BLOCK DIAGRAM OF : Digital Communication System

Fig (1.1) shows the block diagram of digital communication system.

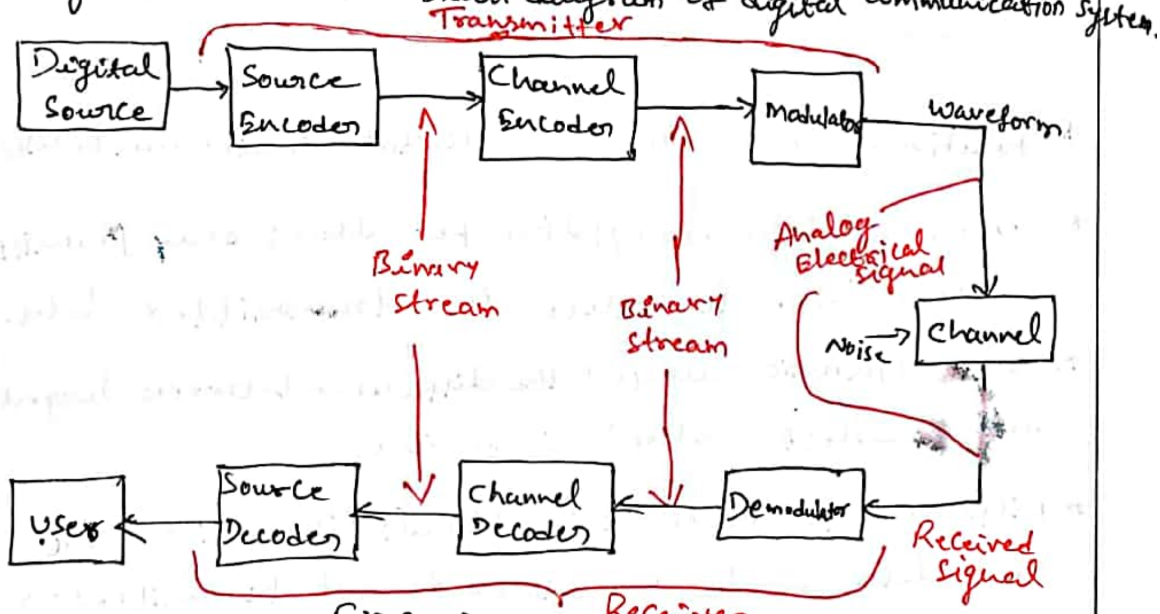


Fig (1.1)

In digital communication, the main three basic processing operations are identified : Source Coding, channel coding and modulation. It is assumed that the source of information is digital by nature or converted into digital by design. The output of digital source is binary bits.

Source Encoder : It maps the digital signal generated by the source into another signal in digital form. The source encoder removes redundant information/bits in the message signal and is responsible for efficient use of channel bandwidth.

Channel Encoder : The binary stream at the output of source encoder is next processed by channel encoder, which adds redundant ~~informat~~ bits ~~to the output of~~ and produces a new binary stream. The redundant bits are added ...

error Control coading technique such that the effect of channel noise is minimized.

Modulator: The output binary stream of channel encoder is converted into the analog waveform by the modulator for efficient transmission of the signal over the channel. The modulation techniques for doing so are refferred to as amplitude-shift-keying (ASK), frequency shift keying (FSK) or phase-shift keying (PSK).

Channel: The connection between transmitter and receiver is established through communication channel. The channel can be wirelines, wireless or fiber optic. The common problems introduced in the channel are additive noise interference, signal attenuation, amplitude and phase distortion, multipath distortion.

- Power and bandwidth are the two important parameters of a channel.

Demodulator: It performs the reverse operation of Modulator. ~~The~~ By using suitable demodulating technique the received analog signal is converted into binary stream.

Channel Decoder: ~~Using the~~ By exploiting the strategy used ~~whi~~ at the channel encoder ~~to~~ in adding redundant bits, the channel decoder reproduces the original bit stream that was available at source encoder by predicting and ~~elect~~ ~~eliminating~~ correcting the errors that might have occured in the channel during the travel.

Source decoder: Source decoder performs reverse operation of channel encoder and ~~pro~~ recollects by digital source at the transmitter end.

Transmission Media for Digital Communication:

The details of modulation and coding used in a digital communication system depend on the characteristics of the channel and the application of interest.

The two channel characteristics "Bandwidth" and "Power" constitute primary resources available in the channel. Other characteristics :- Amplitude & phase resp.

- Linear or non-linear
- How free from interference

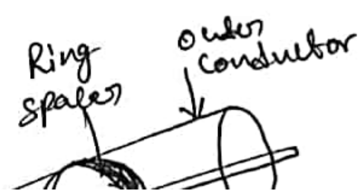
* Telephone Channel:

- A telephone channel is designed to provide voice-grade communication, also used for long distance data communication.

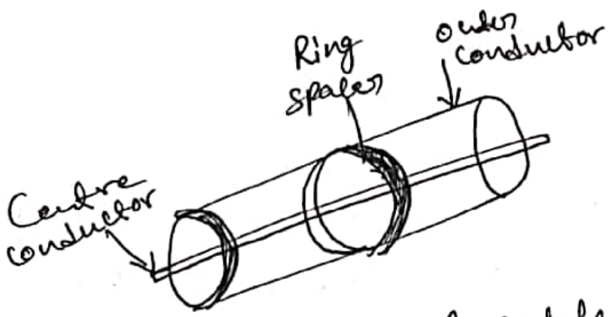
* Carries the signals of frequencies range 300 to 3400 Hz.
* High signal to noise (SNR) ratio of about 30dB.
* Approximately linear response.
* Transmission rates upto 16.8 kbps

* Coaxial Cable:

- Consists of single wire conductor centered inside an outer conductor which are insulated from each other by dielectric material.



other by dielectric material.



Co-axial cable.

* Advantages
  - Relatively wide bandwidth
  - Free from external interference.
* Disadvantage
  - Requires closely spaced repeaters.
* Data rate of 274 Mbps with repeaters spaced at 1 km intervals.

Optical fiber :

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SAMPLING

First step in Analog to Digital conversion

Sampling converts a ...

