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Difference b/w analog & digital communication

① Def

Analog Communication :- ~~[illegible]~~ is the technology which uses analog signal for the transmission of information

Digital comm :- is the technology which uses digital signal for the transmission.

② Noise & distortion

Analog comm :- get affected by noise.

Digital comm :- Immune from less noise and distortion.

③ Hardware -

In analog comm. hardware is complicated and less flexible than digital system.

④ Cost :

Low cost in analog comm.
Hight cost in digital ".

⑤ Bandwidth :

Low bandwidth in analog.
High bandwidth in digital.

⑥ Power requirement :

High power is required in analog.
Low power requirement in digital.

⑦ Modulation used :

Analog → amplitude modulation & angle modulation.

Digital → ① Pulse code modulation (PCM)
② Differential PCM (DPCM)
③ ASK (Amplitude shift Keying).
④ PSK (Phase shift Keying)

⑧ Examples.

Analog :- Sine & cosine.
Digital :- Square wave.

Q. Practical examples:

Analog :- Voice & sound.

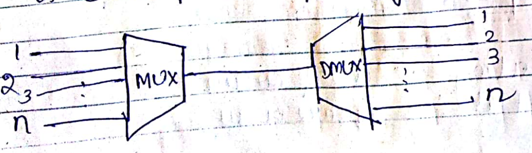
Digital s/g :- computers.

Multiplexing :-

Types:-

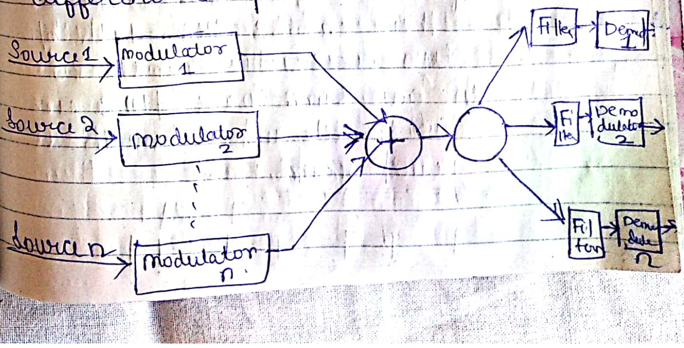
1. Time divison multiplexing
2. Frequency divison multiplexing.

Multiplexing is defined as a technique that allows simultaneous transmission of multiple signals across a ~~signal~~ single channel / data link. When the bandwidth of medium is greater than individual s/g to be transmitted through channel, a medium can be shared by more than one signals. The process of making the most effective use of the available channel capacity is called multiplexing.

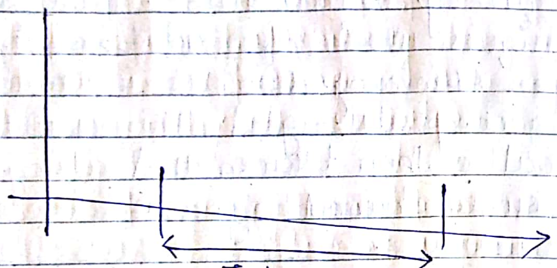


Frequency DM (FDM) :- In FDM the available bandwidth of a single physical medium is sub-divided into several independent frequency channels. Independent message signals are translated into different frequency bands using modulation techniques which are combined by a linear summing circuit in multiplexer to a composite signal, the resulting signal is then transmitted along a single channel by a electro-magnetic wave.

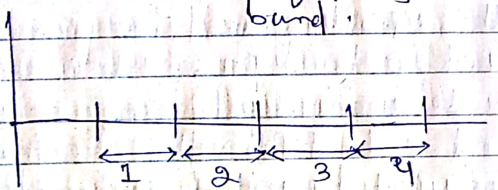
At the receiving end the signal is applied to band pass filters which separate individual frequency channel. The BPF O/P's are then demodulated and distributed to different output channels.



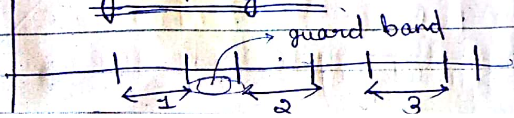
→ We will set the frequency of filter according to which message signal we want.
Then we will use demodulator to retrive our different message signals.



Total frequency band.



If we do continuous distribution there can be cross-talk b/w two consecutive signals so we give some space b/w 2 consecutive signals and is known as ~~guard signal~~ guard band.



