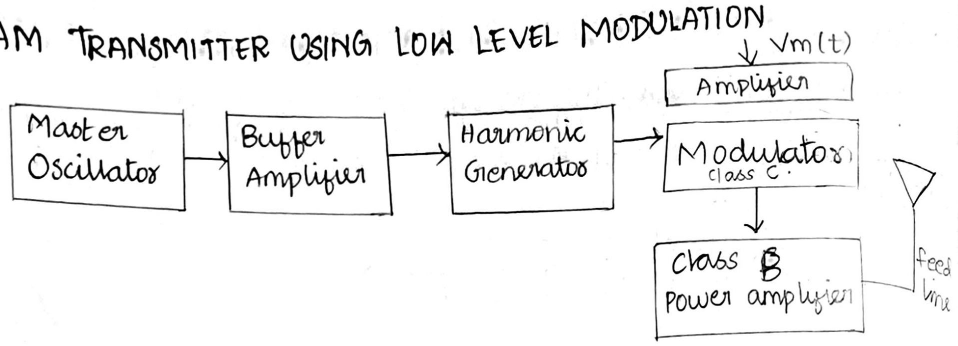
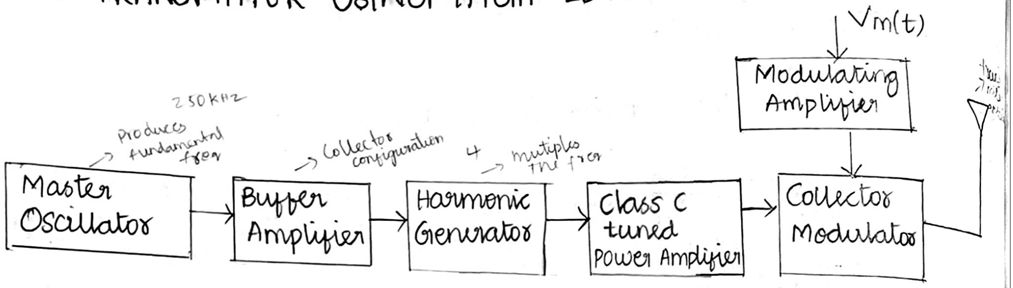
9/9/22

UNIT 2: TRANSMITTER AND RECEIVER

AM TRANSMITTER USING LOW LEVEL MODULATION



AM TRANSMITTER USING HIGH LEVEL MODULATION



Modulation takes place at low power level and then amplified by a Class B power amplifier (low level).

The carrier is first amplified by a class C amplifier and then modulation is done at high power level. (high level)

MASTER OSCILLATOR.

Generates carrier frequency at fundamental frequency. frequency is increased to the desired frequency value by Harmonic generator

BUFFER AMPLIFIER

Buffer amplifier is used between master oscillator and Harmonic oscillator and provides high input impedance from the MO.

amplifier → to match the antenna power rating

HARMONIC GENERATOR.

Harmonic generator produces frequency through frequency multiplication. The desired frequency is Chosen through class C tuned power amplifier.

NOTE::

One more stage of class C tuned power amplifier is used at high level modulation used to amplify the C.S.

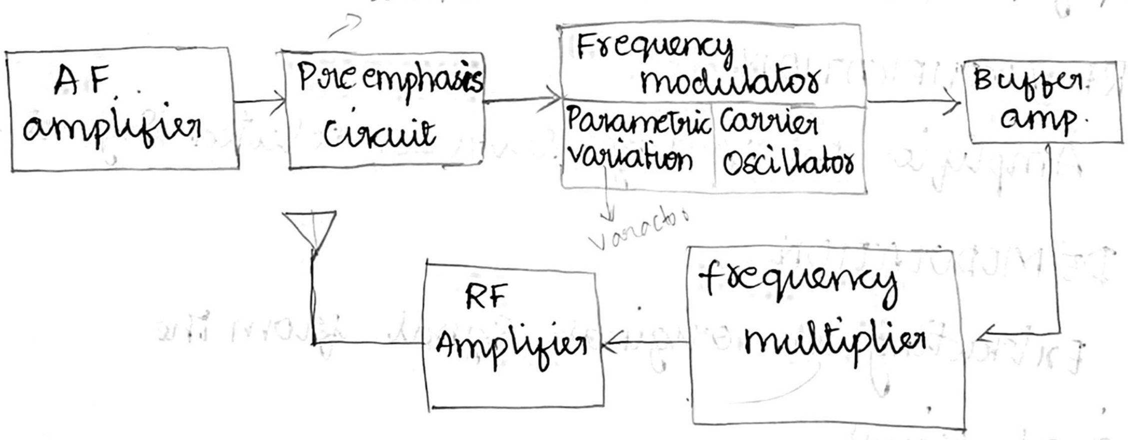
FEEDER AND THE ANTENNA

The transmitted power is fed to the antenna for radiation through a properly designed feeder (transmission line).

FM TRANSMITTERS (DIRECT METHOD)

$$\phi_{FM}(t) = V_c \cos\left[\omega_c t + K_f V_m(t)\right] \qquad K_f = \frac{\Delta\omega}{V_m} = \frac{\Delta f}{V_m}$$

artificial boosting of amp of high freq Components



Show that a square law circuit provides harmonics of the FM signal and also the frequency deviation.

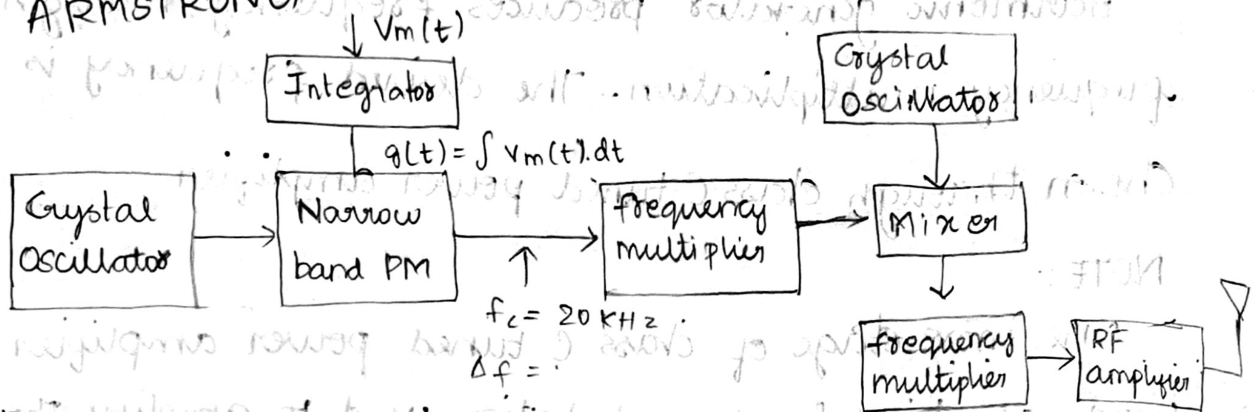
$$\phi^2_{FM}(t) = V_c^2 \cos^2\left[\omega_c t + k_f V_m(t)\right] \qquad \frac{1+\cos(2\theta)}{2}$$

$$= \frac{V_c^2}{2}\left[1 + \cancel{2}\cos\left[2\omega_c t + 2K_f V_m(t)\right]\right]$$

Received Signal → Noisy Version of the Transmitted Signal.