SAMPLING

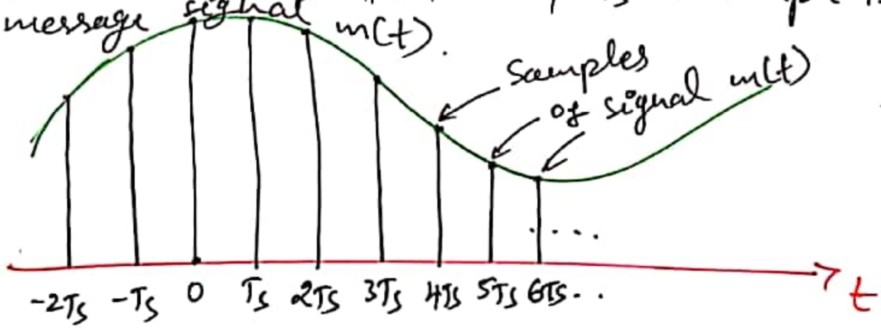
First step in Analog to Digital conversion

Sampling converts a continuous time signal to discrete time signal.

Extracting one sample every Ts.

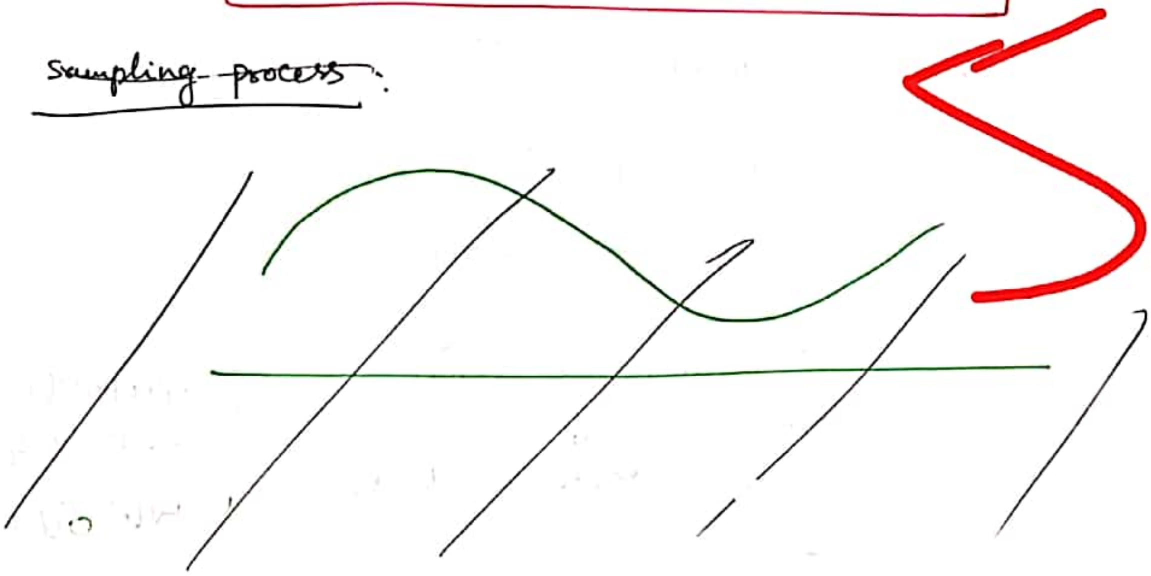
$\boxed{T_s \rightarrow \text{Sampling interval}}$

It is a periodic process, after every T_s a sample is extracted from the message signal m(t).



$$\boxed{\text{Sampling frequency} = f_s = \frac{1}{T_s}}$$

Sampling process:



Proof of Sampling Theorem:

Time domain Analysis:

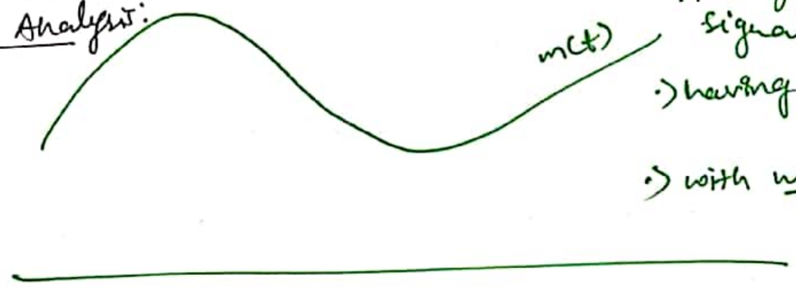
Multiplying |

Proof of Sampling Theorem:

Time domain Analysis:

Multipling m(t) with Impulse train

Ideal sampling or Impulse sampling



Original Analog signal (Bandlimited)
- having finite energy
- with max frequency f_m.

X. (product)

Impulse train (periodic signal of impulses with period T_s), Impulses at ..., $-2T_s, -T_s, 0, T_s, 2T_s$, ...

Impulse train:

$$\cdots + \delta(t+2T_s) + \delta(t+T_s) + \delta(t) + \delta(t-T_s) + \delta(t-2T_s) + \cdots$$

Impulse shifted by T_s

$$g_\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$$

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Sampled Signal $(m_\delta(t))$

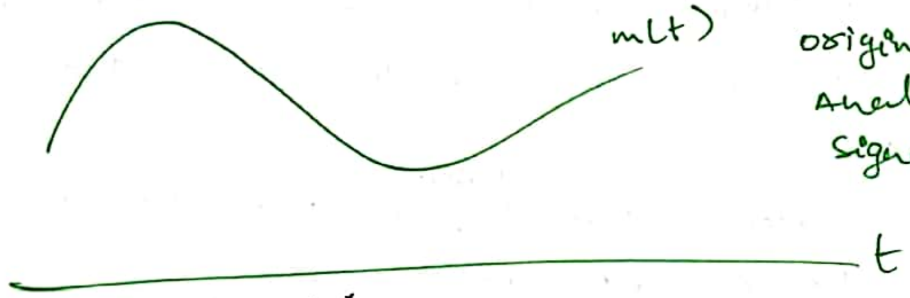
$$= m(t) \times g_\delta(t)$$

$$= m(t) \times \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$$

$$= \sum_{n=-\infty}^{\infty} \underbrace{m(t)\,\delta(t-nT_s)}_{m(nT_s)\,\delta(t-nT_s)}$$