→ If the channels are very close to one another it leads to inter-channel cross talk.
Channels must be separated by the strips of unused bandwidth to prevent interchannel cross talk.
These unused channels b/w each successive channel are known as guard bands

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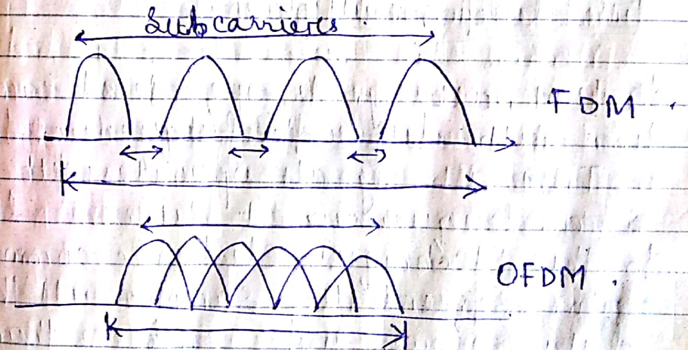
Othogonal frequency division multiplexing

★ Previously, with base band signal we have greater bandwidth. To reduce bandwidth we come to OFDM.



OFDM distributes the data over a large

number of carriers that are spaced apart at a precise frequencies.

- This spacing provides the orthogonality in this technique, which prevents the demodulator from seeing frequencies other than their own.

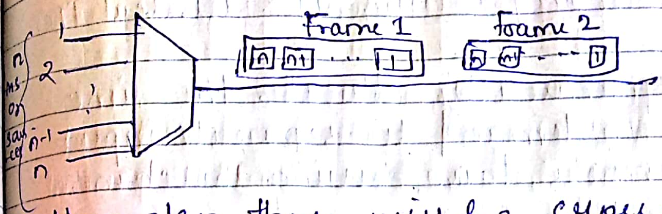
The benefits of OFDM are:-

① High spectral efficiency
② Lower/reduce interference.
③ Lower multipath distortion.

OFDM is being implemented in broadband wireless access system as a way to overcome wireless transmission problems and to improve bandwidth.

Time division multiplexing :

In FDM, all signals operate at same time with different frequencies, but in TDM all signals operate with same frequency at different time. In this technique, electronic commut-ator sequentially samples all the data source and combines them to form composite base band signal which travels through the media and is being demultiplexed into app. independent message signal by the corresponding commutator at the receiving end.



* Here also there will be cross talk, so to prevent cross talking we are sending dead space.

The composite signal have some dead space between successive sampled pulses which is essential to prevent inter-channel cross talk. Along with the sampled pulses 1 synchr-onising pulse is send in each cycle. These data pulses along with the control information form a frame.

Synchronous TDM is called synchronous mainly because each time slot is preassigned to a fixed source.
The time slots are transmitted

irrespective, whether the sources have any data to send or not, so the channel capacity is wasted in synchronous TDM.

Asynchronous / Intelligent / Statistical TDM

It dynamically allocates the time slots on demand to separate input channels, thus saving the channel capacity.

Statistical multiplexer also have many input/output with the buffer associated to each of them. During the input, multiplexer scans the input buffer, collecting data until the frame is filled and send the frame.

Many slots remain unutilized in synchronous TDM but the slots are fully utilized leading to smaller time for the transmission and better utilization of bandwidth of the medium.

- - - . | · | B2 | · · · | A2 | | n · · · | B1 | · · · | A1 |

Synchronous.

| B2 | A2 | | B1 | A1 |

Asynchronous

In synchronous, if we have n sources or ~~multiple~~ message signals then if we have to send only A_1 and B_1 messages then we have to still allot n slots keeping n-2 slots free. So there is wastage of slots.

In asynchronous, if we have to send only two signals then we have to allot only 2 slots (not all n slots). So this is the advantage of asynchronous TDM.

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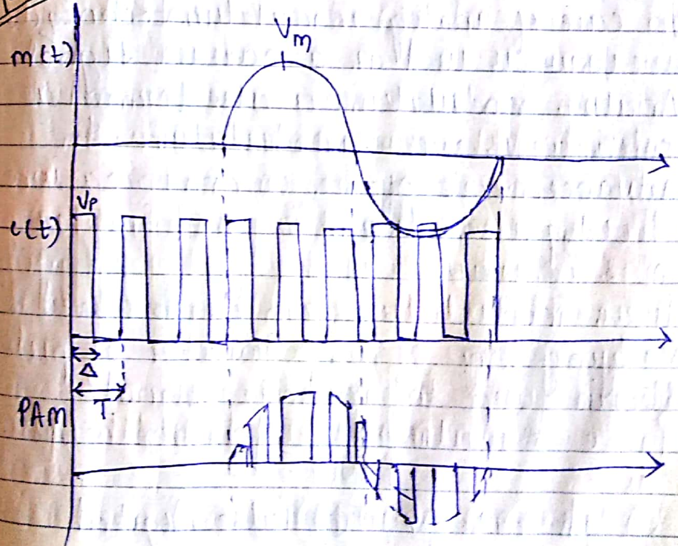
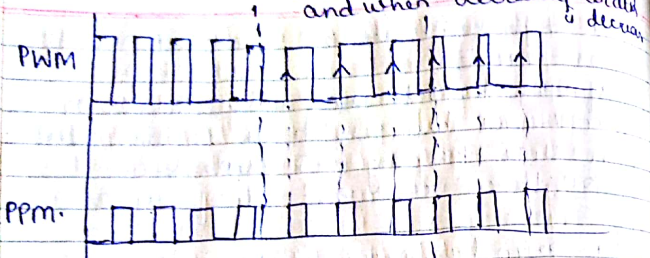


When the amplitude is increasing width is increasing and when decreasing width is decreasing



All frequency, width & amplitude will be same here. Just the position will change.