FM TRANSMITTER

ARMSTRONG



RECEIVERS.

Functions of Receiver.

1. Interception
2. Selection → tuned circuit
3. RF amplification
4. Demodulation (Detection) → AF sig
5. Reproduction

1. INTERCEPTION

Interception is the process of accepting electromagnetic signals.

2. SELECTION.

Selecting the particular desired RF carrier frequency using tuned circuit.

3. RF AMPLIFICATION.

Amplifies the voltage level of selected signal.

4. DEMODULATION.

Extracting the original signal from the received signal.

5. REPRODUCTION.

Conversion of audio signal into message signal.

CHARACTERISTICS OF RECEIVERS.

1. Selectivity
2. Sensitivity
3. FEIDILITY

SELECTIVITY

Ability of the receiver to select desired frequency.
For better selectivity proper tuned circuit with sharpened response

SENSITIVITY.

Ability of the receiver to sense the smallest Change in the received Signal

IF → intermediate freq
455 MHz.
IF = Incoming freq − tuned freq

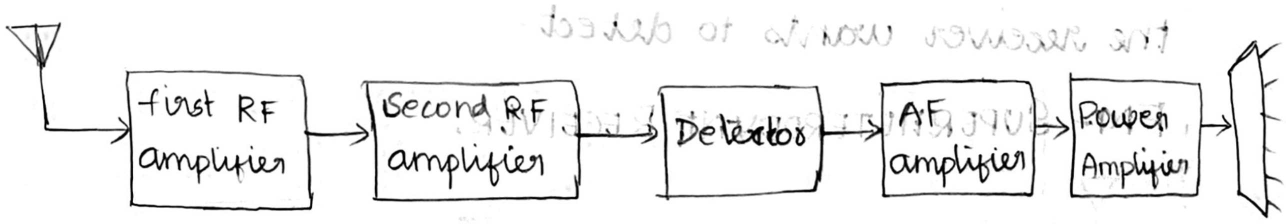
FEIDILITY

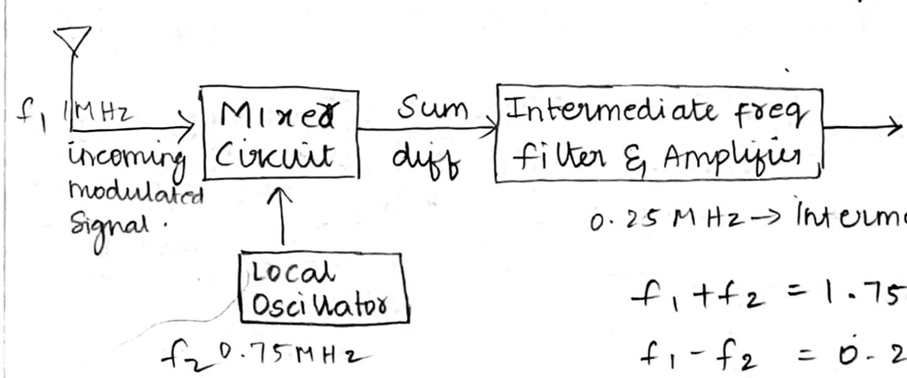
Ability of the receiver to produce exact replica of message Signal at the detector output.

TYPES OF RECEIVER.

Tuned Radio Frequency (TRF)

Principle used : Superheterodyne





0.25 MHz → Intermediate frequency

$f_1 + f_2 = 1.75$ MHz sum (invalid)

$f_1 - f_2 = 0.25$ MHz diff (valid)

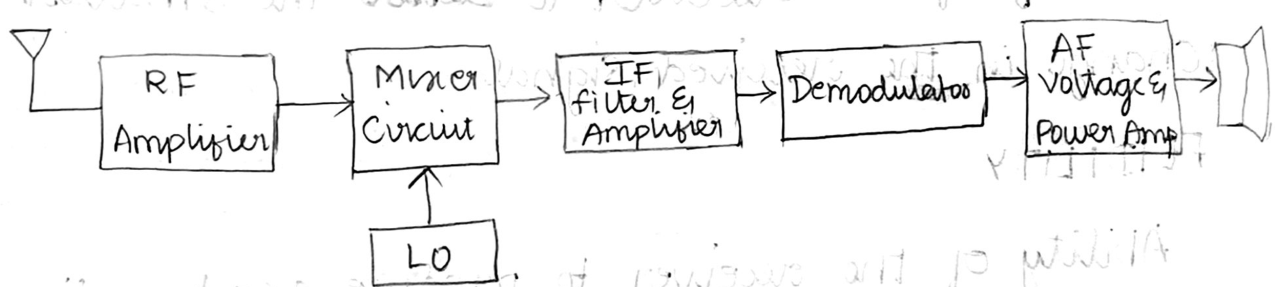
Superhetorodyne principle works based on mixer circuit, the local oscillator in the above circuit is act as variable oscillator filter frequency filter and tunning is performed by varying the tuned circuit within the receiver.

→ The intermediate frequency value for AM radio is 455 KHz

→ For FM broadcast receiver the IF is 10.7 MHz.

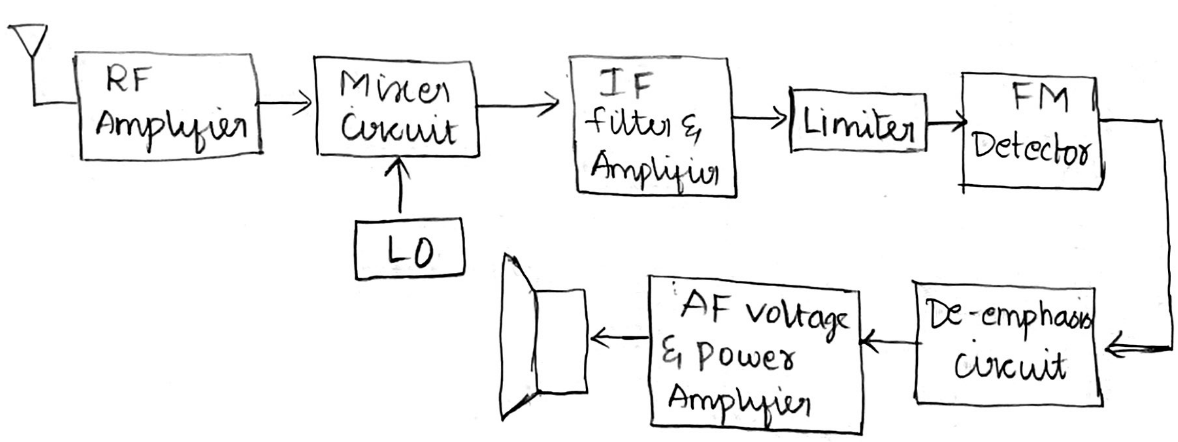
→ For Satellite and terrestrial, Microwave communication the IF is 70 MHz.