[$\because$ Property: $m(t)\delta(t-t_0) = m(t_0)\delta(t-t_0)$]

$$m_\delta(t) = \sum_{n=-\infty}^{\infty} m(nT_s)\,\delta(t-nT_s)$$

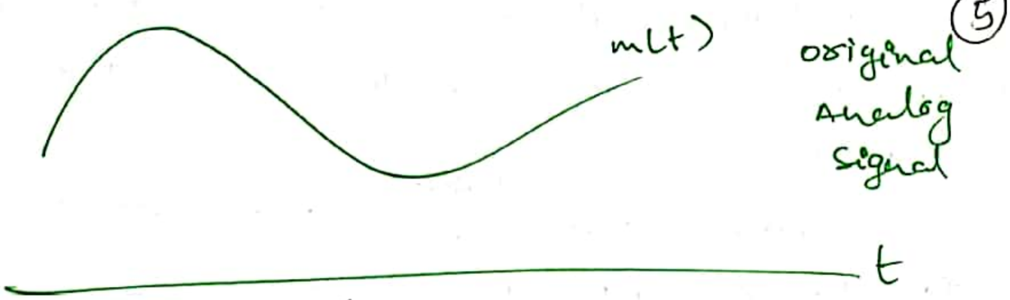


original Analog signal

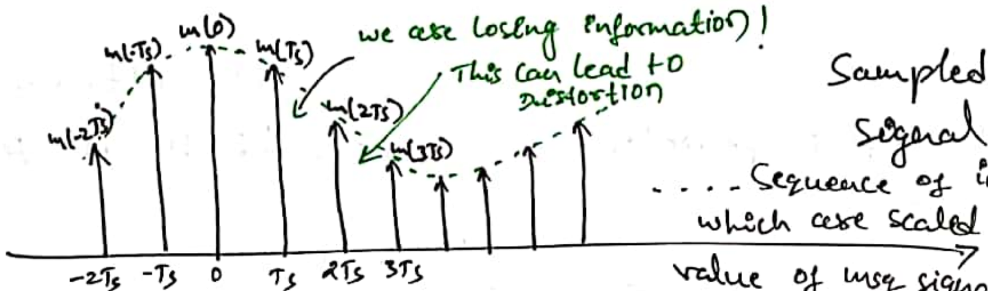
(5)

X

$\delta(t-nT_s)$ Train of



$\times$



.... Sequence of impulses which are scaled by the value of msg signal at each sampling instant

$$\boxed{m_\delta(t) = \sum_{n=-\infty}^{\infty} m(nT_s)\,\delta(t - nT_s)}$$

Frequency Domain Analysis:

Taking Fourier Transform of Sampled Signal.

$$\boxed{m_\delta(t) \xrightarrow{F.T} M_\delta(F)}$$

Here idea is to analyze how $M_\delta(F)$ is related to $M(F)$

$M(F)$ = spectrum of original message signal $m(t)$

$$\boxed{m(t) \xrightarrow{F.T} M(F)}$$

WKT $m(t)$ is sampled with impulse train with sampling frequency F_s, sampling interval $= \frac{1}{F_s} = T_s$.

NOTE: In sampling process, we are losing the information between 0 to T_s, T_s to $2T_s$, $2T_s$ to $3T_s$, a..... This can lead to distortion.

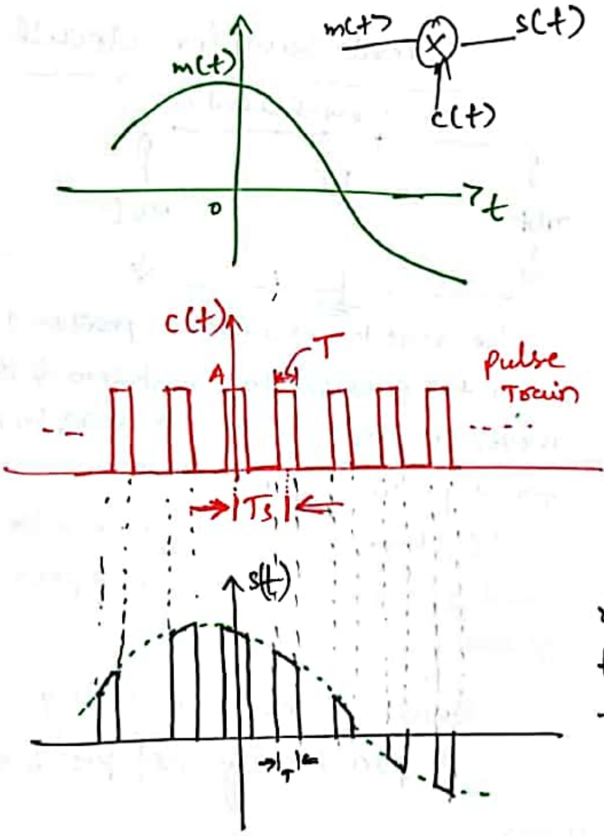
Therefore in order to minimize/avoid the distortion, how to fix the parameters of sampling?

Scanned with CamScanner

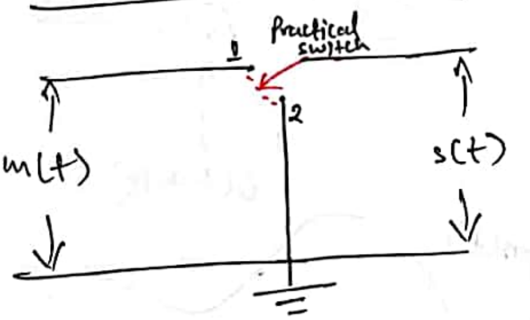
2. Natural Sampling:

Natural Sampling is also called as chopper Sampling. In natural sampling, the pulse train with pulse width 'T' is used. The Sampled Signal appears to be chopped off form of original signal, hence natural sampling is also called as chopper Sampling.

Let the message signal $m(t)$ be applied to a switching circuit shown in fig. The switching is Controlled by a Sampling function $c(t)$ that consists of ~~an~~ train of pulses with ~~ap~~ duration T and amplitude A and occuring ~~at~~ with period T_s. The output is denoted by $s(t)$.



Natural Sampling Circuit



The switch remains in position for a brief interval of time 'T' and change to position 2 and remains at 2 for the remaining time of each sampling period 'T_s'. The switch changes its position at a rate $F_s = \frac{1}{T_s}$.

and $F_s \geq 2F_m$

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The output $s(t)$ is expressed as

$$s(t) = m(t)\,c(t) \longrightarrow (1)$$

Similar to impulse train, the rectangular pulse train $c(t)$ has a Complex Fourier Series representation given by

$$c(t) = F_s T A \sum^{\infty} \operatorname{sinc}(kF_sT)\, e^{j2\pi kF_st} \longrightarrow (2)$$

Substituting ② in ①

$$s(t) = F_s T A \sum_{K=-\infty}^{\infty} \text{sinc}(KF_sT)\, m(t)\, e^{j2\pi KF_s t}$$

$$\boxed{\begin{array}{c}\text{Property (Frequency shift property)} \\ x(t)\, e^{j2\pi\beta t} \xleftrightarrow{F.T} X(F-\beta)\end{array}}$$

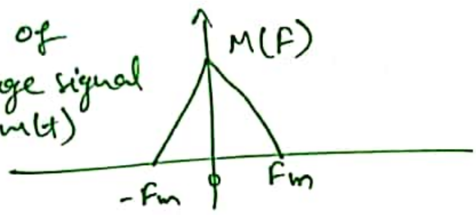
Taking F.T.

$$\boxed{S(F) = F_s T A \sum_{K=-\infty}^{\infty} \text{sinc}(KF_sT)\, M(F-KF_s)} \rightarrow ③$$

Equation ③ represents the relation between the spectra $M(F)$ and $S(F)$ as illustrated in fig.