Leading edge → low to high.
trailing edge → high to low.

In case of analog modulation technique sine wave is used as a carrier signal. Analog modulation is also termed as continuous wave modulation. In advancement of the communication field, pulse train is used in place of sine wave.
Pulse modulation can be obtained by varying one of the parameter of pulse train wrt to message signal.
Pulse modulation is further classified as:-
① Pulse analog ② Pulse Digital.

Modulation (Base Band)
- Analog (continuous wave) modu.
  - AM
  - Angle mod.
    - FM
    - PM
- Pulse modu.
  - Analog Pulse
    - PAM
    - PWM
    - PPM
  - Digital Pulse
    - PCM
    - DPCM
    - DM
    - ADM

PAM → Pulse amplitude modulation
PWM → Pulse width modulation.
PPM → Pulse position modulation.

All the modulation we are studying are base-band modulation.

In Band-pass modulation we have:-
- ASK - Amplitude shift Keying
- FSK - Frequency " "
- PSK - Phase " "
- QPSK - Quadrature Phase shift keying

PCM - Pulse code modulation.
DPCM - Differential pulse code modulation.
DM - Delta modulation
ADM - Adaptive delta modulation.

PAM: (Pulse amplitude modulation)

It is defined as the process of varying the amplitude of the pulse in proportion to the instantaneous variation of the message signal.

$m(t) = V_m \sin \omega_m t$

$c(t) = \begin{cases} V_p & 0 \le t \le \Delta \\ 0 & \Delta \le t \le T \end{cases}$

$x(t) = m(t) \cdot c(t)$

$= \begin{cases} V_p V_m \sin \omega_m t & 0 \le t \le \Delta \\ 0 & \Delta < t \le T \end{cases}$

Pulse width modulation. (PWM)

It is defined as the process of varying the width of the pulse in proportion to instantaneous variation of message.

Δm = modulated width of PWM.

$\Delta m \propto m(t)$

$\Delta m = \Delta[1 + m(t)]$

(I) $m(t) = 0$

$\Delta m = \Delta$

(II) $m(t) = V_m$

$\Delta m = \Delta(1 + V_m)$

(III) $m(t) = -V_m$

$\Delta m = \Delta(1 - V_m)$

Δ = width of pulse in the unmodulated pulse strain.

Δm = width of ~~pulse of modu~~ PWM pulse

PPM: (Pulse position modulation).

It is defined as the process of varying the position of the pulse wrt instantaneous variation of message signal. When there is no message, then the position of leading or trailing edge of the pulse will be equal to original position.

If for the +ve value of message signal, the position will be proportionality shifted right and for the -ve value of message signal the position will proportionality shifted left.

If PWM is generated by the width of trailing edge, then this edge will be extracted to get the position of the pulse in each period.

∵ PPM is used for theoretical interest only and has a limited use in signal processing & communication field.

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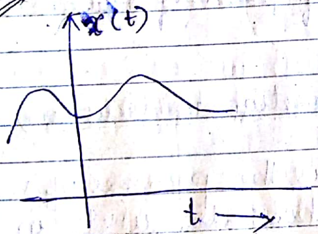
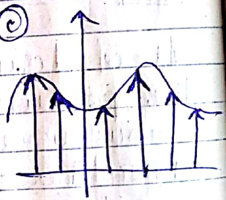
(a)

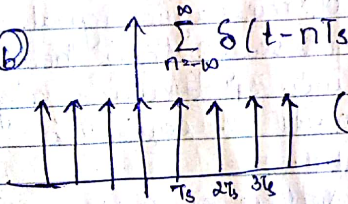


On multiplying (a) and (b) we get (c)

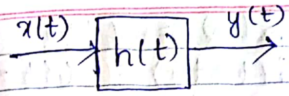
(c)



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



(Impulse strain)



① $x(t) * h(t) = y(t)$

$Y(f) = X(f) \cdot H(f)$ / $X(\omega) . H(\omega)$

② $x(t) \cdot h(t) \longleftrightarrow X(\omega) * H(\omega)$ /

$X(f) * H(f)$

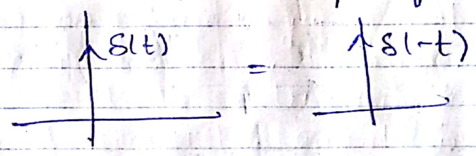
③ Duality property

$x(t) \xleftrightarrow{FT} X(\omega)$

$X(t) \longleftrightarrow 2\pi x(-\omega)$ / $x(-f)$

$\delta(t) \longleftrightarrow 1$

$1 \longleftrightarrow 2\pi \delta(-\omega)$ / $\delta(-f)$.



