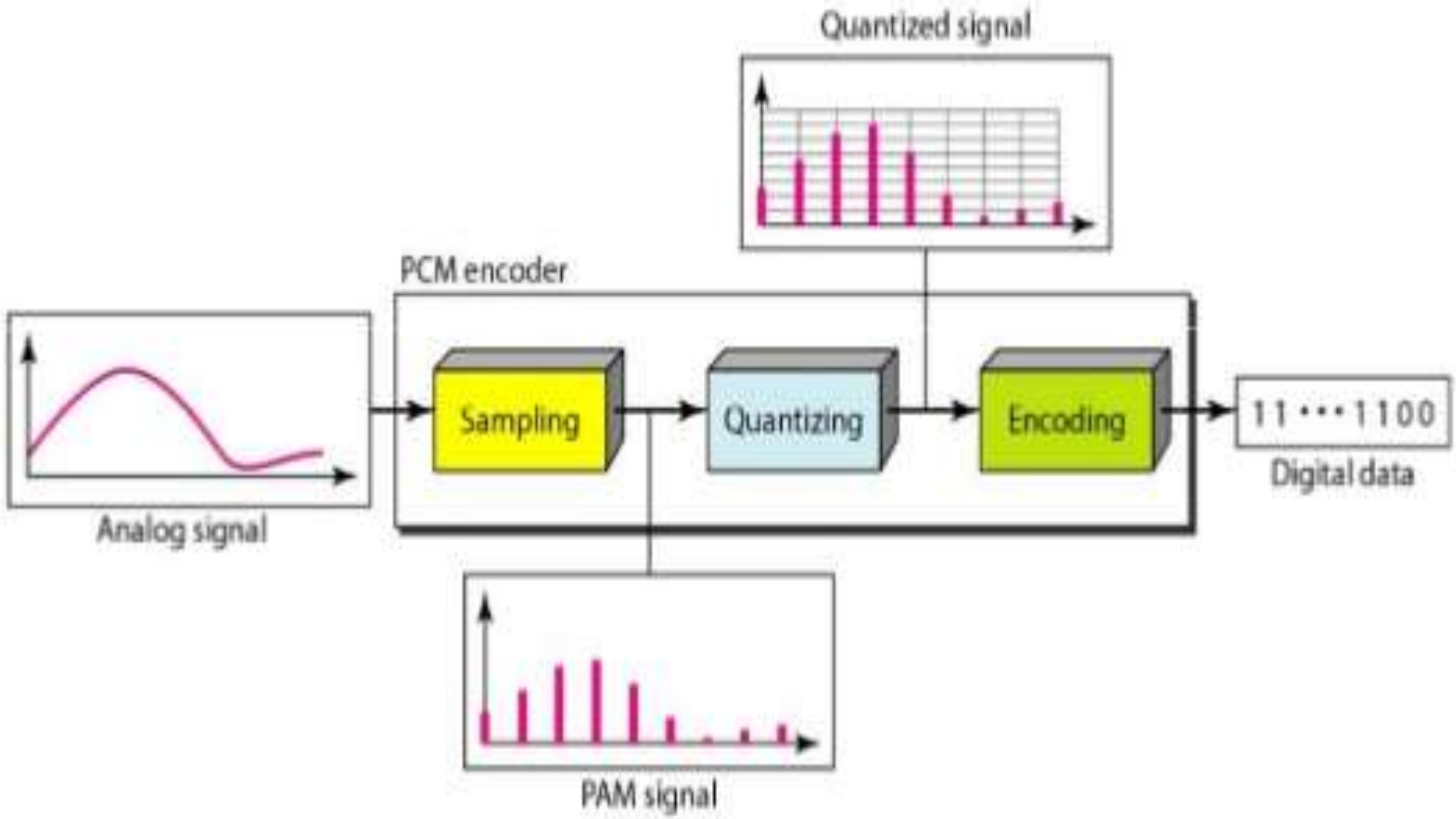
μ-law and A-law Compression Characteristics





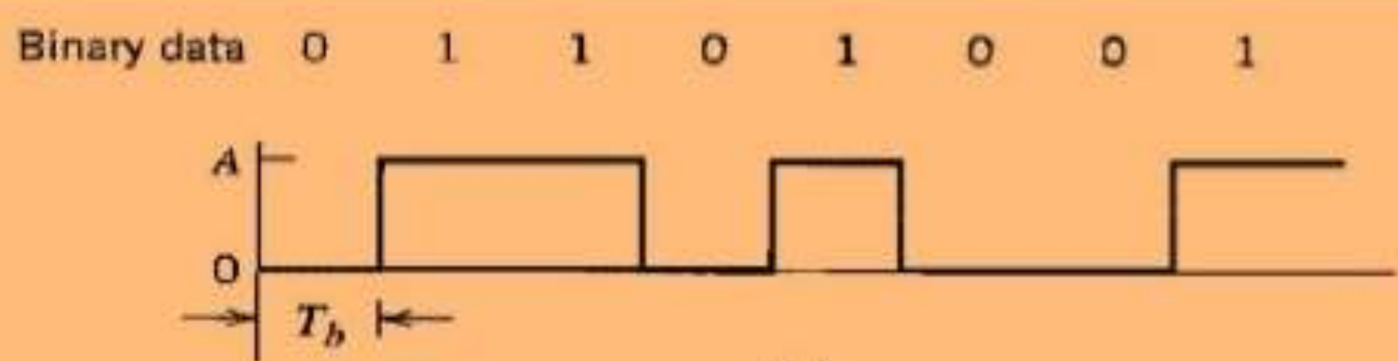
Compression Vs Expansion Curves




Quantized signal
PCM encoder
Sampling
Quantizing
Encoding
11 ••• 1100
Digital data
Analog signal
PAM signal

ENCODING

- To make the transmitted signal more robust to noise interference and make it suitable for transmission over the given channel
- Encoding process translates a discrete set of sample values to a more appropriate form of a signal called as codeword
- The presence or absence of pulse is called symbol
- The binary symbol 1 is represented by a pulse of constant amplitude for a one bit duration and symbol 0 is represented by switching off the pulse for one bit duration



(a)

Line codes:

1. Unipolar nonreturn-to-zero (NRZ) Signaling
2. Polar nonreturn-to-zero(NRZ) Signaling
3. Unipor nonreturn-to-zero (RZ) Signaling
4. Bipolar nonreturn-to-zero (BRZ) Signaling
5. Split-phase (Manchester code)