AM - SUPERHETORODYNE RECEIVER.



RF amplifier and tuning is added to select the right frequency and reject the image frequencies. Demodulators are added according to what signal the receiver wants to detect.

FM- SUPERHETERODYNE RECEIVER.

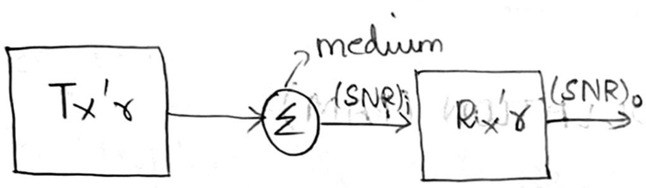


$\gamma \rightarrow$ figure of merit $= \frac{(SNR)_o}{(SNR)_i}$ AWGN - Additive White Gaussian Noise

LIMITER

FM detector needs a constant amplitude FM signal at its input so an IF amplifier output voltage is further amplified to a pre determined value to remove amplitude fluctuation due to noise.

NOISE

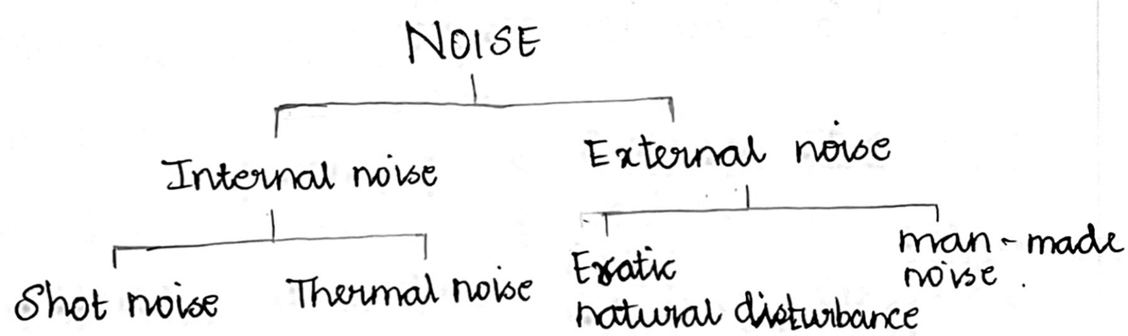


Noise will always appear in Signal's amplitude Variation

$$(SNR)_i = \frac{\text{Signal Power}}{\text{Noise Power}}$$

SNR - Signal to Noise Ratio

$$\gamma (\text{figure of merit}) = \frac{(SNR)_o}{(SNR)_i}$$

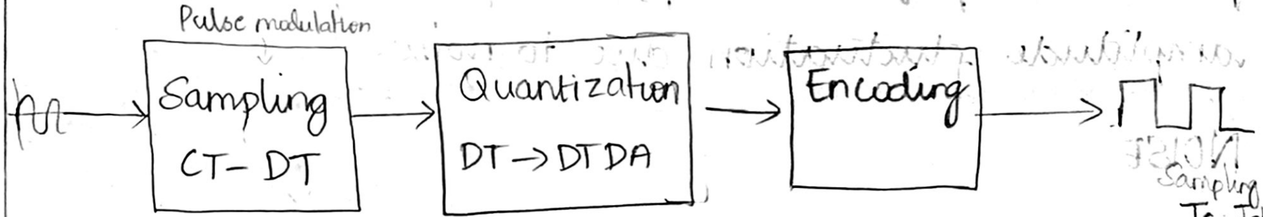


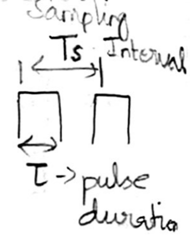
23/9

UNIT-3 PULSE MODULATION SYSTEM

Sampling ⇒ PAM · Thresold detector

CONVERTING ANALOG TO DIGITAL.





PULSE AMPLITUDE MODULATION (PAM)

Message signal

$f_s \geq 2 f_m$

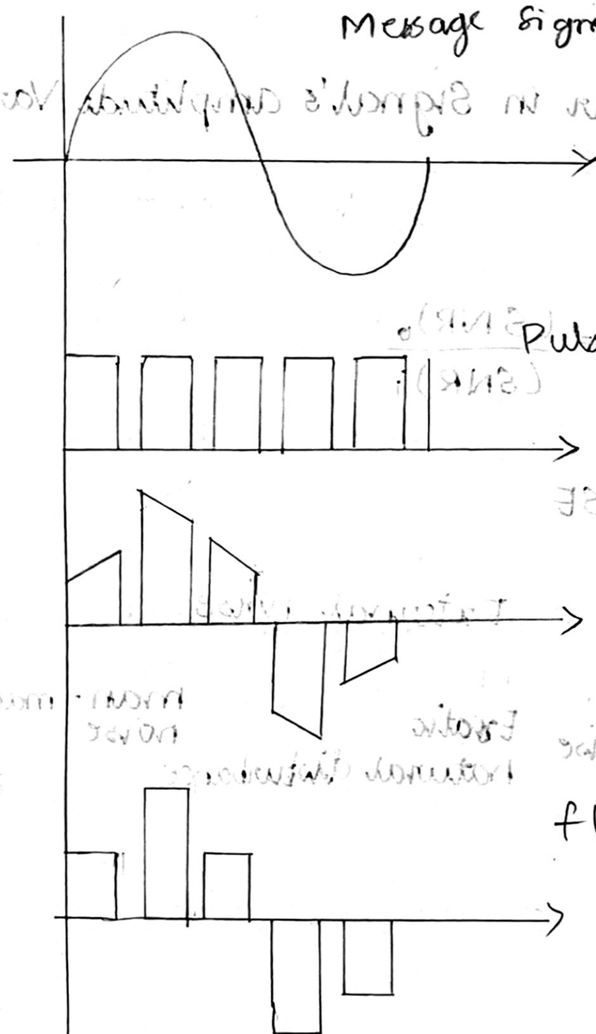
PWM => freq. Modulation

Pulse

Natural Sampled o/p.