So we can write,

$1 \longleftrightarrow 2\pi \delta(\omega)$ / $\delta(f)$

(4) $x(t) * \delta(t) = x(t)$

$x(t) * \delta(t-t_0) = x(t-t_0)$

Proof →

$= x(t) * \delta(t)$

$= X(f) \cdot \delta(f)$

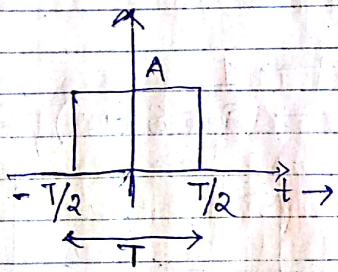
⇓

$X(f) \cdot 1 \xleftrightarrow{\text{Inverse FT}} x(t)$

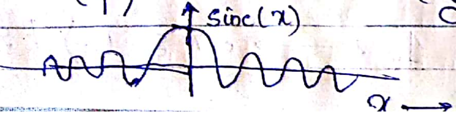
Rectangular function

$x(t) = A\,\mathrm{rect}\left(\frac{t}{T}\right)$

A = amplitude



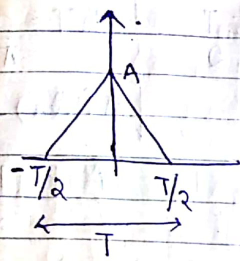
$A\,\mathrm{rect}\left(\frac{t}{T}\right) \xrightarrow{F.T.} AT\,\mathrm{sinc}\left(\frac{\omega T}{2}\right)$



$\mathrm{sinc}(x) \simeq \frac{\sin x}{x} \simeq \frac{\sin \pi x}{\pi x}$

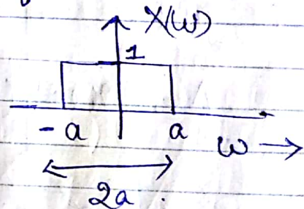
Triangular function:

$A\,\Delta\left(\frac{t}{T}\right) \xleftrightarrow{F.T} \frac{AT}{2}\,\mathrm{sinc}^2\left(\frac{\omega T}{4}\right)$



Using duality property, we can find the fourier transform of sinc function.

$\frac{\sin at}{\pi t} \xleftrightarrow{FT}$



↓

$\frac{a}{\pi}\,\frac{\sin at}{at}$

↓

$\frac{a}{\pi}\,\mathrm{sinc}(at)$

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Proof :-

$$x(t) \longleftrightarrow X(\omega)$$

$$X(t) \longleftrightarrow 2\pi x(-\omega)$$

$$A\,\mathrm{rect}\left(\frac{t}{T}\right) \longleftrightarrow AT\,\mathrm{sinc}\left(\frac{\omega T}{2}\right)$$

$$AT\,\mathrm{sinc}\left(\frac{tT}{2}\right) \longleftrightarrow 2\pi A\,\mathrm{rect}\left(\frac{-\omega}{T}\right)$$

Here assume $at = \frac{tT}{2}$ $\quad\Big|\quad AT = \frac{a}{\pi}$

$T = 2a$ $\quad\Big|\quad A(2a) = \frac{a}{\pi}$

$$A = \frac{1}{2\pi}$$

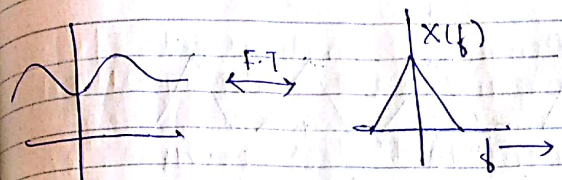
$$\frac{a}{\pi}\,\mathrm{sinc}(at) \longleftrightarrow 2\pi\left(\frac{1}{2\pi}\right)\mathrm{rect}\left(\frac{-\omega}{2a}\right)$$

$$= \mathrm{rect}\left(\frac{-\omega}{2a}\right)$$

As, $\mathrm{rect}\left(\frac{-\omega}{2a}\right) = \mathrm{rect}\left(\frac{\omega}{2a}\right)$

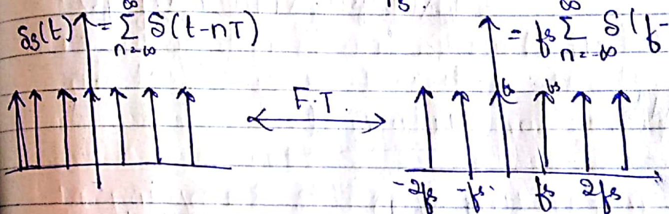
$$\boxed{\text{So, } \frac{a}{\pi}\,\mathrm{sinc}(at) = 1\cdot\mathrm{rect}\left(\frac{\omega}{2a}\right)}$$

