UNIT-2

WAVEFORM CODING.

Prediction filtering → Linear Prediction.

Special form of estimation → prediction
↓
finite set of present & past samples to predict the future sample.

Prediction → linear ⇒ combination of the given samples of the process is linear

Filter → prediction ⇒ Predictor

Define prediction error.

actual value of the process − predictor o/p = prediction error

mean square value of the prediction error is minimized by designing the predictor using Wiener filter theory

Consider a random samples $x_{n-1} \dots x_{n-m}$ from the stationary process $x(t)$. prediction sample x_n.

$$\hat{x}_n = \sum_{k=1}^{M} h_{ok}\, x_{n-k}$$

h_{ok} → optimum predictor coeff.

Write the special case of wiener filter.

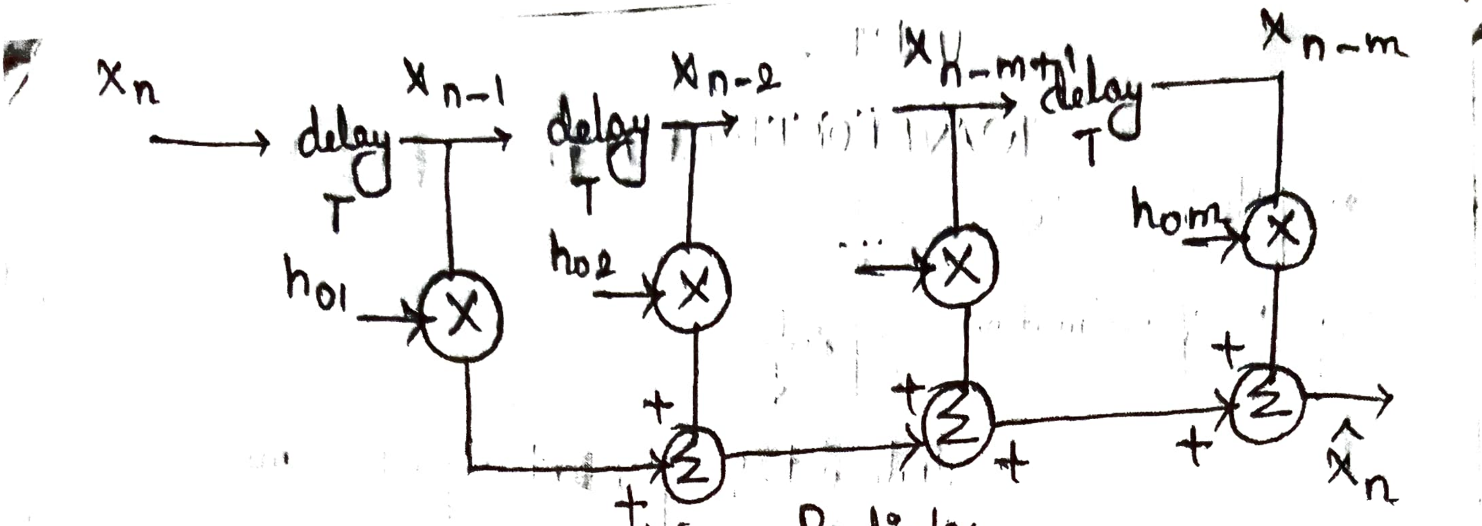
Special case → Wiener filter.