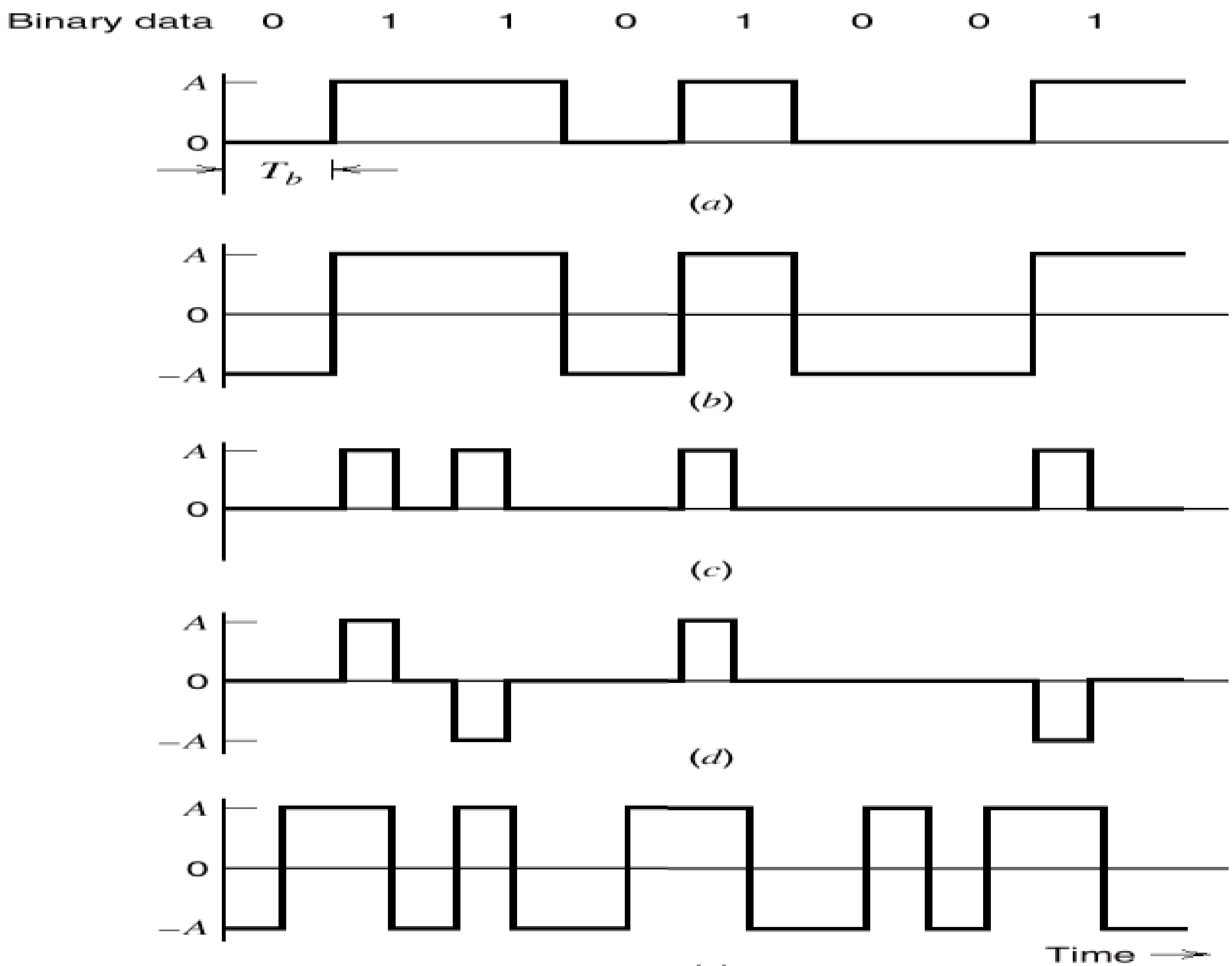
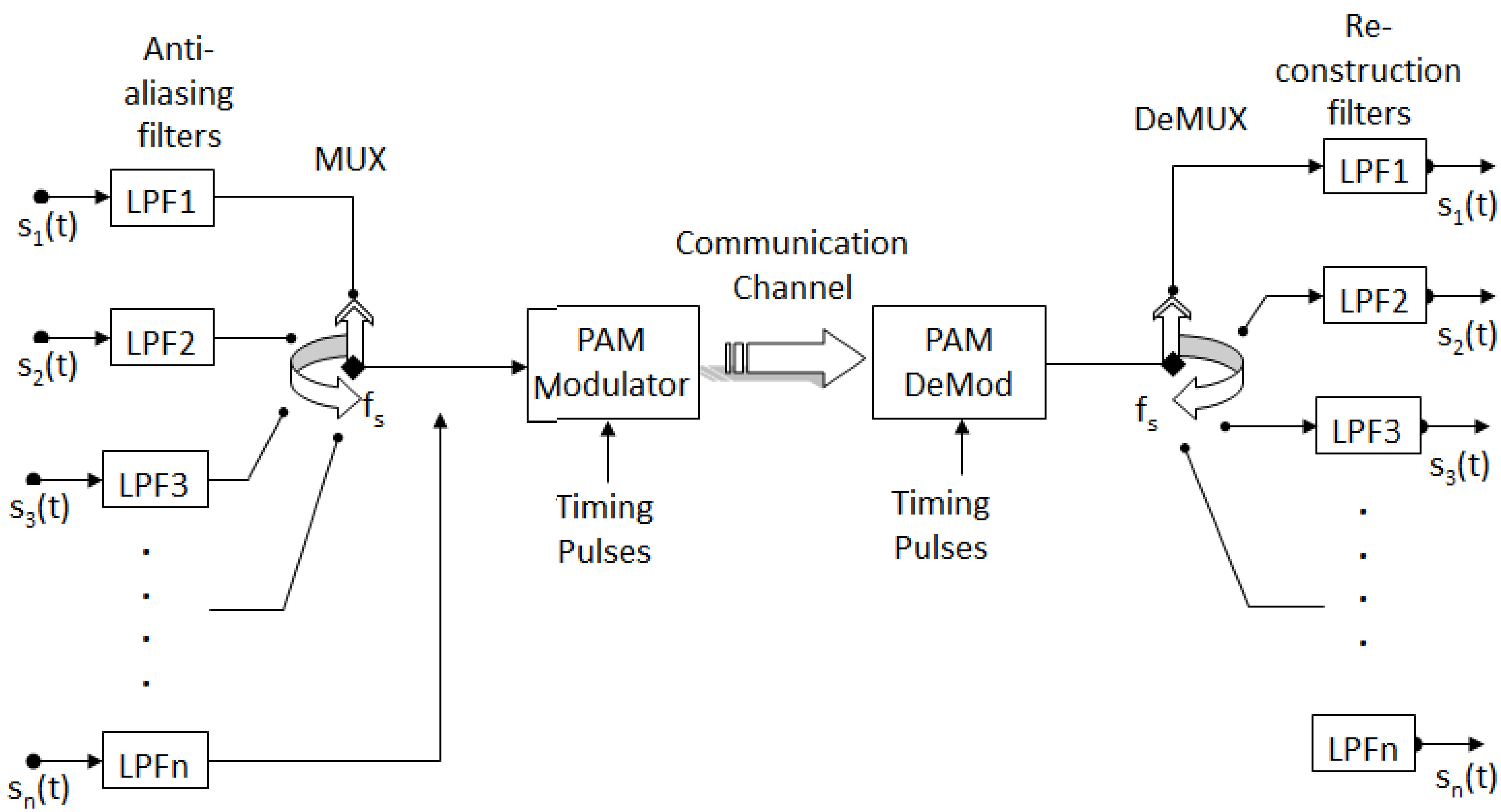


Figure 3.15 Line codes for the electrical representations of binary data.
(*a*) Unipolar NRZ signaling. (*b*) Polar NRZ signaling.
(c) Unipolar RZ signaling. (*d*) Bipolar RZ signaling.
(*e*) Split-phase or Manchester code.

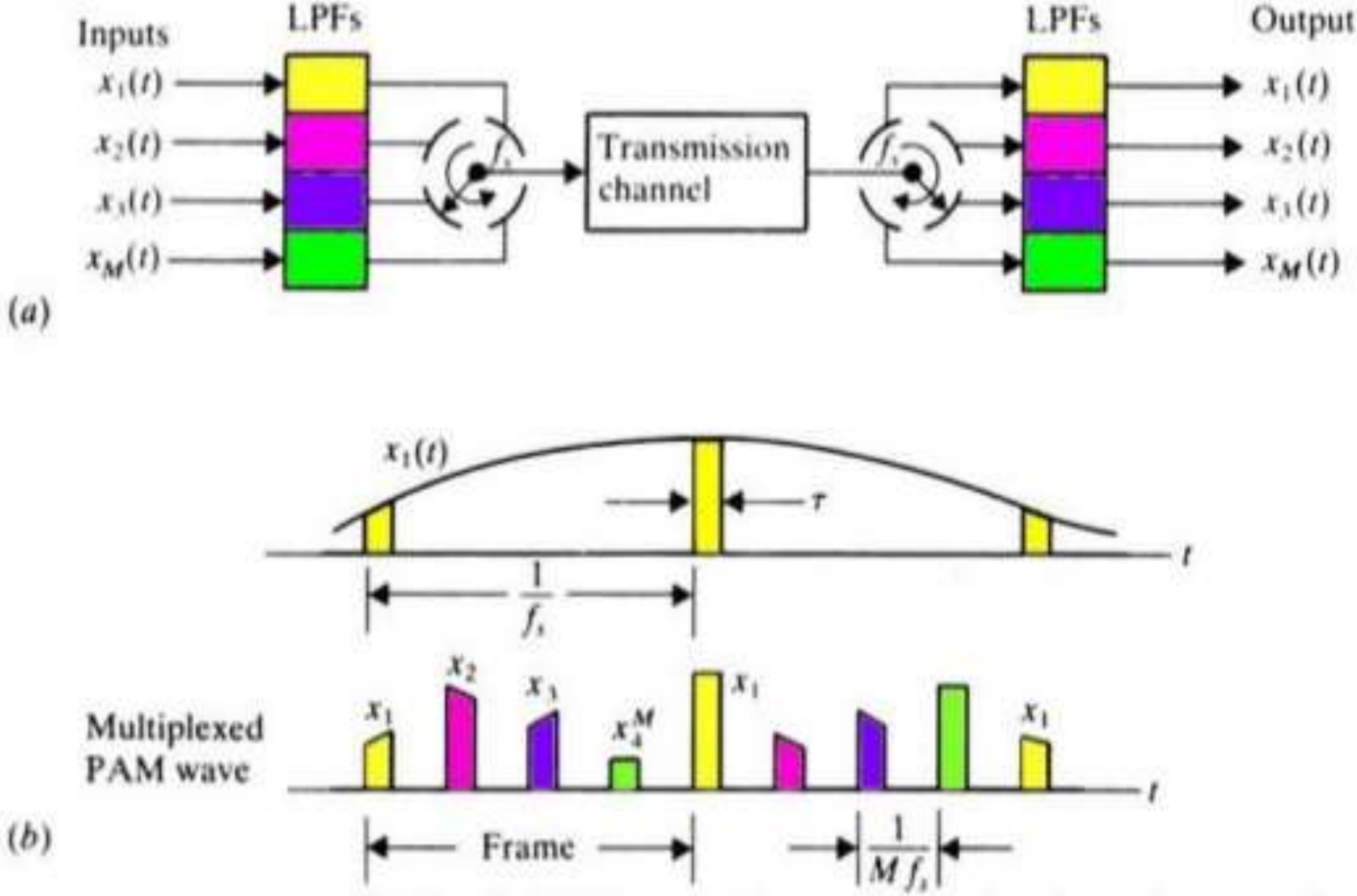
Time Division Multiplexing (TDM)

- An important feature of PAM is conservation of time. That is, for a given message signal, transmission of the associated PAM engages the communication channel for only a fraction of the sampling interval on a periodic basis.
- Hence, some of the time interval between adjacent pulses of the PAM wave is cleared for use by other independent message signals on a time shared basis.
- Joint utilization of a common channel by independent message signals without mutual interference.

TIME DIVISION MULTIPLEXING



Time Division Multiplexing (TDM)



Analog to Digital Conversion

- A digital signal is superior to an analog signal.
- Digital is less prone to noise and distortion.
- We can ’ t use analog signals for long distance (lose their strength, which means amplifiers are needed to amplify signal. However the amplifier creates distortion in the signal and adds some noise).
- The tendency today is to change an analog signal (such as audio ,voice and music) to digital data.
- **Pulse Code Modulation (PCM)** is a technique to convert analog data to digital signal.

Analog to Digital Conversion

- The idea of digitizing analog signal started with telephone companies, to provide long distance services; They digitized the analog signal at the sender; The signal is converted back to analog at the receiver.