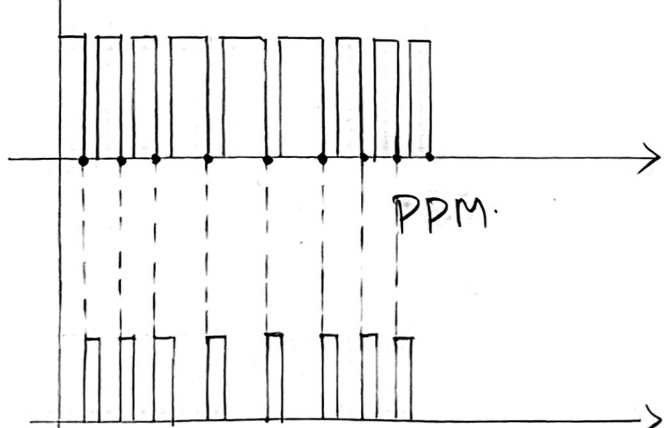
PPM => phase Modulation

flat top o/p

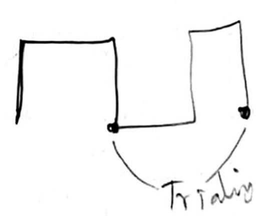


PWM.

PPM.



for every trialing edge of PWM o/p draw a PPM pulse of duration noted for PPM o/p



no of bits ↑ Quantization ↑
resolution ↓

$N \rightarrow$ no of bits used to represent each dis. amplitud

PULSE CODE MODULATION SYSTEMS (A/D Converter)

> disadvantage (redundant bits)

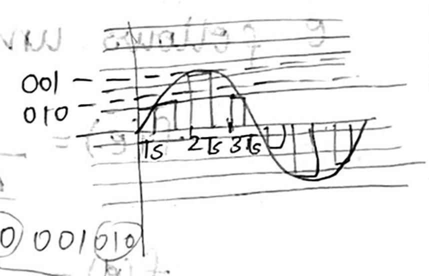
$$\text{Analog} \rightarrow \boxed{\text{Sampling}} \rightarrow \boxed{\text{Quantization}} \rightarrow \boxed{\text{Encoding}} \rightarrow \text{digital}$$

$f_s \geq 2f_m$ $\qquad L = 2^N$ $\qquad N \rightarrow$ no of bits used to represent each discrete amplitude level.

Sampling

Sampling of continuous signal at regular intervals. The sampled output is a discrete time continuous amplitude signal.

$$f_s \geq 2f_m.$$



Quantization

Round off the sample amplitude to the closest amplitude levels.

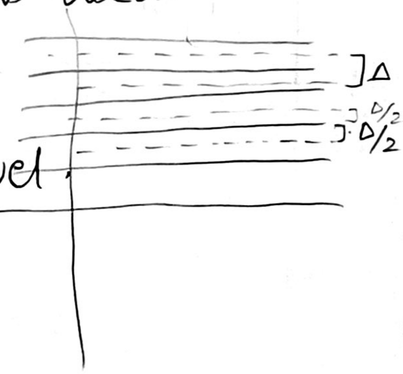
$L = 2^N$ ($N \rightarrow$ no of bits)

$q_e \Rightarrow$ The difference between the original amplitude and the quantized amplitude is known as quantization error.

RESOLUTION

Degree of q_e present in A/D converter ~~with~~ wrt the power consumption & no of bits useds.

$L \rightarrow$ no of quantization level



System = Transmitter ...

SNR for PCM system. (Signal to Noise Ratio)

$$SNR = \frac{\text{Signal Power}}{\text{Noise power}}.$$

$$\text{Signal Power} = \left(\frac{V_{max}}{\sqrt{2}}\right)^2 = \frac{V_{max}^2}{2}. \qquad R=1$$

QUANTIZATION NOISE POWER (q_N)

Quantization noise power, $q_N = E[e^2] - (E[e])^2$

$e \rightarrow$ quantization error.

e follows uniform distribution, $f(x) = \frac{1}{b-a} \quad a<x<b$

$$f(e) = \frac{1}{\frac{\Delta}{2} - \left(-\frac{\Delta}{2}\right)} \qquad -\frac{\Delta}{2} < e < \frac{\Delta}{2}.$$

$\Delta \rightarrow$ step size

$$f(e) = \frac{1}{\frac{\Delta}{2} + \frac{\Delta}{2}} = \frac{1}{\Delta}.$$

$$E(x) = \int_a^b x f(x).dx$$

$$E(e) = \int_{-\Delta/2}^{\Delta/2} e \cdot \frac{1}{\Delta} \cdot de$$

$$E(x^2) = \int_a^b x^2 f(x).dx$$

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

$$= \frac{1}{2\Delta}(0) = 0.$$

$$E[e^2] = \int_{-\Delta/2}^{\Delta/2} e^2 \cdot \frac{1}{\Delta} \cdot de$$

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

$$= \frac{1}{3\Delta}\left[\frac{\Delta^3}{8} + \frac{\Delta^3}{8}\right] = \frac{1}{3\Delta}\left[\frac{2\Delta^3}{8}\right] = \frac{\Delta^2}{12}$$

$$q_N = E(e^2) - [E(e)]^2$$
$$= \frac{\Delta^2}{12} - 0$$
$$q_N = \frac{\Delta^2}{12}$$

$\Delta \rightarrow$ step size.

$$\Delta = \frac{2V_{max}}{L}$$
$$= \frac{2V_{max}}{2^N}$$

$$SNR = \frac{\frac{V_{max}^2}{2}}{\frac{\Delta^2}{12}} = \frac{V_{max}^2}{2} \times \frac{12}{\Delta^2}$$
$$= \frac{6V_{max}^2}{\Delta^2} = \frac{6V_{max}^2}{\left(\frac{2V_{max}}{2^N}\right)^2}$$
$$= \frac{6V_{max}^2}{\frac{4V_{max}^2}{2^{2N}}}$$
$$= \frac{6V_{max}^2 \times 2^{2N}}{4V_{max}^2} = \frac{3}{2} \times 2^{2N}$$

$$SNR = \frac{3}{2} \times 2^{2N}$$

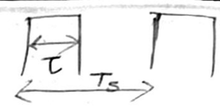
$N \rightarrow$ no of bits

$$SNR \text{ in dB} = 10 \log SNR$$
$$= 10 \log \left[\frac{3}{2} \times 2^{2N}\right].$$
$$= 10\left[\log\left(\frac{3}{2}\right) + \log\left(2^{2N}\right)\right]$$
$$= 10\left[\log(1.5) + 2N\log(2)\right]$$
$$= 10\left[0.176 + 2N \times 0.3010\right]$$

SNR for PCM $= 1.76 + 6N$

The abv eqn is The SNR of PCM for Sinusoidal i/p

PCM requires more bandwidth



BANDWIDTH OF PCM SYSMEM.

For a N bit binary PCM, if F_s is the Sampling frequency and n time slots are there in one T_s (Sampling time) then pulse width τ is

more Quantization level
less noise

$$\tau = \frac{T_s}{n} = \frac{1}{nF_s}$$

The channel bandwidth $B_T \geq \frac{1}{2\tau} = \frac{nF_s}{2}$.