1) Variance of sample x_n $\sigma_x^2 = E(x_n^2) = R_x(0)$
→ mean = 0

2) ~~corr~~ cross correlation fn $E(x_n x_{n-k}) = R_x(k)$

3) auto correlation fn $E(x_{n-k}\, x_{n-m}) = R_x(M-k)$

Prediction $R_x(k) = \sum_{m=1}^{M} h_{om} R_x(m-k)$

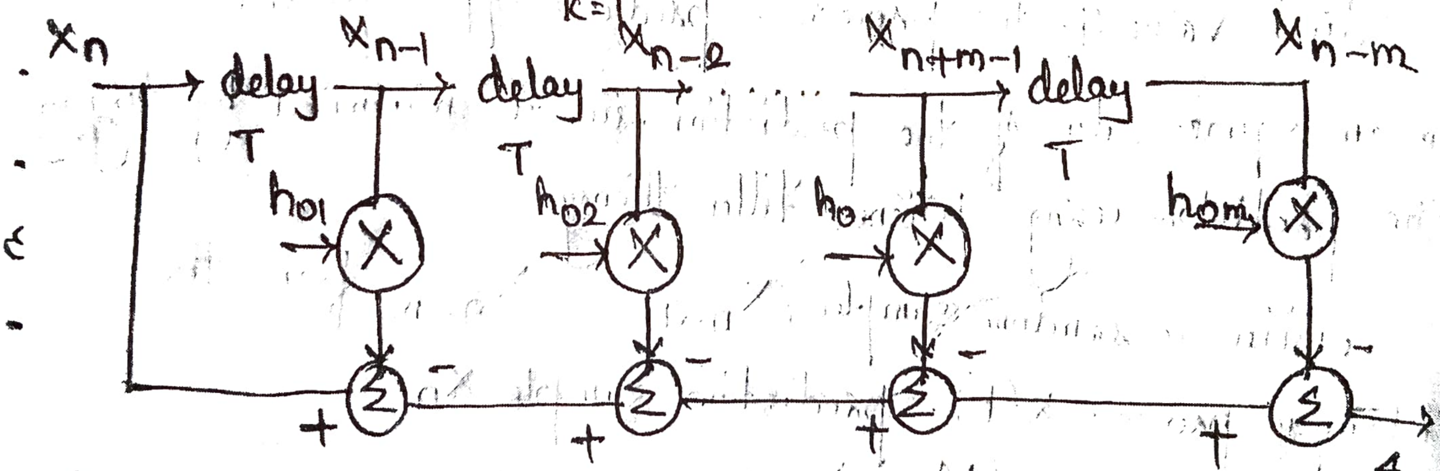


Linear Predictor

Prediction Error $\varepsilon_n = x_n - \hat{x}_n$

$$= x_n - \sum_{k=1}^{M} h_{0k}\, x_{n-k}$$

$$x_n = \sum_{k=1}^{M} h_{0k}\, x_{n-k} + \varepsilon_n$$



Present sample of the process x_n is a linear combination of past samples of the process + the prediction error ε_n.

Prediction Error Variance $\sigma^2_\varepsilon = E(\varepsilon_n^2)$

$$= R_x(0) - \sum_{k=1}^{M} h_{0k} R_x(k).$$

Prop of Linear Prediction: List out the prop of linear prediction.

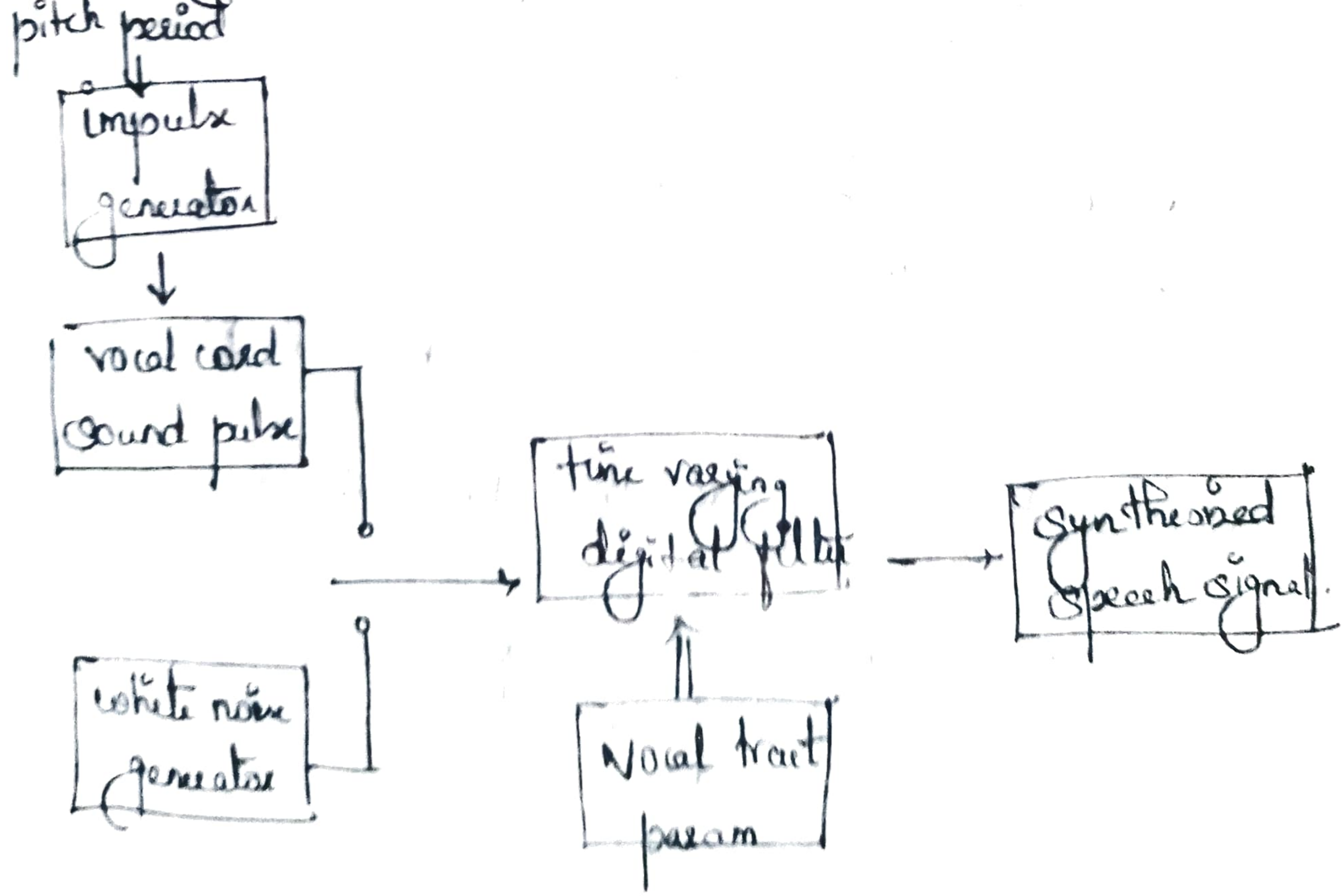
1) Prediction error variance decreases with increase predictor order → design of waveform coders.