- **Pulse Code Modulation (PCM):**

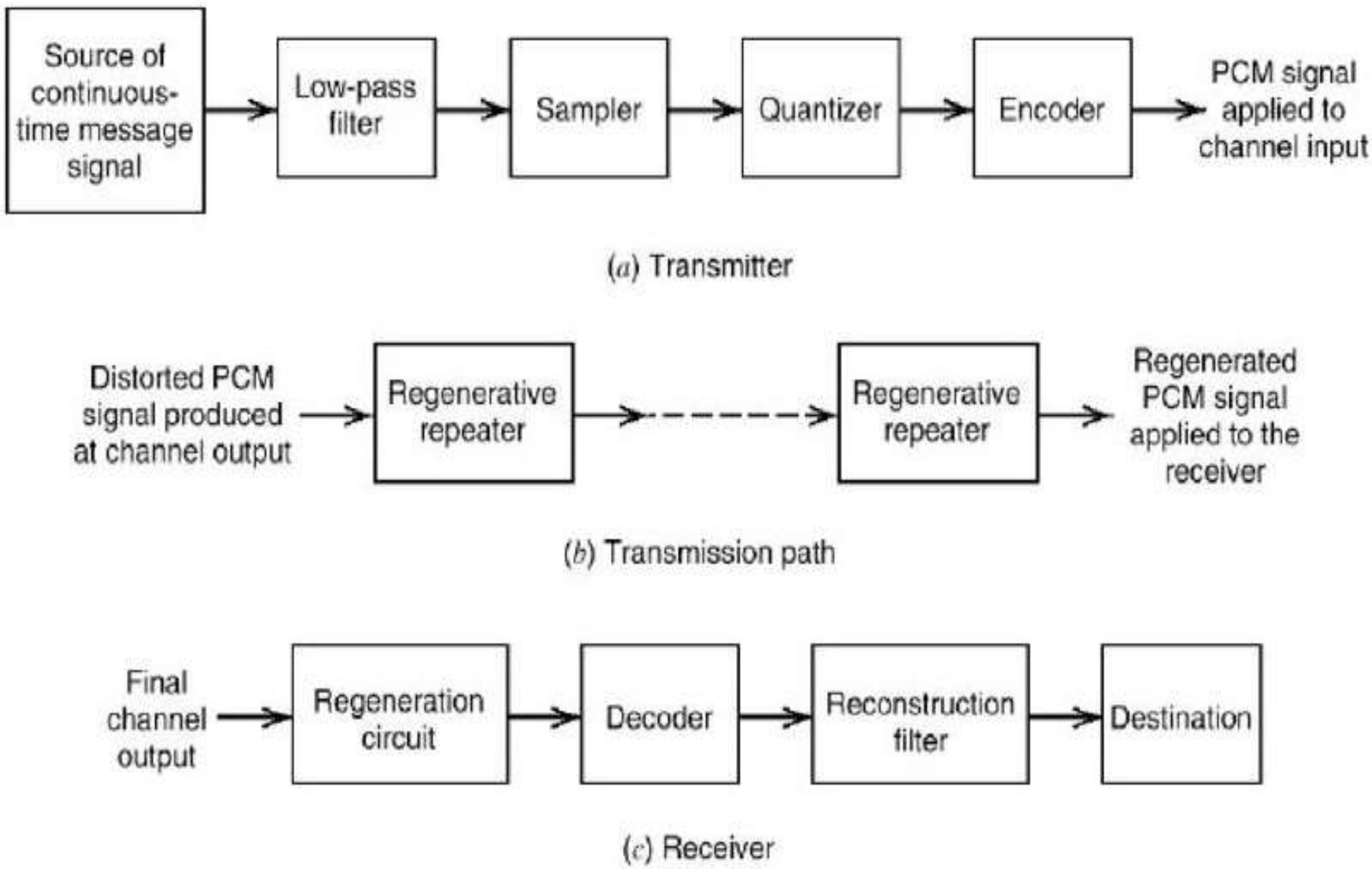
1- Sampling (PAM).

2- Quantization.

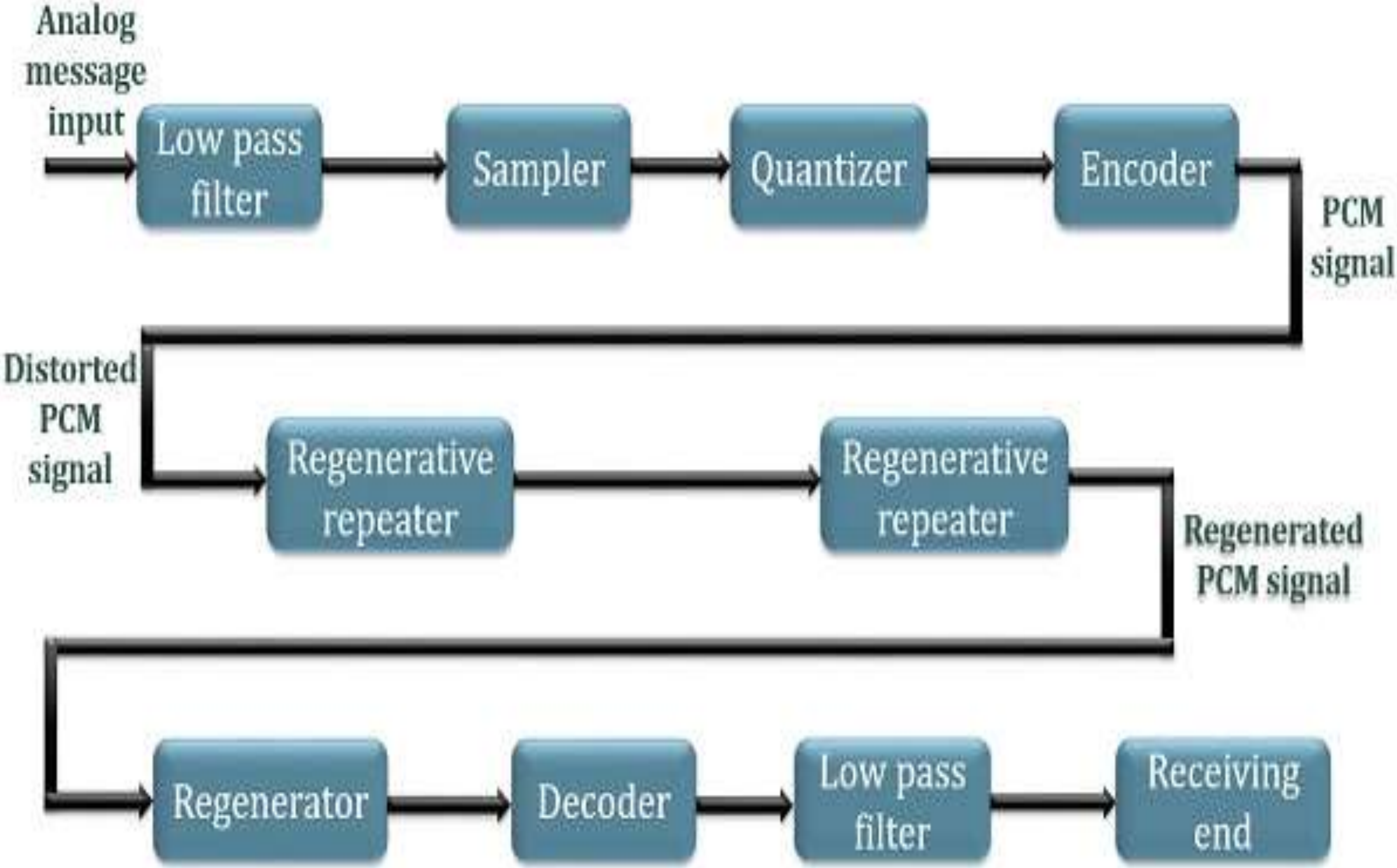
3- Binary encoding.

4- Line or block coding.

PCM BLOCK DIAGRAM



The basic elements of a PCM system



Block diagram of PCM system

Basic Elements of PCM

The transmitter section of a Pulse Code Modulator circuit consists of **Sampling, Quantizing** and **Encoding**, which are performed in the analog-to-digital converter section. The low pass filter prior to sampling prevents aliasing of the message signal.

The basic operations in the receiver section are **regeneration of impaired signals, decoding,** and **reconstruction** of the quantized pulse train. Following is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.

Low Pass Filter

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

This is the technique which helps to collect the sample data at instantaneous values of message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component **W** of the message signal, in accordance with the sampling theorem.

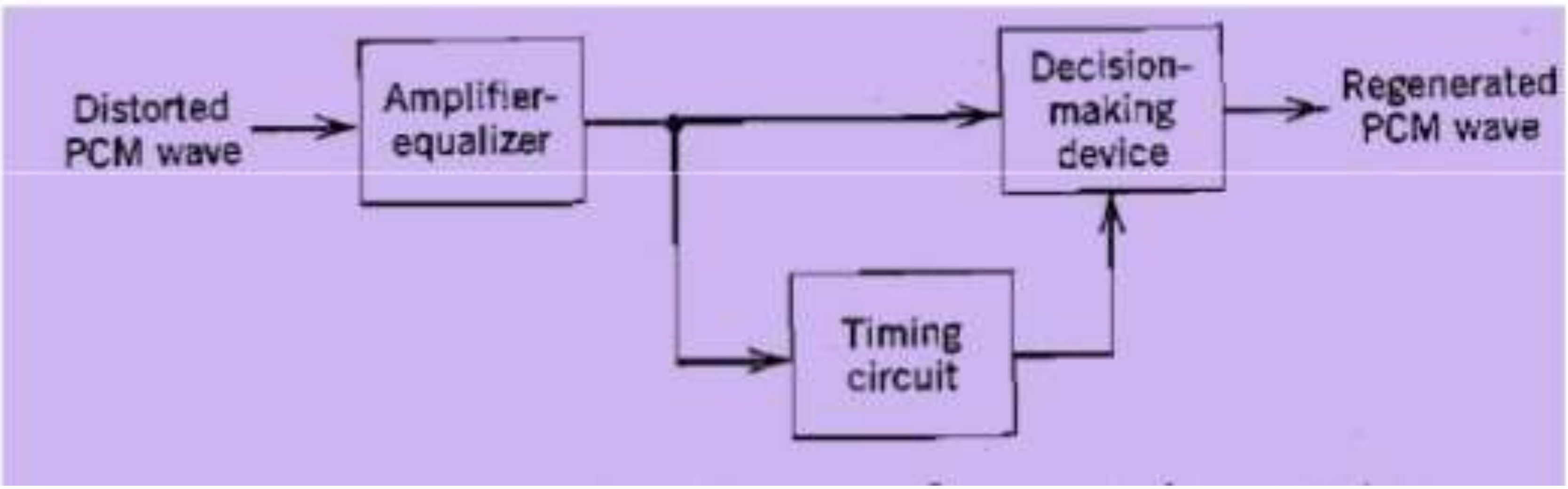
Quantizer

Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

Encoder

The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The sampling done here is the sample-and-hold process. These three sections LPF, Sampler and Quantizer will act as an analog to digital converter. Encoding minimizes the bandwidth used.