Data Rate (R_b) (bits/seconds)

No of bits transmitted per second (Bps)

$$R_b = nF_s$$

$$B_T \geq \frac{R_b}{2}$$

two types of noise
→ Quantization noise — eliminated by increases Quantization level
→ channel noise — eliminated by Regenerative Repeater

BLOCK DIAGRAM FOR PCM SYSTEMS.

a) TRANSMITTER.

Source of CT Signal → LPF → Sampler → Quantizer → Encoder →

b) TRANSMISSION PATH

Distorted PCM Signal → Regenerative Repeater (Remove noise) → . . . → Regenerative Repeater →

c) RECEIVER

Channel output → Regenerative Circuit → Decoder → Reconstruction filter → Destination →

Created with Scanner Pro

Explain the block diagrams of PCM System and Noise in PCM System.

TRANSMITTER.

SOURCE OF CT SIGNAL.

Source of CT signal is a function generator which generates a sinusoidal waveform of desired frequency and amplitude (message signal).

LOW PASS FILTER (LPF)

The message signal which is in the continuous time form, is allowed to pass through a LPF. This LPF has a cut off frequency fm eliminates the high-frequency components of the message signal and passes only the frequency components that lie below fm.

SAMPLER (o/p of Sampler is PAM)

The output of the LPF is then fed to a sampler which samples the analog signal at regular intervals. The sampling of the signal is done at the sampling frequency fs. fs is selected using sampling theorem $f_s \geq 2f_m$. The output of sampler is a discrete time continuous amplitude signal.

QUANTIZER.

The Sampler produces a distrete time continuous amp. signal. The amplitude is still analog. A quantizer rounds off each amplitude to the nearest quantization level. Approximation of amplitude levels introduces some distortion or noise into the signal, this error is known as quantization error.

ENCODER. (min the bandwidth used)

Encoder converts the quantized signal into binary codes. This generates a digitally encoded signal which is a sequence of binary pulses.

No of quantization level, $\Delta' = 2^N$

$N \rightarrow$ no of bits

TRANSMISSION PATH

REGENERATIVE REPEATER.

A regenerative repeater boosts the signal strength along the transmission path. The medium or channel, through which the signal travels is called transmission path. The medium introduces distortion in the signal during transmission. This distortion is eliminated by the regeneration repeater.

Regenerative repeater detects and regenerates the original signal strength which is free from noise.

RECEIVER.

REGENERATIVE CIRCUIT.

Regenerative circuit at the receiver end is used to compensate the signal loss and reconstruct the signal and also increase its strength.

DECODER.

The decoder circuit decodes the pulse coded waveform to reproduce the original (analog) signal. This acts as a demodulator for the encoded output

RECONSTRUCTION FILTER.

~~After~~ the decoded signal (analog signal) is fed to the reconstruction filter or LPF which retains the original (message) signal by eliminating the higher frequency components..

NOISES IN PCM System.

There are two types of noises produced in PCM system.

→ Quantization Noise.

→ Channel Noise.

QUANTIZATION NOISE.

Quantization noise is produced in quantizer due to the rounding off of the amplitude. The difference between the original amplitude and the quantized amplitude is known as quantization noise or error.

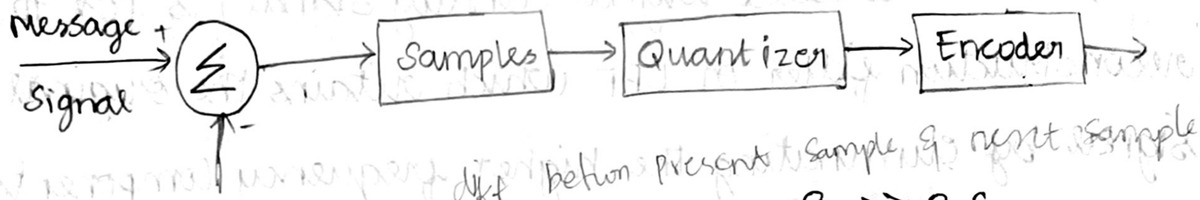
This can be eliminated by increasing the quantization level (L)

CHANNEL NOISE.

The channel introduces distortion in the signal during transmission. This distortion can be eliminated by the regenerative repeater in order to provide a distortionless output (replica of message signal).

DIFFERENTIAL PULSE CODE MODULATION (DPCM)

more no of samples in DPCM



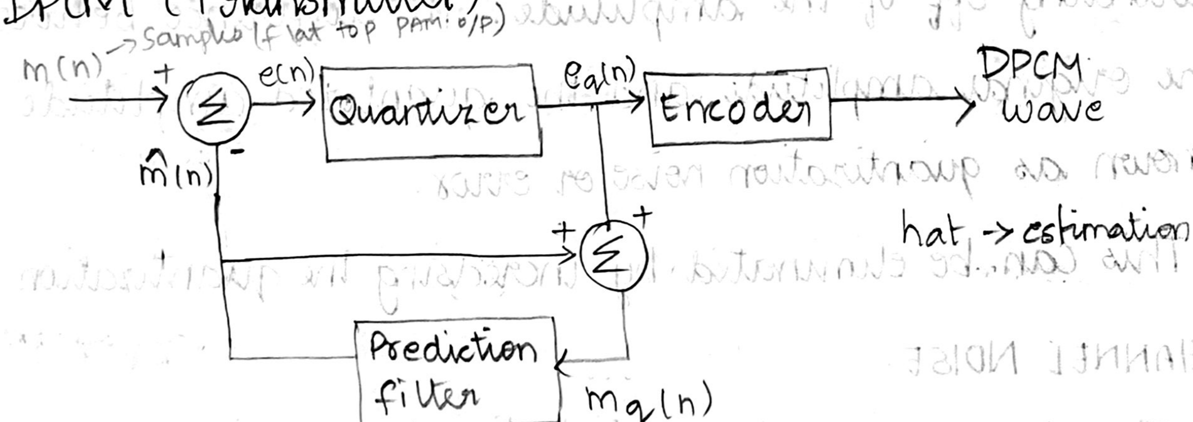
→ Condition to perform DPCM ⇒ $f_s \gg 2f_m$.

→ DPCM is suitable for voice and video signals

There is a high correlation between the first and second sample (adjacent) i.e the signal does not undergo rapid change from one sample to the next sample; in such cases the difference between the adjacent signal is quantized and encoded.

Prediction filter → predicts the next sample value.