2) When the prediction error variance reaches its min value, the prediction error process assume the form of white noise. Prediction error filter → whitening filter → design of source coders.

Linear Predictive Vocoders.

→ Linear Predictive Vocoding → linear prediction for the digitization of speech signal.

→ Vocoding → contraction of voice coding, device → vocoder



Speech Production Process

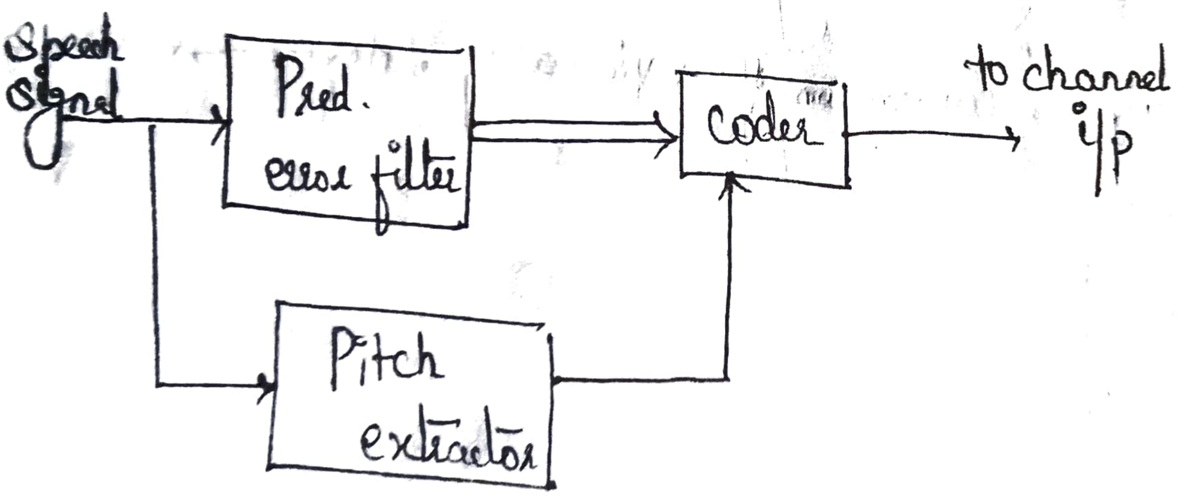
→ The sound generating mechanism is linearly separable from the vocal tract filter.

→ Excitation depends on whether sound is voiced / unvoiced

voiced: as the response of the vocal tract filter excited with a periodic seq of impulse response.

unvoiced: as the response of the vocal tract filter excited with a white noise seq.