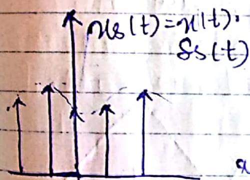
Sampling Theorem



$$\sum_{n=-\infty}^{\infty} \delta(t-nT_s) \longleftrightarrow f_s \sum_{n=-\infty}^{\infty} \delta(f-nf_s)$$

$$f_s = \frac{1}{T_s}$$





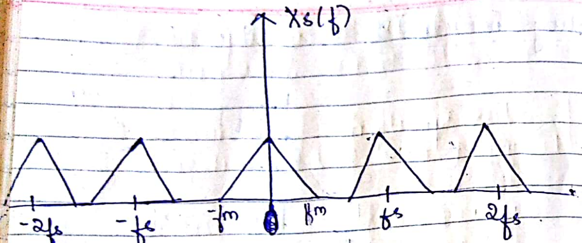
$$x_s(t) \longleftrightarrow X_s(f)$$

$$x(t)\cdot\sum_{n=-\infty}^{\infty}\delta(t-nT_s) = X(f) * f_s\sum_{n=-\infty}^{\infty}\delta(f-nf_s)$$

$$x(t) * \delta(t-t_0) = x(t-t_0)$$

$$X_s(f) = f_s\sum_{n=-\infty}^{\infty} X(f-nf_s)$$

If in time domain we have multiplication then in frequency domain we have convolution.



$$X_s(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)$$

$$= \ldots + f_s X(f+f_s) + f_s X(f) + f_s X(f-f_s) \ldots$$

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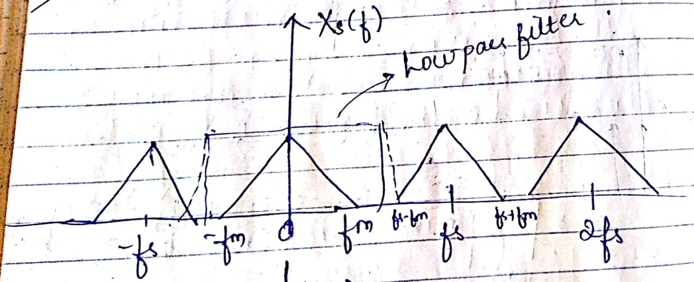
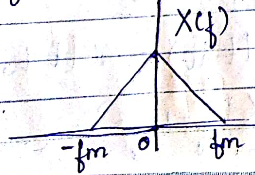
Tutorial





$\longrightarrow x(t)$

① Condition $f_s - f_m \geq f_m$

$f_s \geq 2f_m$ } ~~Overlap~~ Oversampling

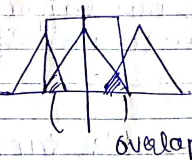
② condition

$f_s = 2f_m$

(Critical sampling)

③ condition

anti-alaising effect



$f_s < 2f_m$

(Undersampling)

Here we can't construct the original signal due to overlapping.

In ideal case we can construct the original signal in critical sampling but ideal low pass filter is not possible.

So we can construct the original s/g only when $f_s \geq 2f_m$.

T_s = sampling interval.
f_s = sampling rate.

→ Minimum sampling rate = $f_s = 2f_m$
So here we call f_s as Nyquist rate.

Then $T_s = \frac{1}{2f_m}$ because $T_s = \frac{1}{f_s}$.

→ Nyquist interval.

→ When there is overlapping we call it as anti-alaising effect in condition 3

The minimum possible sampling rate is called nyquist rate and the maximum possible sampling interval allowed to avoid alaising is called nyquist interval.

$NR = 2f_m$ samples/sec

$NI = \frac{1}{2f_m}$ sec.

Types of sampling. → Done till now

1. Ideal sampling or Instantaneous Sampling.
2. Natural sampling
3. Flat top sampling

Ideal sampling

Natural sampling.

Flat-top sampling

Q $x(t) = \sin 100\pi t$.
Find NR and NI.

$\omega = 100\pi$ $f_m = 50Hz$

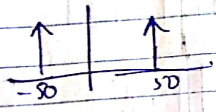
$2\pi f = 100\pi$.

$NR = 2f_m = 2 \times 50$
$= \underline{100 Hz}$ or samples/sec.

$NI = \frac{1}{2f_m} = \frac{1}{100}$ sec.
$= 0.001$ sec.

$$\frac{e^{j2\pi(50)t} - e^{-j2\pi(50)t}}{2j}$$

$= \delta(f-50) - \delta(f+50)$



frequency spectrum

$f_m = 50$.