DPCM (Transmitter)



$$e(n) = m(n) - \hat{m}(n) \quad \text{—— (1)}$$

$$\hat{m}(n) = m(n) - e(n) \quad \text{—— (1a)}$$

$$e_q(n) = e(n) + q(n) \quad \text{—— (2)}$$

$$M_q(n) = e_q(n) + \hat{m}(n) \quad \text{—— (3)}$$

$$M_q(n) = e(n) + q(n) + m(n) - e(n)$$

$$M_q(n) = q(n) + m(n) \quad \text{—— (4)}$$

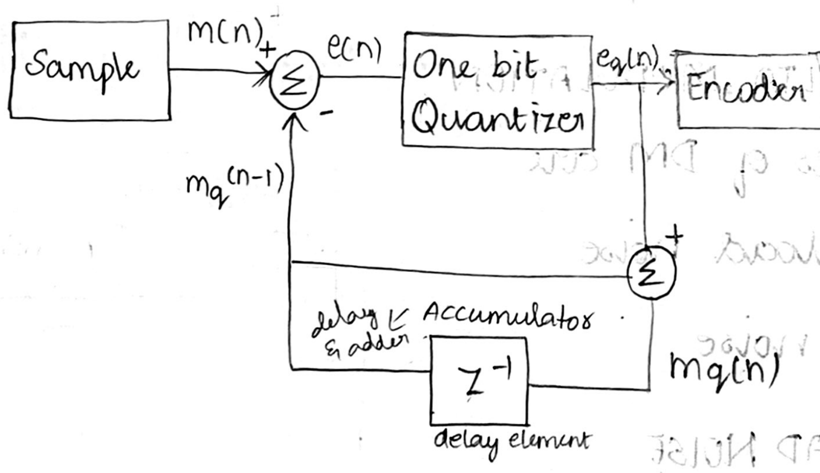
from (4) we understand that the ~~quantized~~ sample the $M_q(n)$ differs from the quantization error $q(n)$. If the prediction filter works good then the error $e(n)$ will be negligible.

29/9/2022

✱ DELTA MODULATION (1 bit modulation) $L = 2^1 = 2$. diff b/w present & previous val.

→ Also called 1 bit modulation

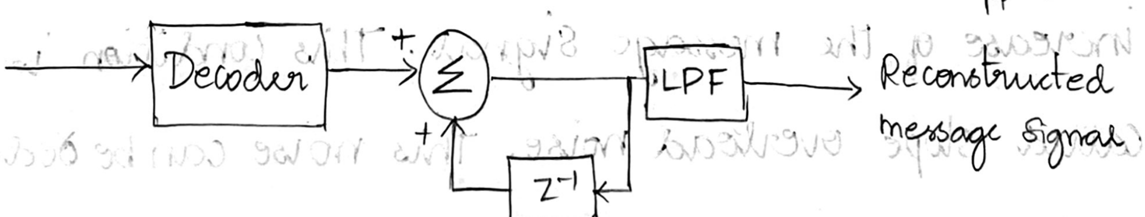
→ Difference between present sample and previous ~~sample~~ quantized value is Quantized. Quantizer error is more than DM



step size increases encoder o/p is 1

decreases encoder o/p is 0.

DM RECEIVER.



$e(n) = m(n) - m_q(n-1) = +ve$ step size ↑ by Δ.

$e(n) = m(n) - m_q(n-1) = -ve$ step size ↓ by Δ.

To reduce Quantization error the step size should vary according to the message signal.

Devices in delta modulation.

Comparator, Accumulator (delay element & summer),

One bit Quantizer, encoder.

$$e(n) = m(n) - m_q(n-1)$$

$$e_q(n) = \Delta \, \mathrm{Sgn}(e(n))$$

Encoder output is 1 for $e_q(n) = +\Delta$

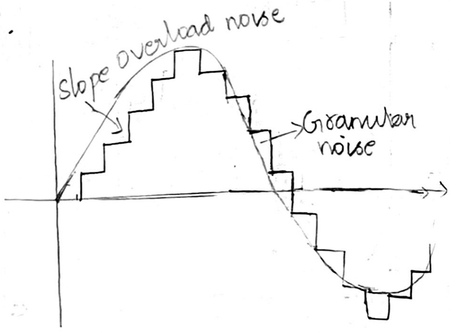
Encoder output is 0 for $e_q(n) = -\Delta$.

NOISE IN DELTA MODULATION.

The two types of DM are

→ slope overload noise

→ Granular noise



SLOPE OVERLOAD NOISE

The step size Δ is too small to follow the steep increase of the message signal. This condition is called slope overload noise. This noise can be reduced by having varying step size.

GRANULAR NOISE.

The step size Δ is too large relative to the slope of the message signal such condition is called Granular noise.

CONDITION TO REDUCE SLOPE OVERLOAD NOISE AND GRANULAR NOISE.

$$\frac{\Delta}{T_s} \geq \max \left| \frac{dm(t)}{dt} \right|$$

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

*1. Given a sine wave of freq. f_m and amplitude A_m applied to a delta modulation having step size Δ show that the slope overload distortion will occur if ~~the~~ Amplitude is $A_m \geq \frac{\Delta}{2\pi f_m T_s}$

$$\frac{\Delta}{T_s} \geq \max \left| \frac{d\,m(t)}{dt} \right|$$

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

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

$$\frac{\Delta}{T_s} \geq \max \left| \frac{d(A_m \sin \omega_m t)}{dt} \right|$$

$$\frac{\Delta}{T_s} \geq \max |A_m \cos \omega_m t \cdot \omega_m|$$

$$\frac{\Delta}{T_s} \geq A_m \cdot \omega_m$$

[∵ max value of $\cos \omega_m t = 1$]
$\cos 0° = 1$.

$$A_m \leq \frac{\Delta}{T_s \cdot \omega_m}$$

$$A_m < \frac{\Delta}{2\pi f_m T_s}$$

Slope overload distortion will occur if $A_m \geq \frac{\Delta}{2\pi f_m T_s}$.

~~Consider~~

SNR OF DELTA MODULATION

We know that, $A_m < \frac{\Delta}{2\pi f_m T_s}$ slope overload will not occur.

The maximum value of the message signal power is given by, $P = \frac{\left(\frac{A_m}{\sqrt{2}}\right)^2}{R}$. Assume $R = 1$

$$P = \frac{A_m^2}{2} \quad \text{——— (1)}$$

The quantization error in delta modulation differs by $+\Delta$ or $-\Delta$. So the uniform distribution is given by.