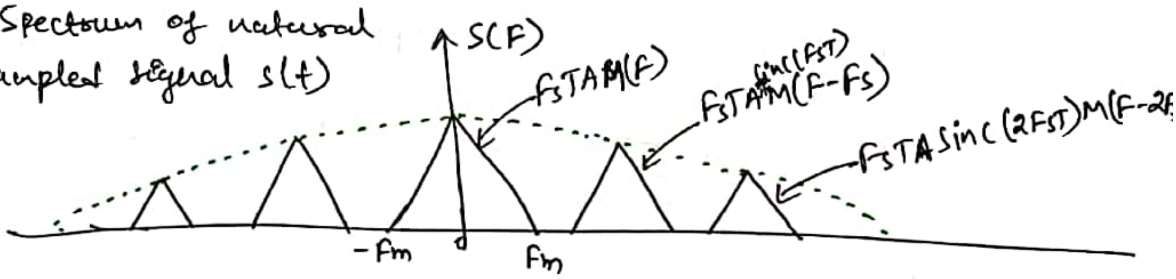
We observe that the effect of the finite duration sampling pulses (pulse train) is to multiply K^{th} lobe of the spectrum $S(F)$ by $\frac{TA}{T_s}\,\text{sinc}(KF_sT)$.

Spectrum of message signal m(t)



for sampling frequency $F_s > 2F_m$

Spectrum of natural sampled signal s(t)



The message signal m(t) can be recovered from s(t) (i.e. M(F) can be separated from S(F)) by passing s(t) through an ideal low-pass filter with bandwidth B such that $F_m < B < F_s - F_m$.

3. Flat-top Sampling:

In Flat-top sampling, the analog signal m(t) is sampled instantaneously at the rate $F_s = \frac{1}{T_s}$ ~~width~~ by sampling function having regular pulses of amplitude $m(nT_s)$ and width equal to 'T'.

From Ideal Sampling we have

$$m_\delta(t) = \sum_{n=-\infty}^{\infty} m(nT_s)\,\delta(t-nT_s) \longrightarrow (1)$$

In Flat-top sampling train of impulses is replaced by a train of ~~regul~~ rectangular pulses having width 'T'.

$\therefore$ Denoting s(t) as the output of Flat-top sampler,

$$s(t) = \sum_{n=-\infty}^{\infty} m(nT_s)\, h(t-nT_s) \longrightarrow (2)$$

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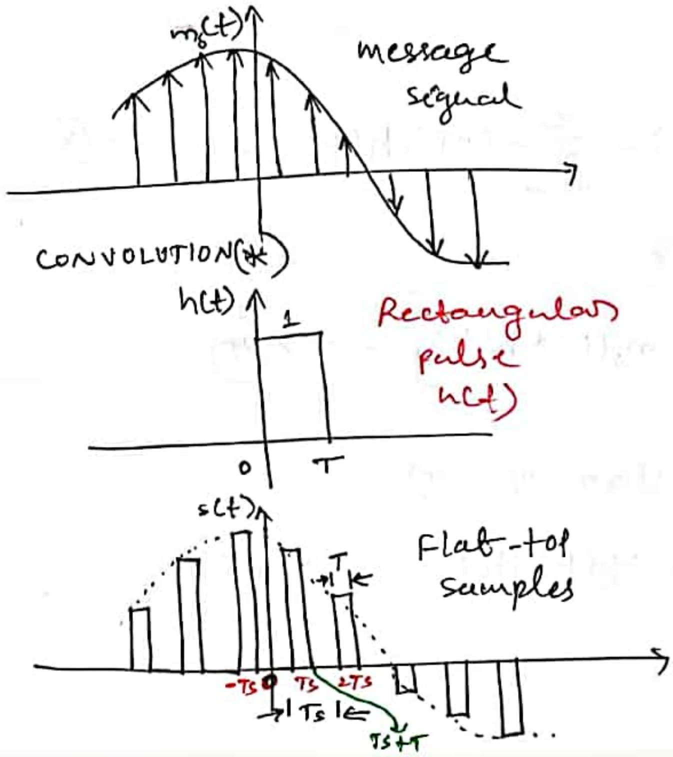

$$h(t) = \begin{cases} 1 & 0 < t < T \\ 0 & t < 0, \text{ and } t > T \end{cases}$$

$$h(t) = \text{rect}\left(\frac{t}{T} - \frac{1}{2}\right)$$

rect

$$\text{rect}(x) = \begin{cases} 1 & -\frac{1}{2} < x < \frac{1}{2} \\ 0 & |x| > \frac{1}{2} \end{cases}$$

Equation (2) can be ~~obtai~~ acheived by convolving eqn (1) i.e $m_\delta(t)$ with h(t).

$$\therefore m_\delta(t) * h(t) = \int_{-\infty}^{\infty} m_\delta(\xi)\, h(t-\xi)\, d\xi$$

$$= \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} m(nT_s)\,\delta(\xi - nT_s)\, h(t-\xi)\, d\xi$$

$$m_\delta(t) * h(t) = \sum_{n=-\infty}^{\infty} m(nT_s) \int_{-\infty}^{\infty} \delta(\tau - nT_s)\, h(t-\tau)\, d\tau$$

Sifting property

$$\int_{-\infty}^{\infty} x(t)\, \delta(t - t_0)\, dt = x(t_0)$$

$$\therefore \int_{-\infty}^{\infty} h(t-\tau)\, \delta(\tau - nT_s)\, d\tau = h(t - nT_s)$$

$$m_\delta(t) * h(t) = \sum_{n=-\infty}^{\infty} m(nT_s)\, h(t - nT_s) \longrightarrow (3)$$

Comparing (2) and (3).