REGENERATION



REGENERATION

- A regenerative repeater consists of an equalizer, a timing circuit and a decision making device.
- The equalizer is used to undo the effect of the transmission channel to get back the pulses in their original shape before transmission. Amplitude and phase distortions caused by some characteristics of channel is removed and shapes the pulses
- Timing circuit is used to recover the clock of the transmitted symbols thus providing a periodic pulse train which is then used in the decision making process
- A decision making device is used to detect the different pulses based on the threshold information. Each sample is compared with the threshold value. If it exceeds a threshold value '1' is transmitted or else '0' is transmitted.

Regenerative Repeater

This section increases the signal strength. The output of the channel also has one regenerative repeater circuit, to compensate the signal loss and reconstruct the signal, and also to increase its strength.

Decoder

The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the demodulator.

Reconstruction Filter

After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low-pass filter is employed, called as the reconstruction filter to get back the original signal.

Hence, the Pulse Code Modulator circuit digitizes the given analog signal, codes it and samples it, and then transmits it in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

Advantages of PCM:

- *Relatively inexpensive*
- *Easily multiplexed*: PCM waveforms from different sources can be transmitted over a common digital channel (TDM)
- *Easily regenerated*: useful for long-distance communication, e.g. telephone
- Better noise performance than analog system
- Signals may be stored and time-scaled efficiently (e.g., satellite communication)
- Efficient codes are readily available

Disadvantage:

- Requires wider bandwidth than analog signals

Transmission Bandwidth

In binary PCM, we have a group of n bits corresponding to L levels with n bits.

Thus, L = 2n or n = log2 (L)

Signal m(t) is band-limited to B Hz which requires 2B samples per second.

For 2nB elements of information, we must transfer 2nB bits/second.

Thus, the minimum bandwidth BT needed to transmit 2nB bits/second is BT = nB Hz Practically speaking, usually we choose the transmission bandwidth to be a little higher than the minimum bandwidth required

Differential Pulse Code Modulation (DPCM)

- **D**ifferential **P**ulse Code **M**odulation (DPCM) is a procedure of converting an analog into a digital signal in which an analog signal is sampled and then the difference between the actual sample value and its predicted value (predicted value is based on previous sample or samples) is quantized and then encoded forming a digital value.

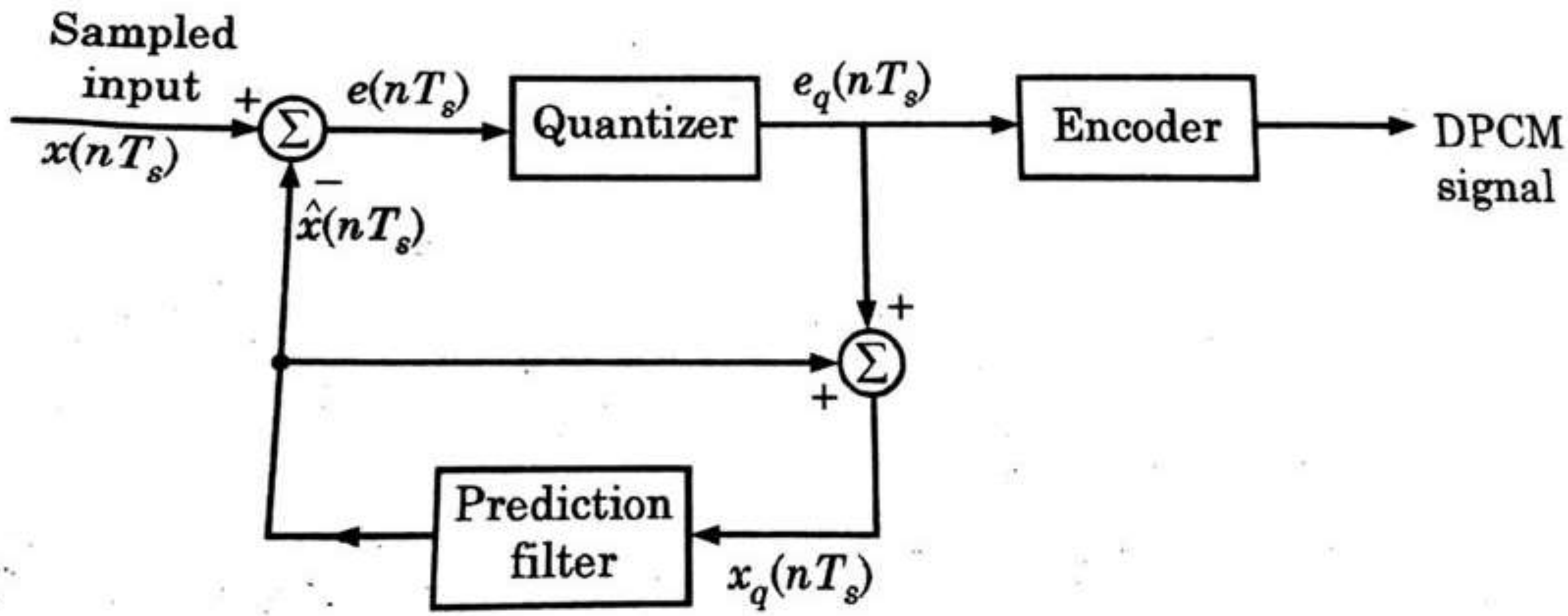
- DPCM code words represent differences between samples unlike PCM where code words represented a sample value.

Differential Pulse Code Modulation(DPCM)

Differential pulse-code modulation (DPCM) is a signal encoder that uses the baseline of pulse-code modulation (PCM) but adds some functionalities based on the prediction of the samples of the signal. The input can be an analog signal or a digital signal. •

If the input is a continuous-time analog signal, it needs to be sampled first so that a discrete time signal is the input of DPCM encoder

DPCM TRANSMITTER



The signals at each point are named as −

- $x(nT_s)$ is the sampled input

- $\hat{x}(nT_s)$ is the predicted sample

- $e(nT_s)$ is the difference of sampled input and predicted output, often called as prediction error

- $v(nT_s)$ is the quantized output

- $u(nT_s)$ is the predictor input which is actually the summer output of the predictor output and the quantizer output

The predictor produces the assumed samples from the previous outputs of the transmitter circuit. The input to this predictor is the quantized versions of the input signal $x(nT_s)$

Quantizer Output is represented as −

$$v(nT_s)=Q[e(nT_s)]=e(nT_s)+q(nT_s)$$

Where **$q\ (nT_s)$** is the quantization error

Predictor input is the sum of quantizer output and predictor output,

$$u(nT_s)=\hat{x}(nT_s)+v(nT_s)$$

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

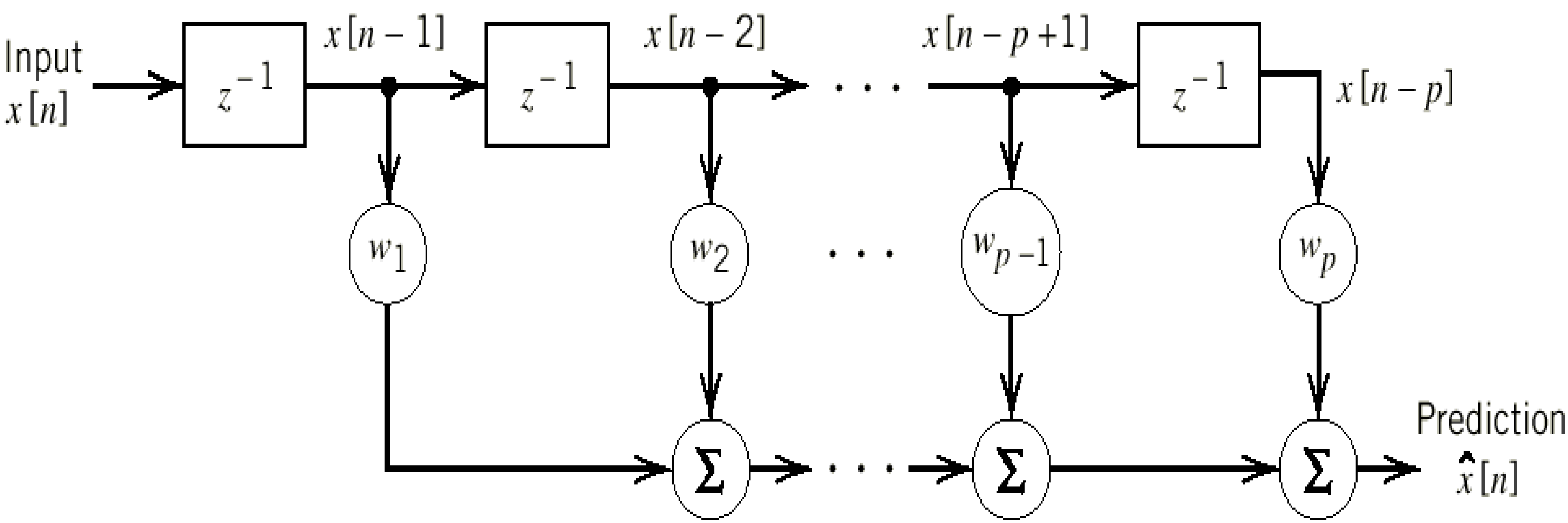
$$u(nT_s)=x(nT_s)+q(nT_s)u(nTs)$$

The same predictor circuit is used in the decoder to reconstruct the original input.