Q $x(t) = \sin 4000\pi t \cos 6000\pi t$

$= \frac{1}{2}(\sin 10K\pi t - \sin 20K\pi t)$

$\omega_1 = 10000\pi$; $\omega_2 = 2000\pi$

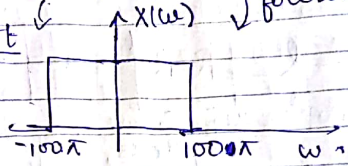
$f_1 = 5000$; $f_2 = 1000$

$f_m =$ max of

$f_s = 2 \times 5000$
$= \underline{\underline{10000 Hz}}$ or samples/sec.

Q sinc (100t). fourier tranform

$= \frac{\sin 100\pi t}{100\pi}$



$\omega = 100\pi$
$2\pi f = 100\pi$
$f_m = 50$

$f_s = 2 \times f_m = 2 \times 50 = 100$ samples/sec

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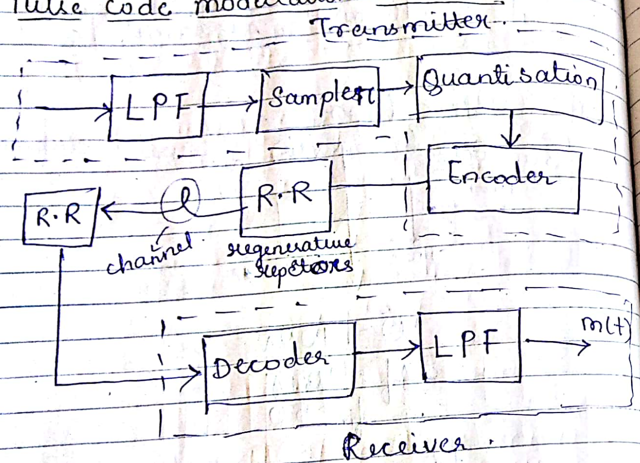
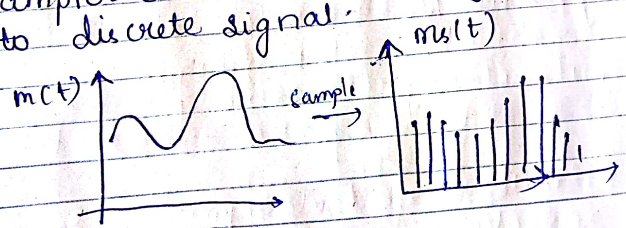
Pulse code modulation (PCM).



Block Diagram of Pulse code mod.

Sampler converts the analog signal to discrete signal.



Quantisation gives levels to the discrete signals as discrete signals are more in number so it is difficult to send these signals.

Eg: 0.1 – 0.5 → 0V
0.6 – 0.9 → 1V
1.0 – 1.5 → 2V
1.6 – 1.9 → 3V

No:- of bits per sample = n which will be defined in encoder.
We have defined one level of logic in binary logic.
If n = 2 we have 4 combinations → 00, 01, 10, 11

Eg:- so we can give logic 1 = 00
" " " " 2 = 01
" " " " 3 = 10

If we increase the number of levels then our number of bits/sample will also increase.

RR → It will remove the noise also.
R.R will transfer the original binary logic as it is (by eliminating or minimizing the noise)

Decoder → will again convert the binary bits into voltage logic like 1V, 2V, etc.

For different sampling voltage the Quantisation voltage is same. So this is known as Quantisation error.

Eg:	S.V	Q.V
	0.6	1V
	0.7	1V

$$\boxed{Q_e = -Q.V + S.V}$$

Quantisation error.

★. The bandwidth of m(t) is more, so LPF limit the bandwidth of m(t)

① The LPF is used as antialiasing filter
② The resulting band limited signal is oversampled by the sampler.
③ Quantizer rounds off each of the sampled voltage to the nearest predefined quantisation voltage.

Q_e = Sample voltage − Quantized Voltage

→ Encoder represents each of the quantization voltage by a unique binary code.

→ R. Repeaters eliminates the channel noise and fresh copy of transmitted electrical s/g is transmitted back.

→ Decoder will do inverse of encoder (Encoding is reversible but Quantisation is irreversible).

→ LPF reconstruct the message s/g from its quantised equivalent.

Levels depends upon no:- of bits/sample

levels = 2^n

n = no:- of bits/sample

$\boxed{L = 2^n}$, L = No:- of levels

$$\boxed{Q_e = S.V. - Q.V.}$$

Δ = step size

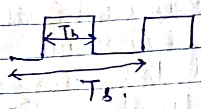
$$= \frac{V_{max} - V_{min}}{L}$$

$$[Q_e]_{max} = \pm \Delta/2$$

R_b = bit rate = Bits/sec.