$$\boxed{s(t) = m_\delta(t) * h(t)} \longrightarrow (4)$$

Taking Fourier Transform of (4)

$$S(F) = M_\delta(F)\, H(F) \longrightarrow (5)$$

$$\cancel{S(F) =}$$

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W.K.T

$$M_\delta(F) = \sum_{K=-\infty}^{\infty} F_s\, M(F - KF_s)$$

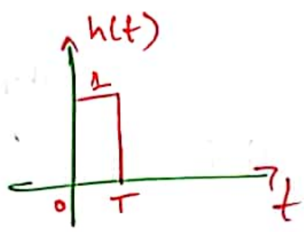
(5) becomes

$$\therefore \boxed{S(F) = F_s \sum_{K=-\infty}^{\infty} M(F - KF_s)\, H(F)} \rightarrow (6).$$

Now, to find $H(F)$

$$H(F) = F.T\{h(t)\}$$

$$H(F) = \int_{-\infty}^{\infty} h(t)\, e^{-j2\pi F t}\, dt$$

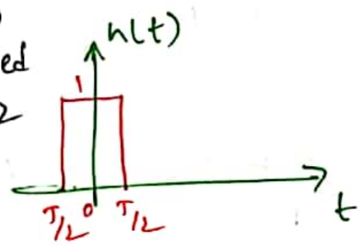


$$T \qquad -j2\pi Ft$$

$$= \int_0^T h(t) e^{-j2\pi F t} \, dt$$

$$= \int_{-T/2}^{T/2} h(t+T/2) e^{-j2\pi F (t + T/2)} \, dt$$

$\rightarrow h(t)$ delayed by $T/2$



$$= e^{-j\cancel{2}\pi F \frac{T}{\cancel{2}}} \int_{-T/2}^{T/2} 1 \cdot e^{-j2\pi F t} \, dt$$

$$H(F) = e^{-j\pi F T} \left[\frac{e^{-j2\pi F t}}{-j2\pi F} \right]_{-T/2}^{T/2}$$

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$$H(F) = e^{-j\pi F T} \left[\frac{e^{-j2\pi F T/2} - e^{-j2\pi F(-T/2)}}{-j2\pi F} \right]$$

$$= e^{-j\pi F T} \left\{ \cancel{-} \left(\frac{e^{j\cancel{2}\pi F T/\cancel{2}} - e^{-j\cancel{2}\pi F T/\cancel{2}}}{\cancel{-}j2\pi F} \right) \right\}$$

$\rightarrow \sin \cancel{2}\pi F T/\cancel{2}$

$$H(F) = e^{-j\pi F T} \frac{\sin \pi F T}{\pi F} \qquad \left[\because \sin\theta = \frac{e^{j\theta} - e^{-j\theta}}{2j} \right]$$

Multiply & Divide by T

$$H(F) = T e^{-j\pi F T} \frac{\sin \pi F T}{\pi F T}$$

$$H(F) = T e^{-j\pi F T} \operatorname{sinc}(FT) \qquad \left[\because \operatorname{sinc}\theta = \frac{\sin \pi\theta}{\pi\theta} \right]$$

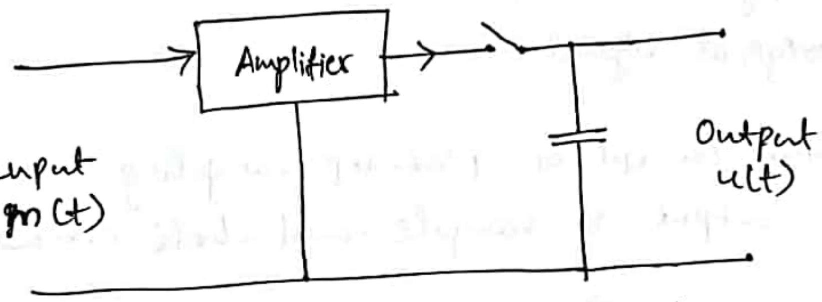
$$\boxed{H(F) = T \operatorname{sinc}(FT) e^{-j\pi F T}} \rightarrow (7)$$

(17)

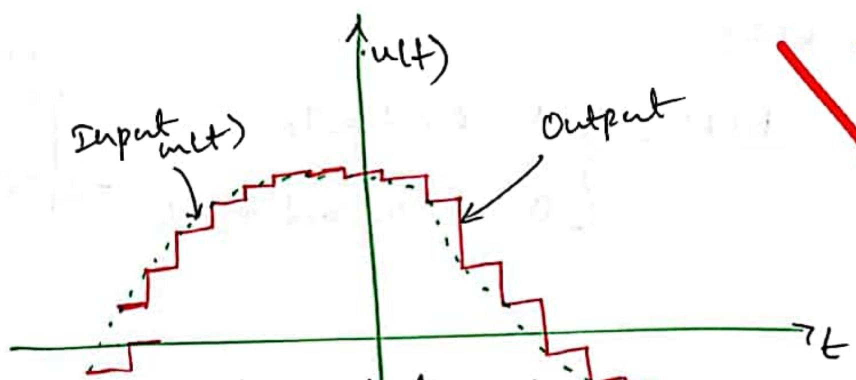
SAMPLE and HOLD Circuit :

In natural sampling and flat-top sampling the spectru of sampled signal is saled by the sampli ratio T/T_s, where T is the sampling pulse duration and T_s is the sampling Period. Typically this ratio is quite small resulting the signal power at the output of low-pass reconstruction filter to be correspondingly small.

We may remedy this situation by the use of ~~amplification~~ a sample-and-hold circuit as shown in fig below.



sample-and-hold circuit



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Ideal output waveform

Working :

The circuit consists of an amplifier of unity gain and low Output impedance, a switch, and a capacitor. It is assumed that the load impedance is large. The

It is assumed that the load impedance is large. The switch is timed to close only for the small duration T of each sampling pulse, ~~which tot~~ during which time capacitor rapidly charges up to a voltage level equal to that of the input sample. which the switch is open, the capacitor retains its voltage level until the next closure of the switch. Thus sample and hold circuit, in its ideal form, produces an output waveform shown in previous fig. that represents a staircase interpolation of the original signal.