$$f(q_e) = \begin{cases} \frac{1}{\Delta-(-\Delta)} & , -\Delta < q_e < \Delta \\ 0 & , \text{otherwise} \end{cases}$$

$$= \begin{cases} \frac{1}{2\Delta} & , -\Delta < q_e < \Delta \\ 0 & , \text{otherwise.} \end{cases}$$

$$E[q_e] = \int_{-\Delta}^{\Delta} q_e \cdot f(q_e) \cdot de$$

$$= \int_{-\Delta}^{\Delta} \frac{1}{2\Delta} \cdot e \cdot de$$

$$= \frac{1}{2\Delta} \int_{-\Delta}^{\Delta} e \cdot de = \frac{1}{2\Delta} \left[\frac{e^2}{2} \right]_{-\Delta}^{\Delta}$$

$$E[q_e] = 0.$$

$$E[q_e^2] = \int_{-\Delta}^{\Delta} e^2 \cdot \frac{1}{2\Delta} \cdot de$$

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

$$= \frac{1}{2\Delta} \cdot \left[\frac{\Delta^3}{3} + \frac{\Delta^3}{3} \right]$$

$$= \frac{1}{2\Delta} \times \frac{2\Delta^3}{3}$$

$$= \frac{\Delta^2}{3}.$$

Noise Power ; Variance $= E[q_e^2] - E[q_e]^2$

$$= \frac{\Delta^2}{3} - 0$$

$$\text{Noise Power} = \frac{\Delta^3}{3}$$

$$\text{Signal power} = \frac{A_m^2}{2} = \frac{\frac{\Delta^2}{4\pi^2 f_m^2 T_s^2}}{2} = \frac{\Delta^2}{8\pi^2 f_m^2 T_s^2} = \frac{\Delta^2 f_s^2}{8\pi^2 f_m^2}$$

$$\text{Noise power} = \frac{\Delta^2}{3}$$

$$SNR = \frac{\text{Signal Power}}{\text{Noise Power}}$$

$$= \frac{\frac{\Delta^2 f_s^2}{8\pi^2 f_m^2}}{\frac{\Delta^2}{3}} = \frac{\Delta^2 f_s^2}{8\pi^2 f_m^2} \times \frac{3}{\Delta^2}$$

$$SNR = \frac{3 f_s^2}{8\pi^2 f_m^2}.$$

The normalized noise power is obtained by multiplying $\frac{\Delta^2}{3}$ by $\frac{f_m}{f_s}$.

$$\text{normalized SNR} = \frac{3 f_s^3}{8\pi^2 f_m^3}$$

Space (S)
Mark (M)

LINE CODING TECHNIQUE

RZ → Return to Zero.

1. Unipolar NRZ → 1 = , 0 → Switch of the pulse
2. Polar NRZ → 1 = , 0 =
3. Bi-Phase Manchaster Coding → 1 = , 0 =
4. Unipolar RZ → 1 = , 0 =
5. Bi-Polar RZ → 1 = , 0 =

If you see the word mark in the bracket if the next bit is 1 from the current state change the state 0 means no change in state irrespective of the type of data format.

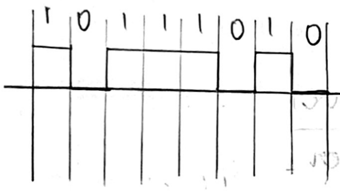
Bipolar RZ → mark

Biphase manchaster (M) 1 0 1 1 0

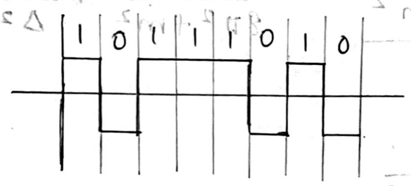
1011010

Homework. (10111010)

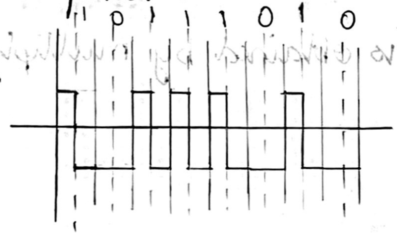
1. Unipolar NRZ.



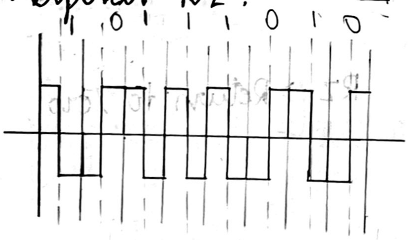
2. Polar NRZ.



3. Unipolar RZ.



4. Bipolar RZ.

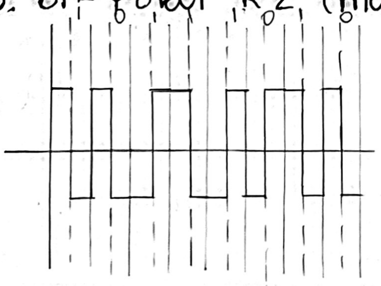


5. Bi-Phase Manchester Coding :

6. Bi-Polar RZ (mark) :-

1 $\xrightarrow{next}$ Change in state

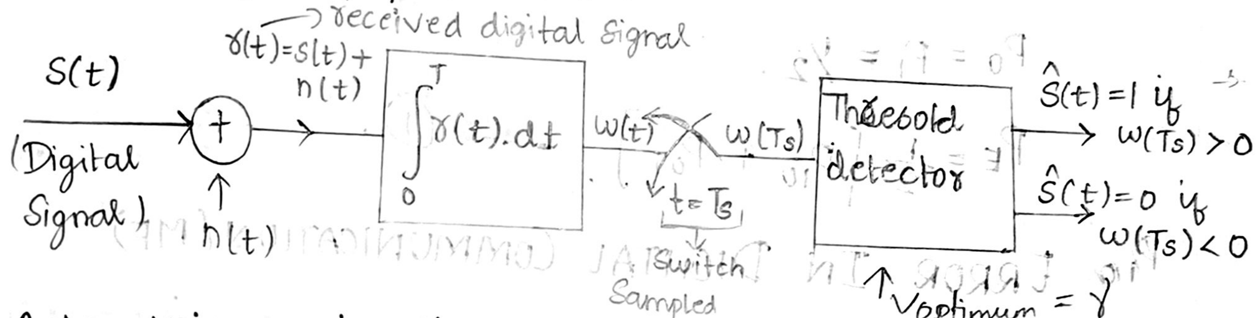
0 $\xrightarrow{next}$ same state



In DC error is
1 interpreted as 0
or 0 interpolated as 1

Matched Filter : (Receiver for DC) DC – Digital Communication

$$s(t) = \begin{cases} A, & \text{for } 0 < t < T \\ 0, & \text{for } 0 < t < T \end{cases}$$



After this o/p is given to the receiver.

ISI (Inter Symbol Interference)

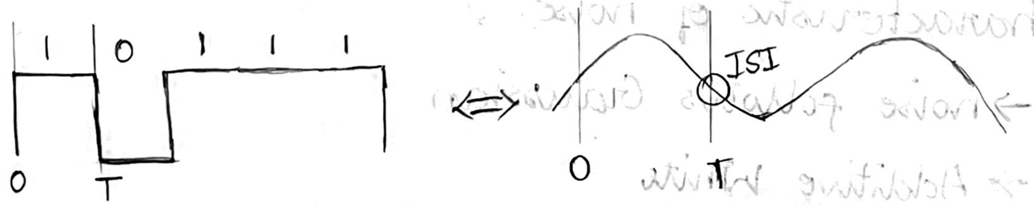
Aliasing in analog communication (overlapping of adjacent samples during the sampling process).