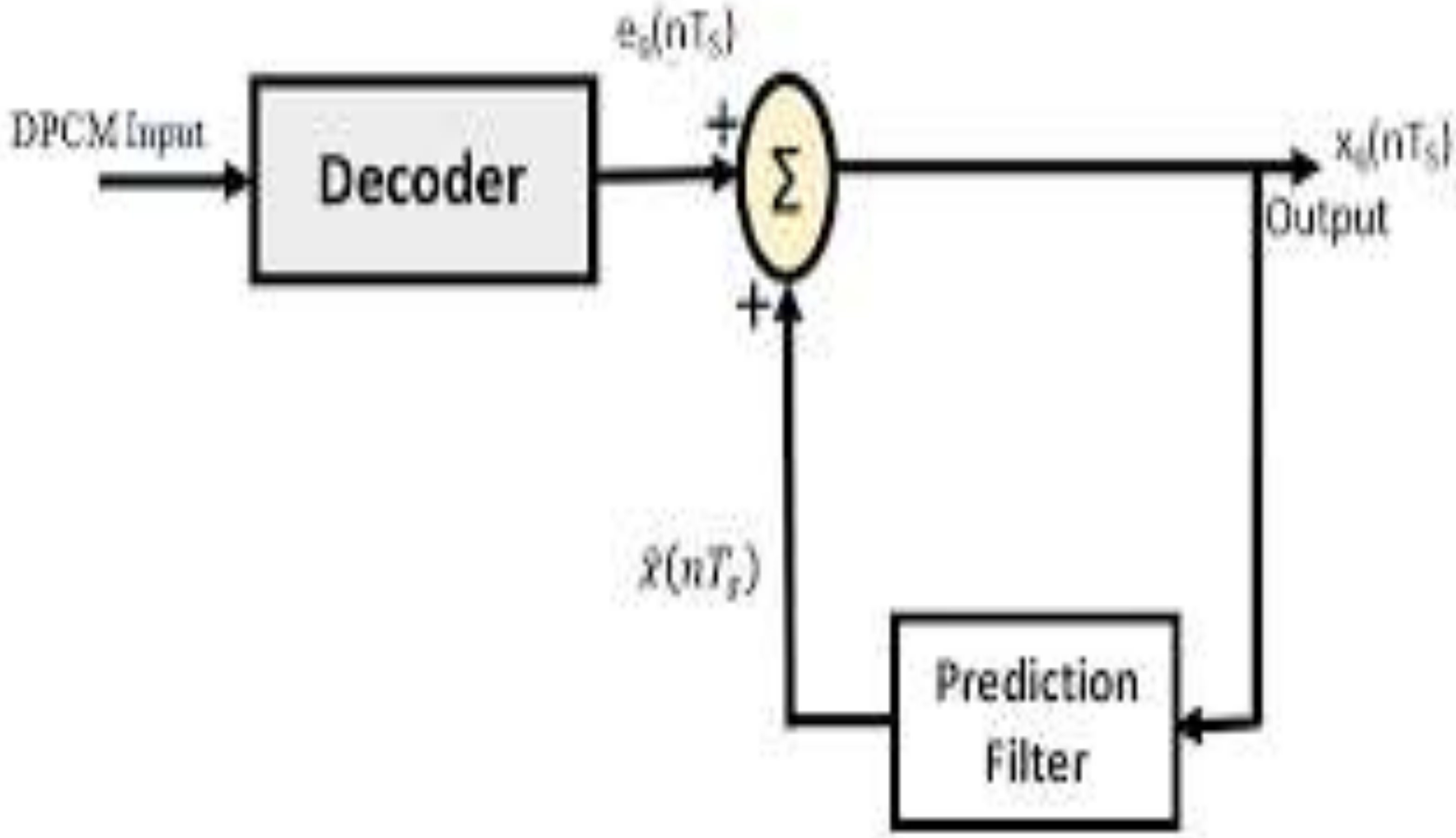
PREDICTION FILTER

Consider a finite-duration impulse response (FIR) discrete-time filter which consists of three blocks :

1. Set of p (***p*: prediction order**) unit-delay elements (z^{-1})
2. Set of multipliers with coefficients $w_1, w_2, \ldots w_p$
3. Set of adders ($\sum$)



DPCM DECODER



The notation of the signals is the same as the previous ones. In the absence of noise, the encoded receiver input will be the same as the encoded transmitter output.

As mentioned before, the predictor assumes a value, based on the previous outputs. The input given to the decoder is processed and that output is summed up with the output of the predictor, to obtain a better output.

The sampling rate of a signal should be higher than the Nyquist rate, to achieve better sampling. If this sampling interval in Differential PCM is reduced considerably, the sampleto-sample amplitude difference is very small, as if the difference is **1-bit quantization**, then the step-size will be very small i.e., **Δ** delta.

DELTA MODULATION

The type of modulation, where the sampling rate is much higher and in which the stepsize after quantization is of a smaller value **Δ**, such a modulation is termed as **delta modulation**.

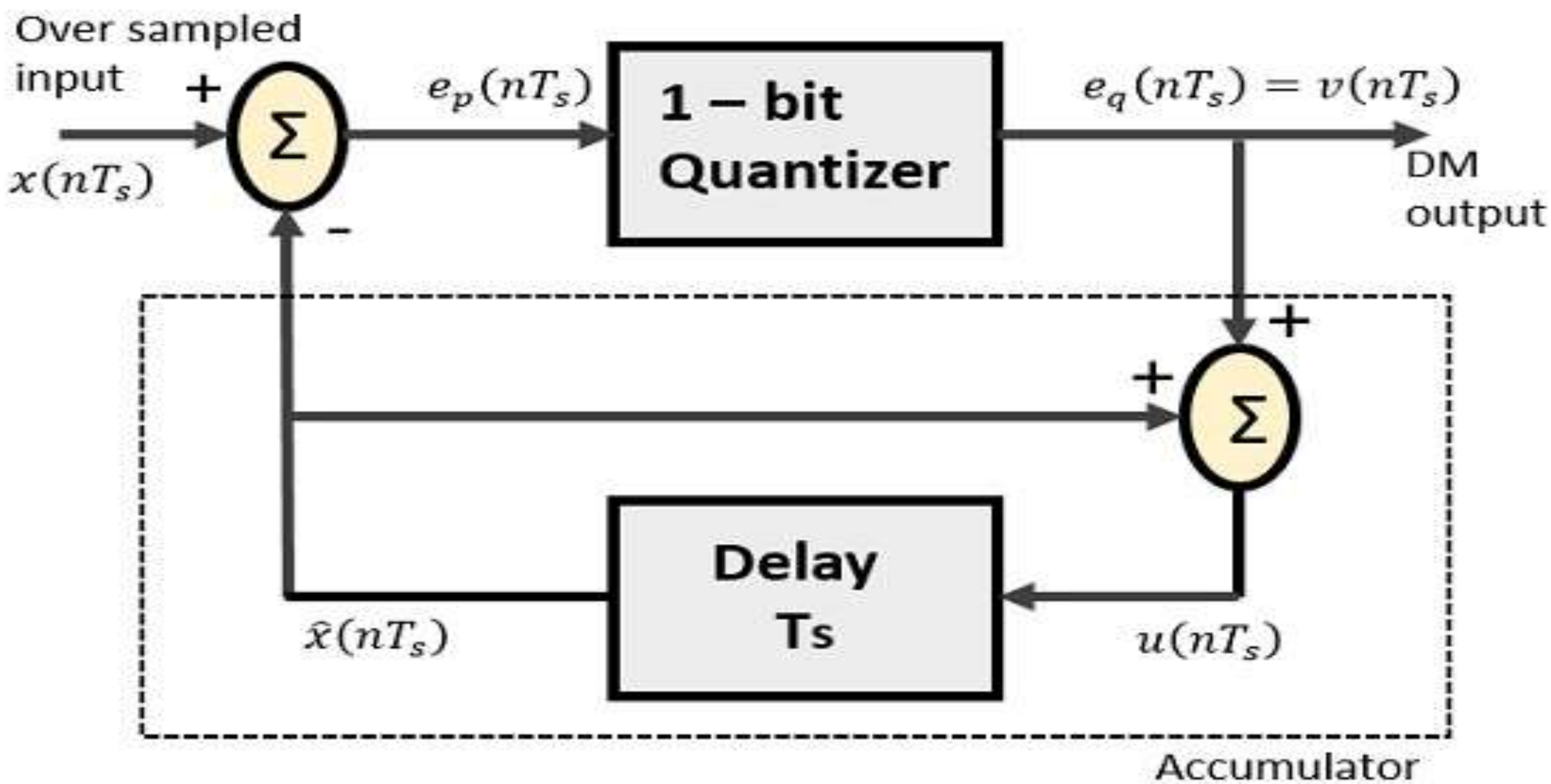
Features of Delta Modulation

1. An over-sampled input is taken to make full use of the signal correlation.
2. The quantization design is simple.
3. The input sequence is much higher than the Nyquist rate.
4. The quality is moderate.
5. The design of the modulator and the demodulator is simple.
6. The stair-case approximation of output waveform.
7. The step-size is very small, i.e., **Δ** deltadelta.
8. The bit rate can be decided by the user.
9. This involves simpler implementation.

Delta Modulation is a simplified form of DPCM technique, also viewed as **1-bit DPCM scheme**. As the sampling interval is reduced, the signal correlation will be higher.

Delta Modulator

The Delta Modulator comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.



The predictor circuit in DPCM is replaced by a simple delay circuit in DM.