From the concept of Flat-top sampling, we can deduce the output of sample-and-hold circuit as

$$u(t) = \sum_{n=-\infty}^{\infty} m(nT_s)\,h(t-nT_s)$$

where ~~h(t)~~

$$h(t) = \begin{cases} 1 & 0 < t < T_s \\ 0 & t < 0,\ \text{and } t > T_s \end{cases}$$

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Further, the spectrum of sample and hold circuit can be written as

$$\boxed{U(f) = f_s \sum_{k=-\infty}^{\infty} M(f-kf_s)\,H(f)}$$

where

$$H(f) = T_s \,\mathrm{sinc}(fT_s)\, e^{-j\pi f T_s}$$

RECONSTRUCTION In sample and Hold sampling

Further, the spectrum of sample and hold circuit can be written as

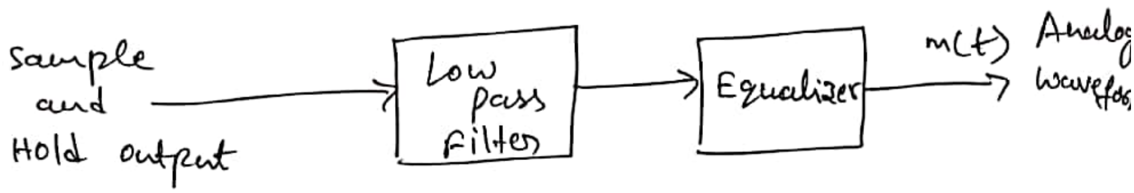
$$\boxed{U(F) = F_s \sum_{K=-\infty}^{\infty} M(F-KF_s)\, H(F)}$$

where

$$H(F) = T_s \operatorname{sinc}(FT_s)\, e^{-j\pi F T_s}$$

RECONSTRUCTION In sample and Hold Sampling

The output of sample and Hold circuit i.e $u(t)$ is passed through a low-pass filter designed to remove components of the spectrum $U(F)$ at multiples of F_s and an equalizer whose amplitude response equals $\frac{1}{|H(F)|}$.



Quantization Noise : OR: Quantization Error: And Signal-to-Noise Ratio (SNR)

Quantization is a many-to-one mapping, in which all sample values in a particular interval are mapped to a quantization level. This implies, there exists an approximation error termed as quantization error.

In a uniform quantizer, the quantization error lies in $\left[-\frac{\Delta}{2}, \frac{\Delta}{2}\right]$.

For example:
Consider the interval $[0, \Delta]$. Its quantization level = midpoint = $\frac{\Delta}{2}$.
error = $\frac{\Delta}{2}$ if $m = 0$
$-\Delta/2$ if $m = \Delta$

Quantization error

$$\boxed{q_e = Q(m) - m}$$

Quantized sample value

True sample value

$$\boxed{\begin{aligned} Q(m) &= m_q \\ m &= m(nT_s) \end{aligned}}$$

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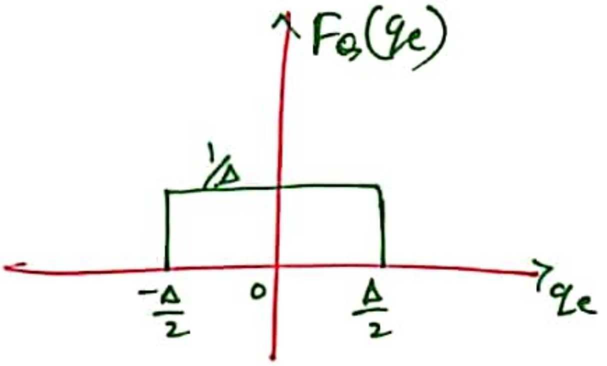
A simple model can be developed by assuming the quantization error to be uniformly distribute between $-\frac{\Delta}{2}$ to $\frac{\Delta}{2}$.

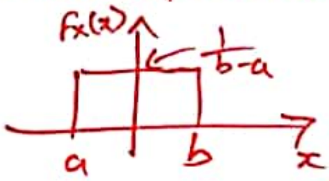


Height $= \frac{1}{\Delta}$. $\Rightarrow$ Area under PDF $= \frac{1}{\Delta} \cdot \Delta = \underline{\underline{1}}$

The Probability Density Function of quantization error

$$\boxed{F_Q(q_e) = \begin{cases} \frac{1}{\Delta} & |q_e| \le \frac{\Delta}{2} \\ 0 & \text{otherwise} \end{cases}}$$

Probability density function (PDF) of uniform distribut



$$\text{PDF} \rightarrow F_X(x) = \begin{cases} \frac{1}{b-a} & a \le x \le b \\ 0 & \text{otherwis} \end{cases}$$

$$\text{Mean} = \mu = \frac{a+b}{2}$$

By Symmetry, average value of quantization error $= 0$.

$\therefore$ The mean of quantization error

$$\boxed{\mu = 0.}$$

$$\boxed{E[Q] = \int_{-\Delta/2}^{\Delta/2} f_Q(q_e)\cdot q_e \, dq_e = 0.}$$

Now, Variance of quantization error

or

$\sigma_Q^2 = E\{Q^2\}$ Power of Quantization error

$$\sigma_Q^2 = E[Q^2] = \int_{-\infty}^{\infty} (q_e - \mu)^2 f_Q(q_e)\, dq_e$$

$$= \int_{-\Delta/2}^{\Delta/2} q_e^2 \times \frac{1}{\Delta}\, dq_e$$

$$= \frac{1}{\Delta}\left[\frac{q_e^3}{3}\right]_{-\Delta/2}^{\Delta/2}$$

$$= \frac{1}{3\Delta}\left[\left(\frac{\Delta}{2}\right)^3 - \left(\frac{-\Delta}{2}\right)^3\right]$$

$$= \frac{1}{3\Delta}\left[\frac{\Delta^3}{8} - \left(-\frac{\Delta^3}{8}\right)\right]$$

$$= \frac{1}{3\Delta}\left[\frac{\Delta^3 + \Delta^3}{8}\right]$$

$$= \frac{1}{3\Delta}\left[\frac{\Delta^3}{4}\right]$$

$$\boxed{\sigma_Q^2 = \frac{\Delta^2}{12}} \rightarrow \text{Quantization Noise power.} \rightarrow (1)$$

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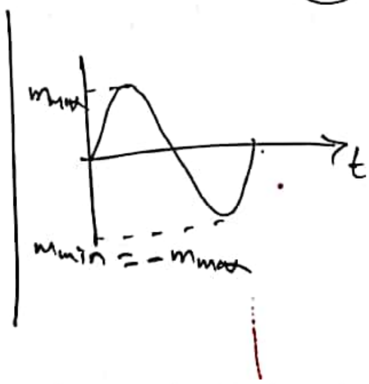
Let us reformulate Δ.