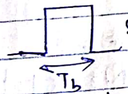
$$\frac{\text{Bits}}{\text{samples}} \times \frac{\text{samples}}{\text{sec}}.$$

$$\boxed{R_b = nf_s.}$$



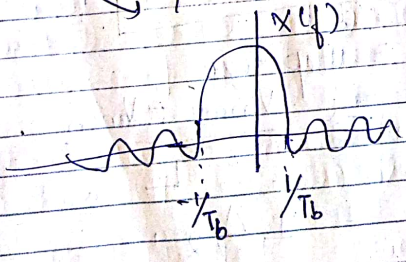
T_b = bit duration

$$\boxed{T_b = \frac{1}{R_b}}$$

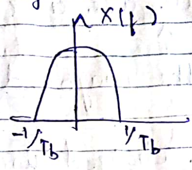


rectangular function

frequency spectrum

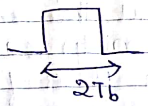


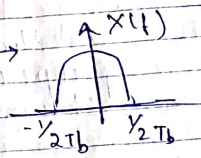
But for calculation of bandwidth, we will only take



Now our BW $= \frac{1}{T_b} = R_b$.

Now for





BW $= \frac{1}{2T_b} = \frac{R_b}{2}$

$BW_{max} = R_b = nf_s$

$BW_{min} = \frac{R_b}{2} = \frac{nf_s}{2}$

In PCM there can be 2 noises:-

1. Channel noise → (we decrease this using RR)
2. Quantization error.

As our $Q_e = \frac{\Delta}{2}$ and $\Delta \propto \frac{1}{L}$

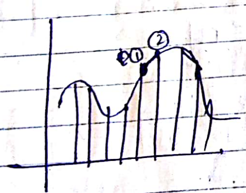
so if we increase the level then, n is increase, then Δ will decrease Q_e is decrease but BW = nfs

so as we are increasing n then BW is ↑

So this is the major drawback.

Δ can be changed using Vmax and Vmin. But this is the property of m(t) so we can't change it as it is fixed.

But in DPCM we will change it.



We will now send the difference of 1 and ② or ② and ③. i.e present value and one future value.

So now our Vmax - Vmin is minimized, Δ also ↓.

Thus Qe ↓.

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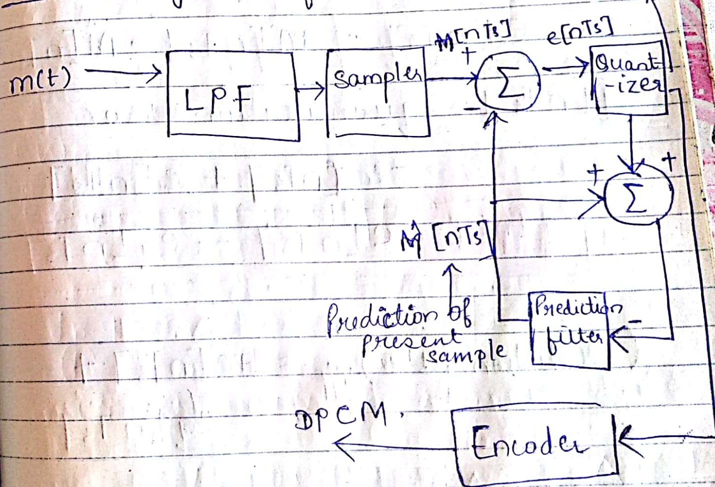
Differential PCM

In DPCM quantization error can be decreased without affecting channel bandwidth requirement. In DPCM quantization error can be decrease by decreasing the dynamic range of a signal to be quantized.

Block diagram of Transmitter part



LPF → defined range of m(t).

Sampler → convert m(t) to discrete.

By analyzing the past behaviour of the signal to sufficient time extent, prediction filter predicts very much nearby future value of the signal.

Dynamic range of $e[nT_s]$ to be quantized will be very small so that corresponding quantization error will also be very small.

$$e[nT_s] = M[nT_s] - \hat{M}[nT_s]$$

$q[nT_s]$ = quantization error

$$= -e_q[nT_s] + e[nT_s]$$

$$\boxed{q[nT_s] = e[nT_s] - e_q[nT_s]}$$

$$(\text{Prediction filter})_{I/P} = \hat{m}[nT_s] + e_q[nT_s]$$

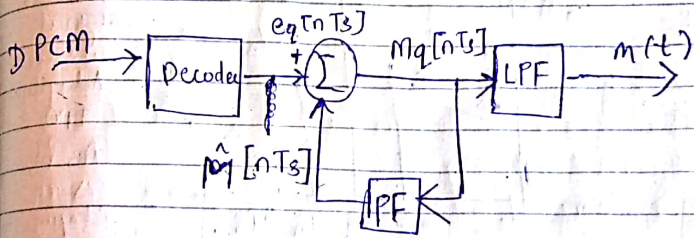
$$= \hat{m}[nT_s] + e[nT_s] - q[nT_s]$$

$$= M[nT_s] - q[nT_s]$$

As $q[nT_s]$ is very small then

$$(\text{Prediction filter})_{I/P} = M[nT_s]$$

<u>Block diagram of receiver part</u>.