From the above diagram, we have the notations as −

- $x(nT_s)$ = over sampled input

- $e_p(nT_s)$ = summer output and quantizer input

- $e_q(nT_s)$ = quantizer output = $v(nT_s)$v(nTs)

- $\hat{x}(nT_s)$ = output of delay circuit

- $u(nT_s)$ = input of delay circuit

Using these notations, now we shall try to figure out the process of delta modulation.

$$e_p(nT_s)=x(nT_s)-\hat{x}(nT_s) \text{ ---------}(1)$$

$e_p(nT_s) = x(nT_s) - u([n-1]T_s)$

$= x(nT_s) - [\hat{x}[[n-1]T_s] + v[[n-1]T_s]]$ ------------------------(2)

Further,

$v(nT_s) = e_q(nT_s) = S.sig.[e_p(nT_s)]$

$u(nT_s) = \hat{x}(nT_s) + e_q(nT_s)u(nTs)$ --------------------------------(3)

Where,

- $\hat{x}(nT_s)$ = the previous value of the delay circuit
- $e_q(nT_s)$ = quantizer output = $v(nT_s)$

Hence,

$u(nT_s) = u([n-1]T_s) + v(nT_s)$ ------------------------(4)

Which means,

The present input of the delay unit

= The previous output of the delay unit + the present quantizer output

Assuming zero condition of Accumulation,

$$u(nT_s)=S\sum_{j=1 \text{ to } n} sig[e_p(jT_s)]$$

Accumulated version of DM output =

$$\sum_{j=1 \text{ to } n} v(jT_s) \text{ ---------------------------------}(5)$$

Now, note that

$$\hat{x}(nT_s)=u([n-1]T_s)$$

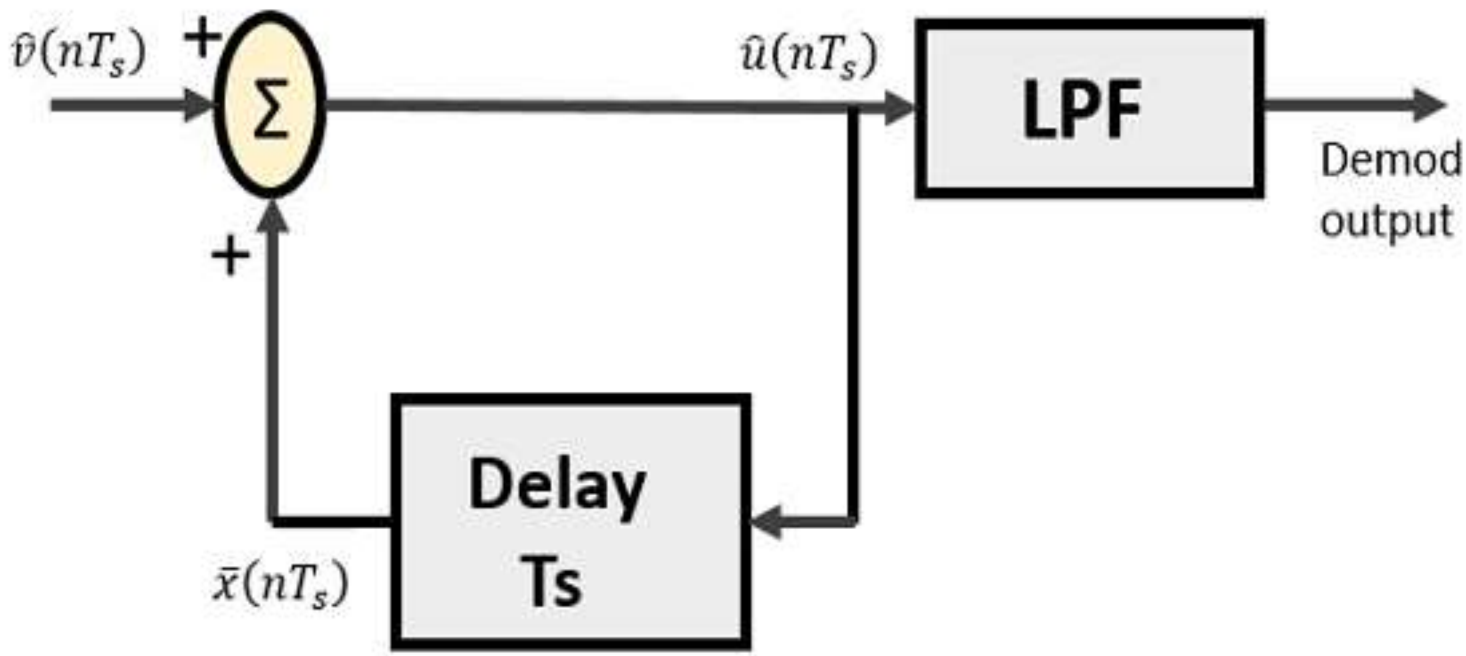
$$=\sum_{j=1 \text{ to } n-1} v(jT_s) \text{ -------------------------}(6)$$

Delay unit output is an Accumulator output lagging by one sample. From equations 5 & 6, we get a possible structure for the demodulator.

A Stair-case approximated waveform will be the output of the delta modulator with the step-size as delta (**Δ**). The output quality of the waveform is moderate.

Delta Demodulator

The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator. Following is the diagram for delta demodulator.



From the above diagram, we have the notations as –

- $\hat{v}(nT_s)$ is the input sample

- $\hat{u}(nT_s)$ is the summer output

- $\bar{x}(nT_s)$ is the delayed output

A binary sequence will be given as an input to the demodulator. The stair-case approximated output is given to the LPF.

Low pass filter is used for many reasons, but the prominent reason is noise elimination for out-of-band signals.

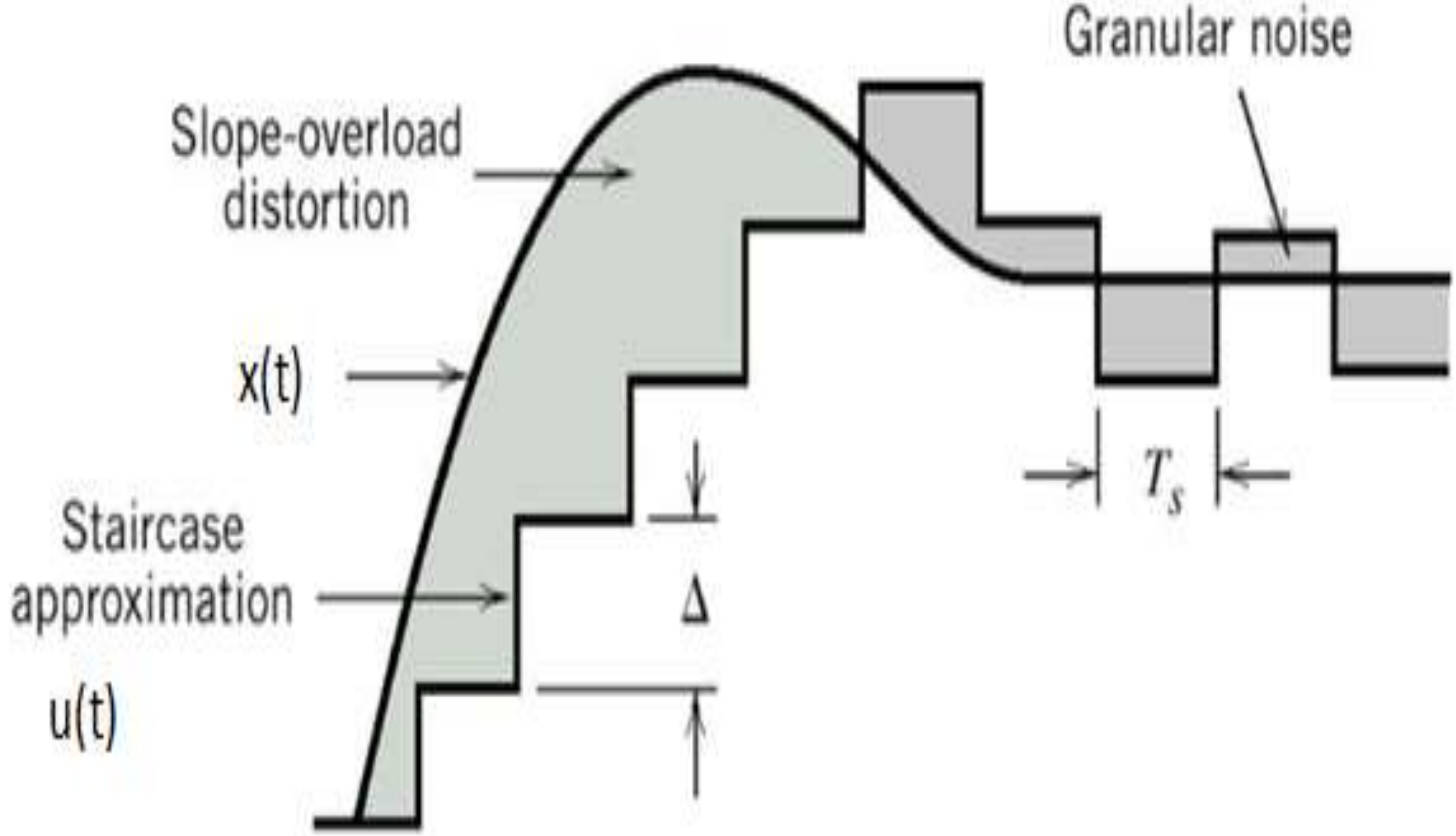
The step-size error that may occur at the transmitter is called **granular noise**, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

Advantages of DM Over DPCM

- 1-bit quantizer
- Very easy design of the modulator and the demodulator

Noise considerations in DM

- Slope Over load distortion (when **Δ** is small)
- Granular noise (when **Δ** is large)



Granular noise
Slope-overload
distortion
x(t)
T_s
Staircase
approximation
Δ
u(t)

Slope Overload Distortion

This distortion arises because of large dynamic range of the input signal.

The rate of rise of input signal x(t) is so high that the staircase signal can not approximate it, the step size ‘Δ’ becomes too small for staircase signal u(t) to follow the step segment of x(t).

Hence, there is a large error between the staircase approximated signal and the original input signal x(t). This error or noise is known as **slope overload distortion** .

To reduce this error, the step size must be increased when slope of signal x(t) is high.

Granular or Idle Noise

Granular or Idle noise occurs when the step size is too large compared to small variation in the input signal.

This means that for very small variations in the input signal, the staircase signal is changed by large amount (Δ) because of large step size.

Fig shows that when the input signal is almost flat , the staircase signal u(t) keeps on oscillating by $\pm\Delta$ around the signal.

The error between the input and approximated signal is called **granular noise**.

The solution to this problem is to make the step size small .

In order to overcome the quantization errors due to slope overload and granular noise, the step size (Δ) is made adaptive to variations in the input signal x(t).

Particularly in the steep segment of the signal x(t), the step size is increased. And the step is decreased when the input is varying slowly.

This method is known as Adaptive Delta Modulation (ADM).