W.K.T

$$\Delta = \frac{m_{max} - m_{min}}{L}$$

Substituting $m_{min} = -m_{max}$



$$= \frac{1}{3\Delta}\left[\frac{\Delta^3}{4}\right]$$

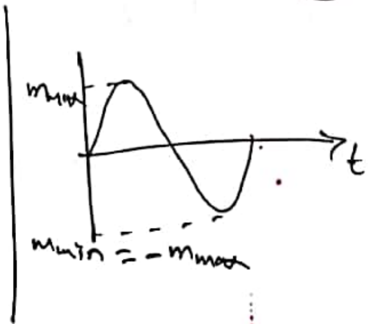
$$\boxed{\sigma_Q^2 = \frac{\Delta^2}{12}} \to \text{Quantization Noise power.} \to (1)$$

2B

let us reformulate Δ.

W.K.T

$$\Delta = \frac{m_{max} - m_{min}}{L}$$



Substituting $m_{min} = -m_{max}$

$$\Delta = \frac{m_{max} - (-m_{max})}{L}$$

$$\boxed{\Delta = \frac{2m_{max}}{L}} \to (2)$$

Substituting (2) in (1)

$$\sigma_Q^2 = \frac{\Delta^2}{12} = \frac{\left[\frac{2m_{max}}{L}\right]^2}{12}$$

$$\sigma_Q^2 = \frac{\cancel{4}^{1}\, m_{max}^2}{\cancel{12}_{3}\, L^2}$$

$L = 2^n$

$$\boxed{\sigma_Q^2 = \frac{1}{3}\frac{m_{max}^2}{2^{2n}}} \to (3)$$

NOTE: Quantization Noise power in dB

Take 10log on both side of (3)

$$10\log\sigma_Q^2 = -10\log 3 + 20\log m_{max} - 2n\log 2$$

$$\sigma^2_{Q(dB)} = -10\log 3 + 20\log m_{max} - 6n$$

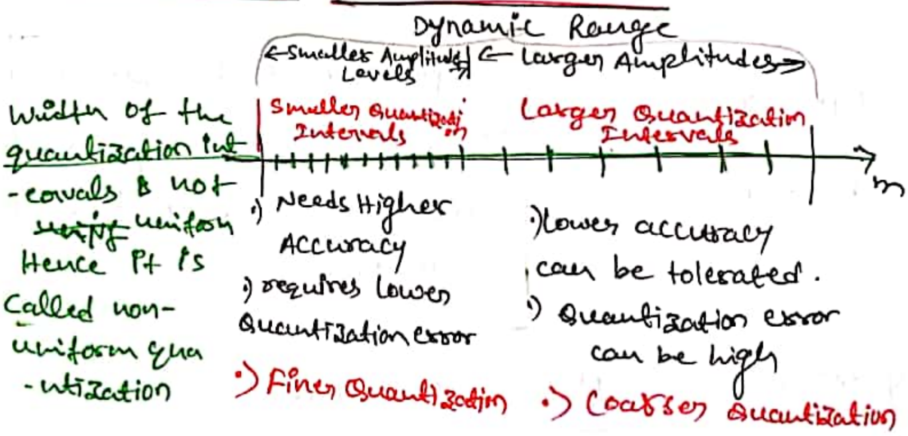
$\therefore$ dB noise variance decreases by a factor of 6dB for each additional bit

ROBUST QUANTIZATION:

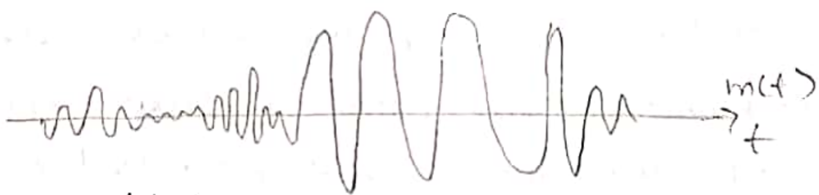
Robust Quantization also referred to as non-uniform Quantization.

A different quantization scheme called "COMPANDING" is used for non-uniform Quantization.

Need for non-Uniform Quantization:



COMPANDING Achieves Non-Uniform Quantization



At lower amplitudes, we need lower quantization error. Hence these amplitudes are quantized with smaller quantization intervals so as to achieve higher accuracy of Reconstruction.

At larger amplitudes, we can tolerate large quantization error. Hence larger amplitudes can be quantized with larger quantization intervals, so that we can have lower accuracy of reconstruction.

Therefore, the width of the ~~signal~~ quantization intervals are different (not uniform). Hence it is called Non-Uniform Quantization.

"COMPANDING is a Technique to achieve non-uniform quantization."

There are Two-methods in COMPANDING:

1. μ-Law COMPRESSOR.
2. A-Law COMPRESSOR.

COMPANDING is the process of Non-Linear mapping from the input to the output.

μ-Law Compressor : or : μ-law COMPANDING :

In the μ-law Companding, the compressor characteristics $c(m)$ is continuous, approximating a linear dependence on m for low input levels and a logarithmic one for high input levels.

The transfer characteristics of μ-law compressor is given as

$$\boxed{c(m) = \frac{\log(1+\mu|m|)}{\log(1+\mu)} \qquad 0 \le |m| \le 1, \quad -1 \le m \le 1} \rightarrow (1)$$

The dynamic range of m is normalized to 0 to 1.

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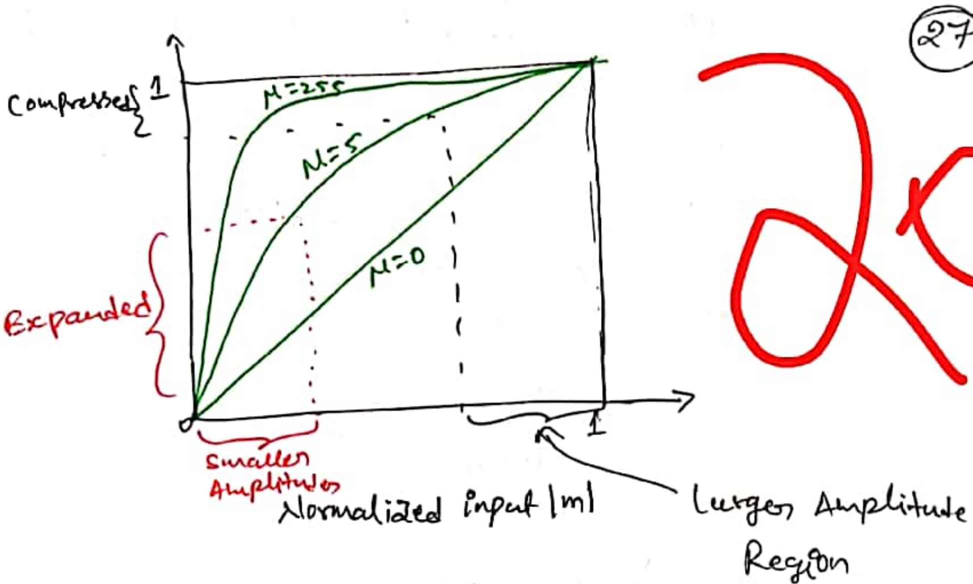

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Smaller amplitude values are expanded so that those sample values can be quantized finely.