$$M_q[nT_s] = \text{Input (Prediction filter)}$$

$$= \hat{M}[nT_s] + e_q[nT_s]$$

$$= \hat{M}[nT_s] + e[nT_s] - q[nT_s]$$

$$= M[nT_s] - \underbrace{q[nT_s]}_{\text{very small}} \approx 0$$

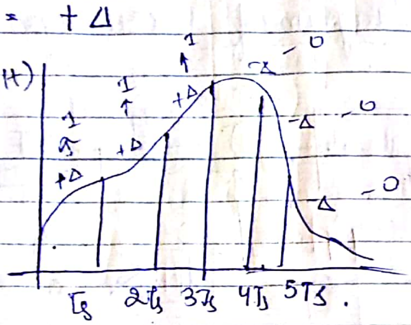
$$\boxed{M_q[nT_s] = M[nT_s]}$$

Delta modulation (DM).

In delta modulation $n = 1$ always.
→ We also call it as 1 bit DPCM.

$0 = -\Delta$
$1 = +\Delta$

Case 1

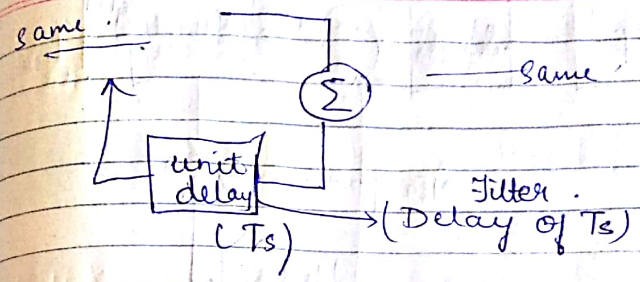


For this type of signal we use Adaptive DM. (As here Δ is changing with time)

Through encoding we will send 111000
So as $n = 1$ we have either 0 or 1 as bits.
So we represent $-\Delta = 0$.
and Δ as 1.

Block diagram of Transmitter section

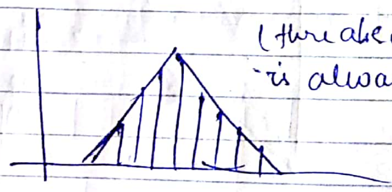
Same as previous DPCM transmitter section but instead of prediction filter we will use unit delay (of T_s).



Drawback:

Here we have taken Δ as difference everywhere.
But it is not always possible.
For straight line only, DM is preferable.
For eg:,

Case 2.



(Here also diff is always Δ).

To overcome this, we move to ADM.
(Adaptive delta modulation).

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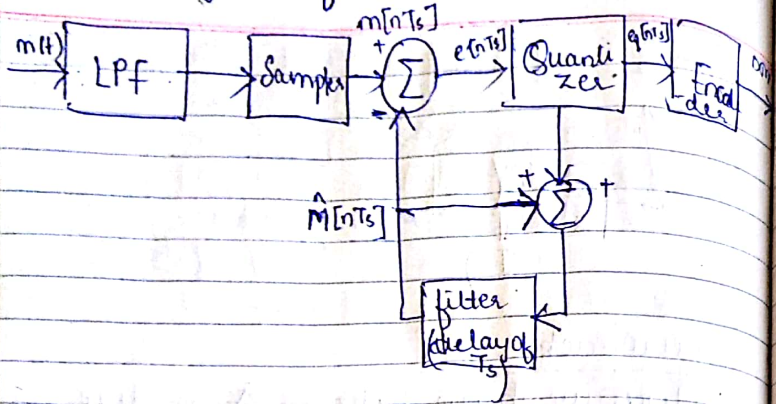
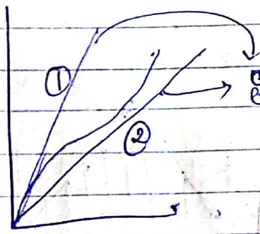
Block diagram of Transmitter



In the receiver section, we have to recover our message signal.
So in case 1:-



either ① or ② will be recovered instead of original signal.

So here we have two types of error.

For ~~case~~ ① we have slope overload error.

$$\Delta < \Delta_{out}$$

(Here we have to decrease Δ) $\Delta\downarrow$.

For ② we have granular error.

$$\Delta > \Delta_{out}$$

(Here we have to decrease $\Delta(\downarrow)$)

$\Delta \rightarrow$ original signal.
$\Delta_{out} \rightarrow$ received signal

$$m(t) = A_m \cos 2\pi f_m t.$$

$$\boxed{\frac{\Delta_{out}}{T_s} = \frac{dm(t)}{dt}} = -A_m \sin 2\pi f_m t \cdot 2\pi f_m = -2\pi f_m A_m \sin 2\pi f_m t$$

This is done for proper reconstruct. So to remove error in delta modulation, we have moved to adaptive delta modulation. In this ADM, we are changing m(t) wrt to time and it becomes equal to $\frac{\Delta_{out}}{T_s}$.