$f_s < 2f_m \rightarrow$ Undersampling & Aliasing.

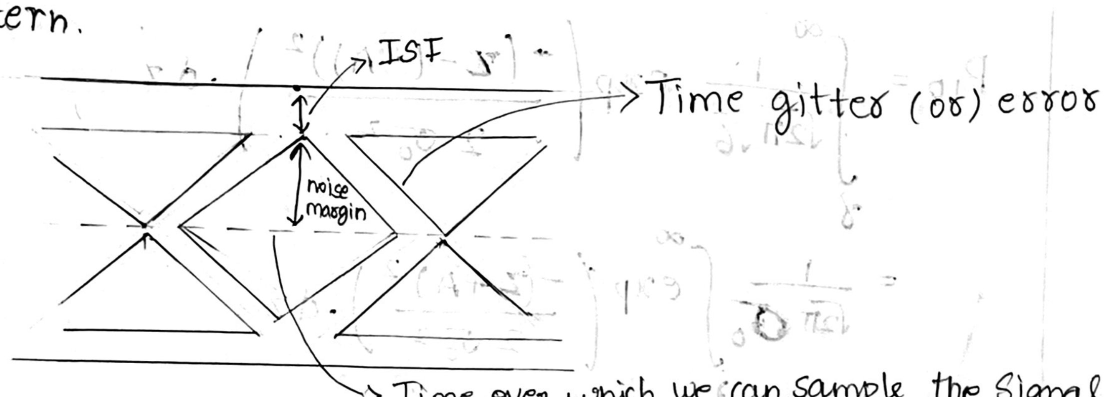
ALIASING: Whenever a signal is sampled below Nyquist rate ($f_s > 2f_m$) (or) undersampling condition, then the signal is undersampling & result is aliasing effect.

ISI (Inter Symbol Interference)

Whenever a digital signal is transmitted through a dispersive channel. The pulses are no longer confined to their time slots, when they arrive at the receiver and they spill over into adjacent time slot.



Eye Pattern.



Digital Communication → Receiver

erfC → Complimentary error function.

PROBABILITY OF ERROR FOR MATCHED FILTER.

$$P_E = \underset{Tx}{P_0}\ \underset{RT}{P_{10}} + \underset{Tx}{P_1}\ \underset{RT}{P_{01}}$$ γ → optimum thresold.

$$P_0 = P_1 = 1/2.$$

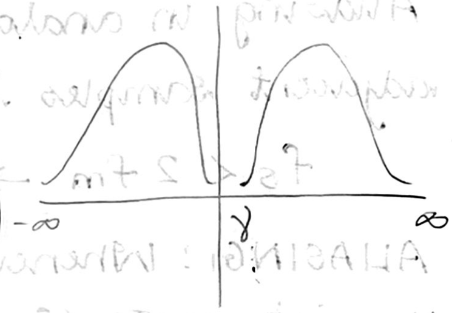
$$P_E = \frac{1}{2}\left[P_{10} + P_{01}\right]. \quad \text{—— (1).}$$

P_{10}. ERROR IN DIGITAL COMMUNICATION (MF)

Error occurs when we transmit 1 MF interprets as 0, and if transmitted 0 is interpreted as 1.

$$P_{10} = \int_{\gamma}^{\infty} f(z|0)\,.dz.$$

$$f(z|0) = \frac{1}{\sqrt{2\pi}\,\sigma_0}\exp\left(\frac{-(x-\mu)^2}{2\sigma_0^2}\right)$$



σ_0^2 ⇒ Noise power $= \frac{N_0}{2}$

In Digital Communication noise is uniform over the bit period T (or) T_b.

μ ⇒ peak amplitude of binary data.

+A ⇒ for 1

−A ⇒ for 0.

Characteristic of noise.

→ noise follows Gaussian

→ Additive White

$$P_{10} = \int_{\gamma}^{\infty} \frac{1}{\sqrt{2\pi}\sqrt{6}} \exp\left(\frac{-(z-(-A))^2}{2\sigma_0^2}\right).dz$$

$$= \frac{1}{\sqrt{2\pi}\,\sigma_0}\int_{\gamma}^{\infty} \exp\left(\frac{-(z+A)^2}{2\sigma_0^2}\right).dz.$$

σ

$$= \frac{1}{\sqrt{2\pi}\sqrt{\frac{N_0}{2T_b}}} \int_{\gamma}^{\infty} exp\left(\frac{-(z+A)^2}{2\frac{N_0}{2T_b}}\right) \cdot dz . \quad e^{-x} \quad -x$$

$$P_{10} = \frac{1}{\sqrt{\frac{\pi N_0}{T_b}}} \int_{\gamma}^{\infty} exp\left(\frac{-(z+A)^2}{\frac{N_0}{T_b}}\right) \cdot dz \quad e \quad y^2 = x$$

$$y = \frac{z+A}{\sqrt{N_0/T_b}} . \qquad z=\gamma,\ y = \frac{\gamma+A}{\sqrt{N_0/T_b}}$$

$$dy = \frac{dz}{\sqrt{N_0/T_b}} ,\ dz = dy\sqrt{\frac{N_0}{T_b}} \qquad z=\infty,\ y=\infty$$

$$P_{10} = \frac{1}{\sqrt{\frac{\pi N_0}{T_b}}} \int_{\frac{\gamma+A}{\sqrt{N_0/T_b}}}^{\infty} exp(-y^2) .dy . \sqrt{\frac{N_0}{T_b}}$$

$$P_{10} = \frac{1}{\sqrt{\frac{\pi N_0}{T_b}}} \cdot \sqrt{\frac{N_0}{T_b}} \int_{\frac{\gamma+A}{\sqrt{N_0/T_b}}}^{\infty} exp(-y^2) .dy .$$

$$erf(u) = \frac{2}{\sqrt{\pi}} \int_{u}^{\infty} exp(-y^2).dy$$

$$P_{10} = \frac{1}{2} erf\left(\frac{\gamma+A}{\sqrt{N_0/T_b}}\right). \quad \text{—— } (2)$$

Similarly $P_{01} = \frac{1}{2} erf\left(\frac{-\gamma+A}{\sqrt{N_0/T_b}}\right).$ —— (3).

~~Sub (2) & (3) in (1).~~

γ is the optimum thresold which is equal to 0 if we transmit +A to −A .

$$P_{10} = \frac{1}{2} erf\left(\frac{\cancel{\gamma +} A}{\sqrt{N_0/T_b}}\right) \quad \text{—— } (4)$$

$$P_{01} = \frac{1}{2} erf\left(\frac{A}{\sqrt{N_0/T_b}}\right) \quad \text{—— } (5)$$

Sub 4 & 5 in 1.

$$P_E = \frac{1}{2}\left[\mathrm{erf}\left(\frac{A}{\sqrt{N_0/T_b}}\right)\right].$$

Created with Scanner Pro