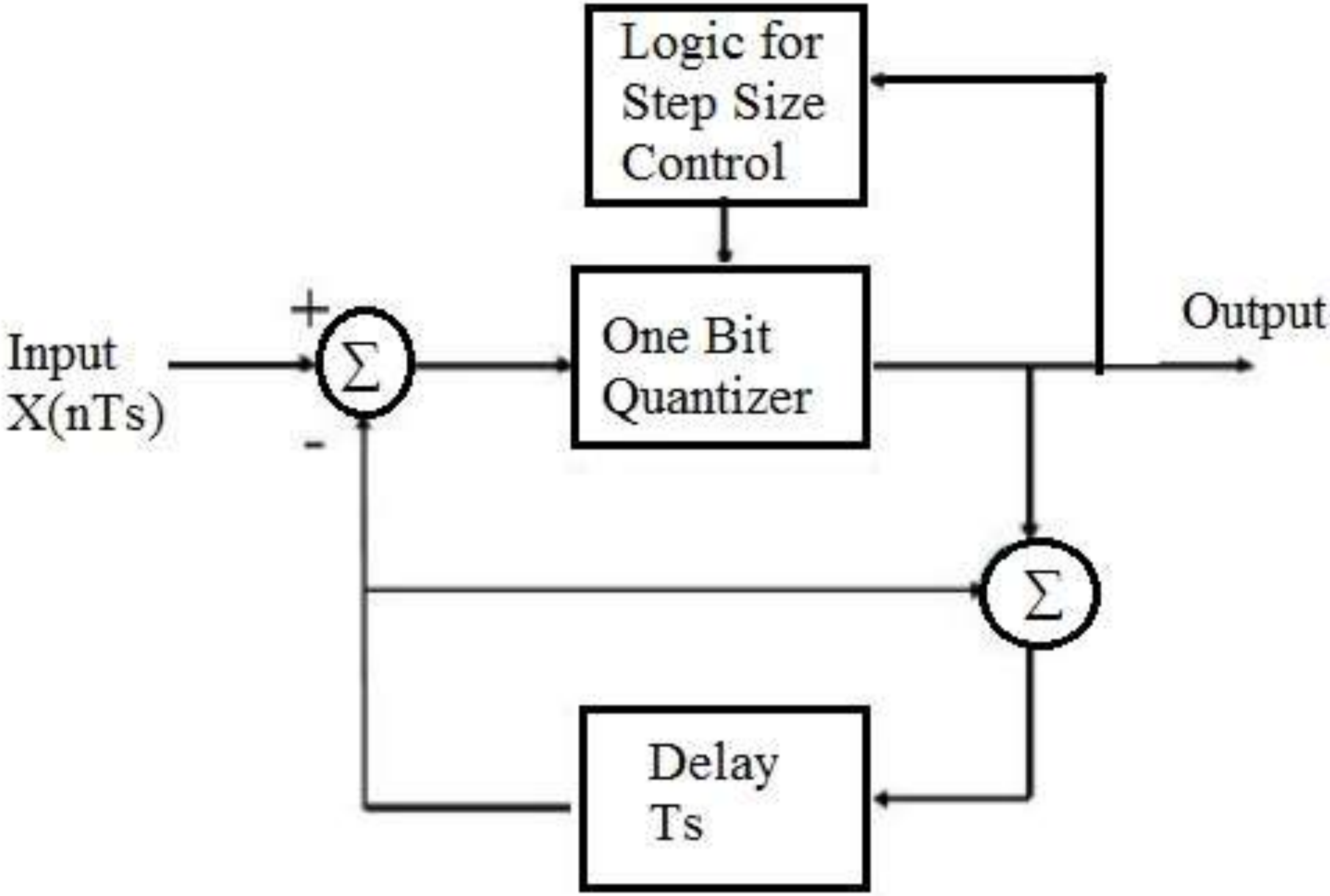
The adaptive delta modulators can take continuous changes in step size or discrete changes in step size.

Adaptive Delta Modulation

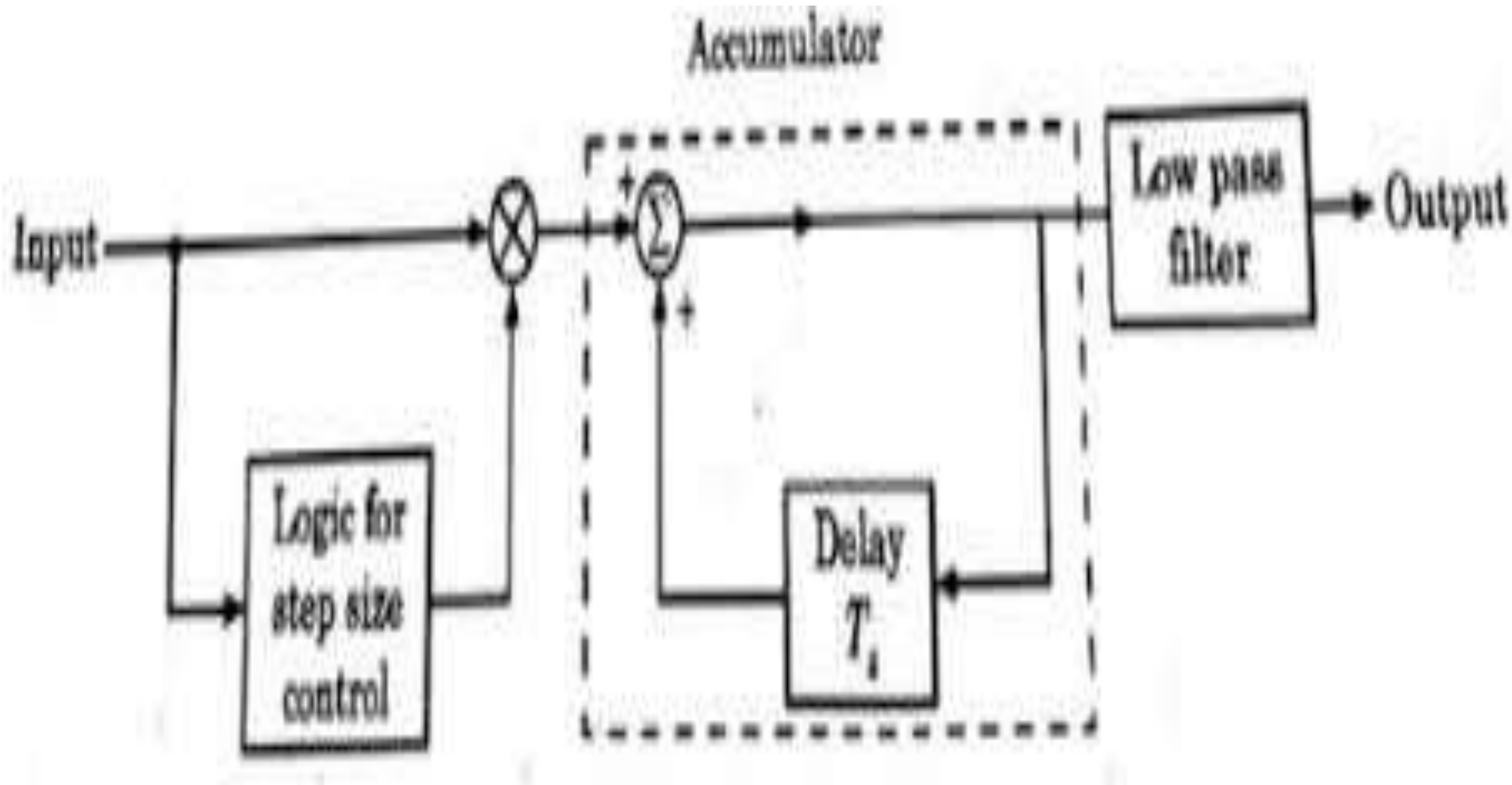
This Modulation is the refined form of delta modulation. This method was introduced to solve the granular noise and slope overload error caused during Delta modulation.

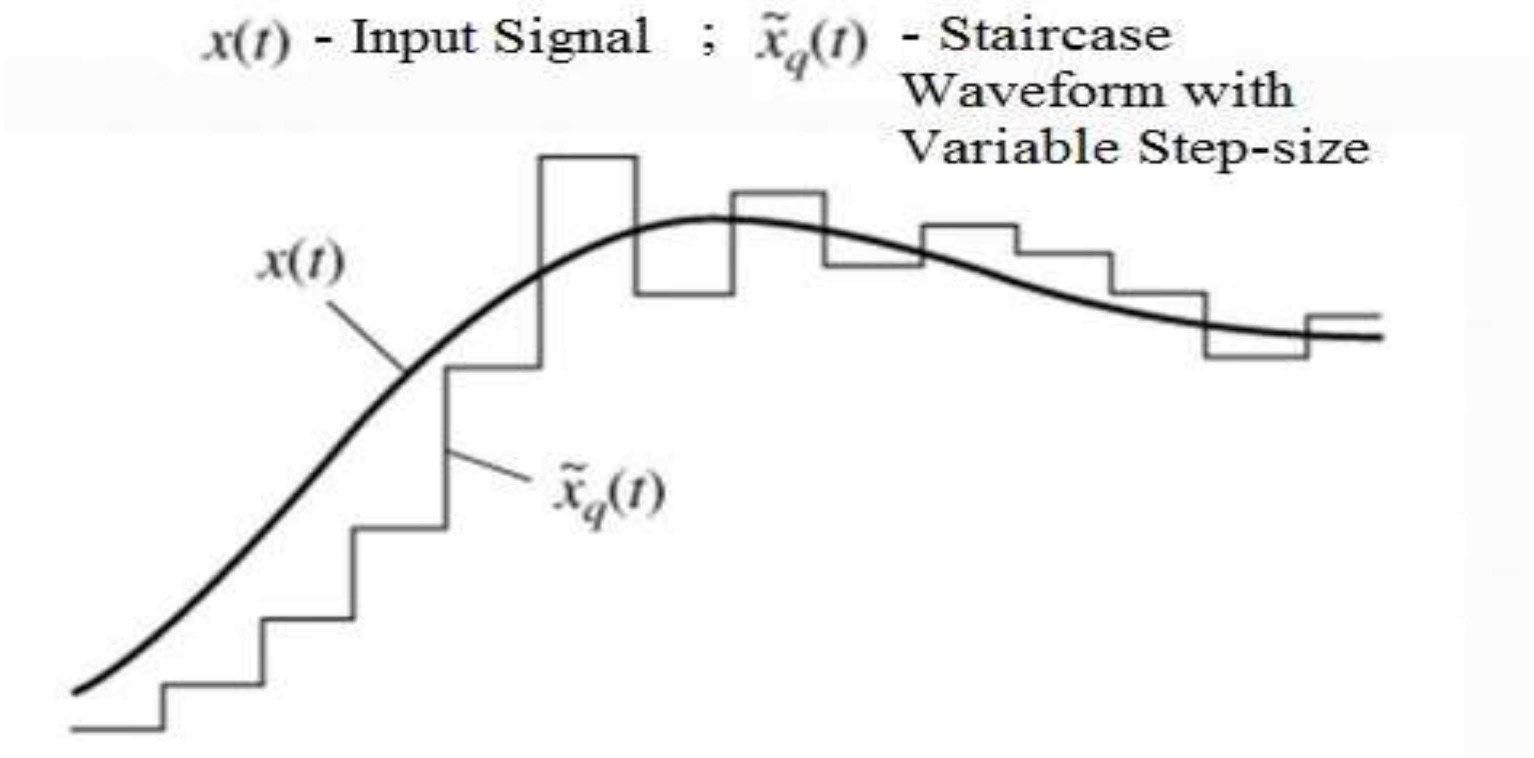
This Modulation method is similar to Delta modulation except that the step size is variable according to the input signal in Adaptive Delta Modulation whereas it is a fixed value in delta modulation.