The larger amplitude values are compressed so that those samples can be quantized roughly (coarsely).

Further, In μ-law transfer characteristics in (1) as μ approaches to zero.

$$c(m) = \lim_{\mu \to 0} \frac{\log(1+\mu|m|)}{\log(1+\mu)}$$

$$\therefore \lim_{\mu \to 0} \frac{\mu|m|}{\mu} = |m|$$

$\lim_{x \to 0} \log(1+x) \simeq x.$

This implies the linear characteristics for $\mu = 0$. as μ increases the characteristics becomes more and more concave. (logarithmic)

as $\mu \uparrow$, COMPANDING $\uparrow$

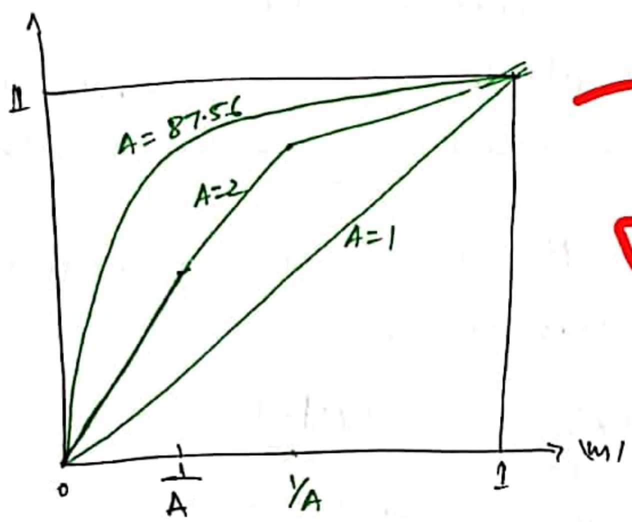
More 'COMPANDING'
$\Rightarrow$ More compression of large amplitude Region
$\Rightarrow$ More expansion of smaller amplitude Region

: A-Law COMPANDING $\boxed{\text{or}}$ A-law Compressor :

The characteristics of A-law compressor is given as

$$c(m) = \begin{cases} \dfrac{A|m|}{1+\log A} & 0 \leq m \leq \dfrac{1}{A} \quad \leftarrow \text{Linear} \\[2ex] \dfrac{1+\log A|m|}{1+\log A} & \dfrac{1}{A} \leq |m| \leq 1 \end{cases}$$

Logarithmic or Concave.

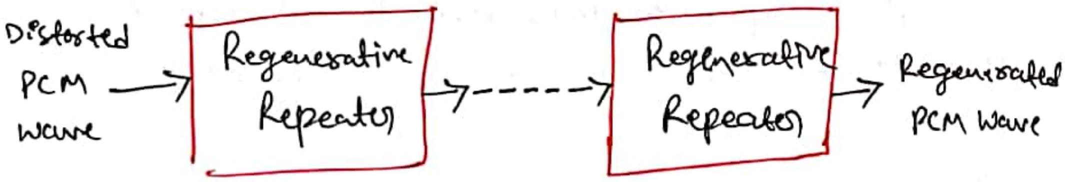


PULSE CODE MODULATION (PCM):

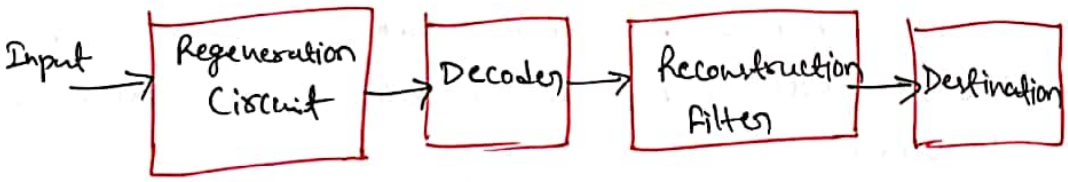
PCM is an analog to digital converter where the information contained in the instantaneous samples of an analog signal are represented by digital codes as a bit stream.

Block diagram:

(a) PCM Transmitter



(b) Transmission path



(c) Receiver

PCM Transmitter:

In practice the low pass filter (pre-alias filter) is used before sampler in order to limit the frequency greater than 'F_m' Hz. Hence message signal is bandlimited to 'F_m' Hz.

Sampler:

The incoming message signal is sampled with a train of Narrow rectangular pulses with sampling rate $F_s \geq 2F_m$. (i.e. above Nyquist rate) to avoid ~~al~~ ALIASING.

Quantizer:

The sampled signal is fed to the quantizer. The quantizer approximate each input sample to a nearest predefined voltage level.

The output of quantizer is discrete time, discrete amplitude signal known as "Quantized signal".

Encoder:

The ~~sam~~ quantized samples are ~~then~~ converted into digital codewords in Encoder. The process of encoding involves allocating some digital codes to each ~~sample~~ quantization level. These digital codes are transmitted as bit stream.

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(31)

Regenerative Repeater:

The PCM signal is reconstructed by means of a regenerative repeater located at sufficiently closed spacing along the transmission path.

The regenerative networks are used at intermediate points between transmitter and receiver. in order to boost up the pulse amplitude

PCM Receiver:

The first operation in the receiver to generate the received pulses.

Decoder :

~~The first operation~~

The decoder converts binary coded signal to a approximated pulses of discrete magnitude.

Reconstruction Filter :

The final operation in the receiver is to recover the original analog signal. This is done by passing the decoder output through a LPF. The output of low pass filter is an analog signal.

Advantages of PCM :

1) Relatively inexpensive digital circuitry is involved in PCM.
2) PCM signals can be multiplexed and transmitted over a common high speed communication link.
3) In long distance transmission, clear waveforms can be regenerated using repeaters.
4) The noise performance of digital system is superior to that of an analog system.

Pulse Degradation In transmission Medium