ADM TRANSMITTER



ADM RECEIVER





At the logic step size control circuit, the output is decided based on the quantizer output. If the quantizer output is high, then the step size is doubled for the next sample. If the quantizer output is low, the step size is reduced by one step for the next sample.

Advantages

Some of the advantages ADM method are listed below-

- 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.

S.NO	Parameter of Comparison	Pulse Code Modulation (PCM)	Delta Modulation (DM)	Adaptive Delta Modulation (ADM)	Differential Pulse Code Modulation (DPCM)
1.	Number of bits	It can use 4,8, or 16 bits per sample.	It uses only one bit for one sample	It uses only one bit for one sample	Bits can be more than one but are less than PCM.
2.	Levels and step size	The number of levels depends on number of bits. Level size is fixed.	Step size is kept fixed and cannot be varied.	According to the signal variation, step size varies.	Number of levels is fixed.
3.	Quantization error and distortion	Quantization error depends on number of levels used.	Slope overload distortion and granular noise are present.	Quantization noise is present but other errors are absent.	Slope overload distortion and quantization noise is present.
4.	Transmission bandwidth	Highest bandwidth is required since numbers of bits are high.	Lowest bandwidth is required.	Lowest bandwidth is required.	Bandwidth required is less than PCM.
5.	Feedback	There is no feedback in transmitter or receiver.	Feedback exists in transmitter.	Feedback exists.	Feedback exists.
6.	Complexity of Implementation	System is complex.	Simple	Simple	Simple

Processing Gain

- Output signal-to-noise ratio (SNR_O)

$$(SNR)_O = \frac{\sigma_M^2}{\sigma_Q^2} \quad (3.79)$$

- σ_M^2 – variance of m[n]
- σ_Q^2 – variance of quantization error q[n]
- rewrite using variance of the prediction error σ_E^2

$$(SNR)_O = \left(\frac{\sigma_M^2}{\sigma_E^2}\right)\left(\frac{\sigma_E^2}{\sigma_Q^2}\right) = G_p (SNR)_Q$$

(3.80)

$$(SNR)_Q = \frac{\sigma_E^2}{\sigma_Q^2} \quad (3.81)$$

signal-to-quantization noise ratio

$$G_p = \frac{\sigma_M^2}{\sigma_E^2} \quad (3.82)$$

processing gain

ADPCM (Adaptive DPCM)

DPCM coder has two components, namely the quantizer and the predictor. In Adaptive Differential Pulse Code Modulation (ADPCM), the quantizer and predictor are adaptive. This means that they change to match the characteristics of the speech (or audio) being coded.

ADPCM adapts the quantizer step size to suit the input. In DPCM, step size can be changed along with decision boundaries, using a non uniform quantizer. There are two ways to do this

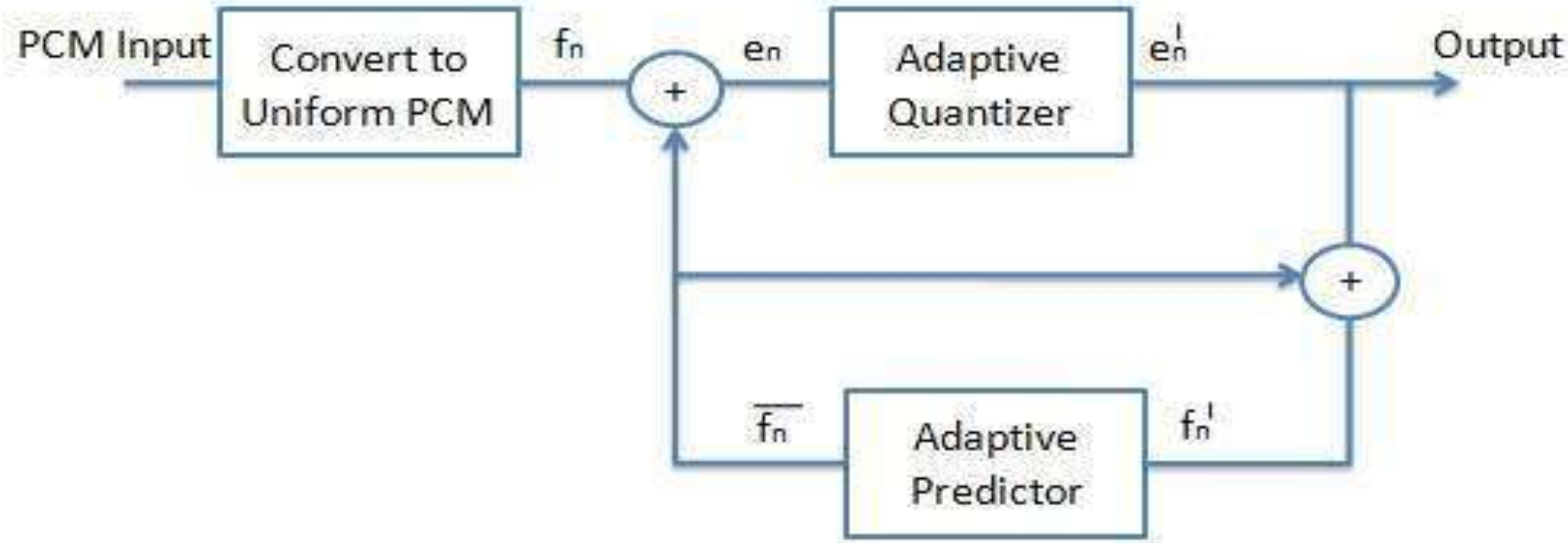
•**Forward Adaptive Quantization** - Properties of the input signal are used.

•**Backward Adaptive Quantizationor** - Properties of quantized output are used. If errors become too large, choose non-uniform quantizer.

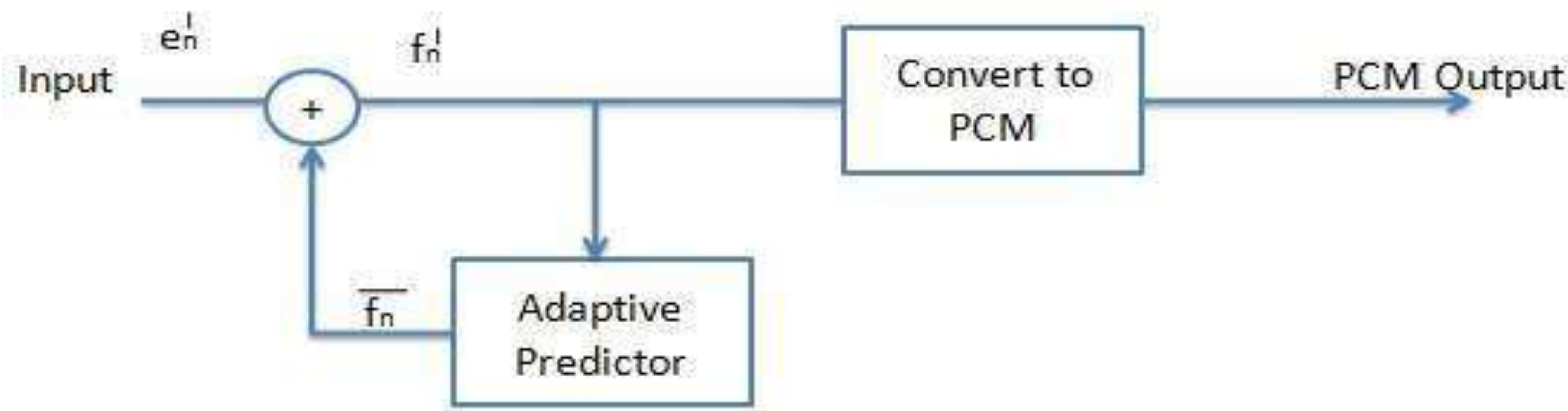
Choose the predictor with forward or backward adaptation to make predictor coefficients adaptive which is also known as Adaptive Predictive Coding (APC). As you know that the predictor is usually a linear function of previous reconstructed quantized value f'_n. The number of previous values used is called the order of the predictor. For example, if we use X previous values, we need X coefficients a_i, i=1, 2,........, X in predictor. Therefore,

$$\bar{f}_n = \text{function of } (f'_{n-1}, f'_{n-2}, - - - - - -)$$

However, we find a different situation, if we try to change the prediction coefficients which multiply previous quantized values to make a complicated set of equations to solve for these cofficients. Figure shows a schematic diagram for the ADPCM encoder and decoder.



Encoder



Decoder