Tx



→ Tx performs the analysis on the i/p speech signal block by block.

→ Each block is 10-30m long as speech production is a stationary process.

→ parameters:
- Prediction error filter coefficient
- voiced / unvoiced parameter
- pitch period.

[illegible] → Encoder → Speech Synthesizer → reproduction

[illegible]

— [illegible] sequence [illegible] followed by synthesis of speech signal.

— Result — artificial sounding reproduction of the signal.

— poor reproduction quality tolerated upto 4 kbps.

Explain the [illegible] of DPCM and [illegible] one of it.

Diff Pulse Code Modulation.

DPCM

— [illegible] of a [illegible] signal the signal is sampled at a rate [illegible] higher than the Nyquist rate

— sampled signal contains a high correlation between adjacent samples

— signal doesnot change rapidly from one sample to next.

— Smaller than the signal [illegible]

— highly correlated samples are encoded

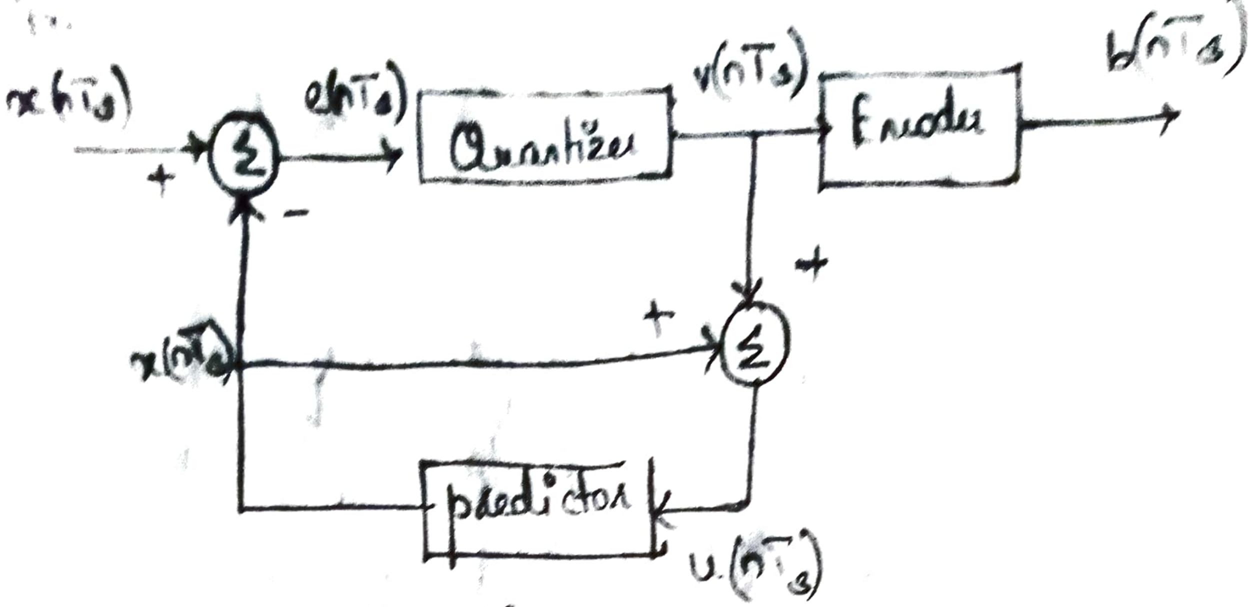
— Resulting encoded signals have redundant info.

[illegible] samples are highly correlated with each other as the signal does not change [illegible]

[illegible] the principle of DPCM?

The redundancy is reduced [illegible] bit rate will be [illegible]

[illegible]



→ Principle → Prediction

↳ value of present sample is predicted from previous sample. prediction may not be exact but it is very close to the actual value.

→ Sampled signal : $x(nT_s)$ Predicted signal : $\hat{x}(nT_s)$

Comparator → difference between actual and predicted sample value called error signal.

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

→ Error difference between $x(nT_s)$ and $\hat{x}(nT_s)$ and predicted value is produced by using a prediction filter

→ Quantizer o/p $e_q(nT_s)$ and $\hat{x}(nT_s)$ are added & given as i/p to the prediction filter $x_q(nT_s)$. Quantized error signal $e_q(nT_s)$

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

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

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

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

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

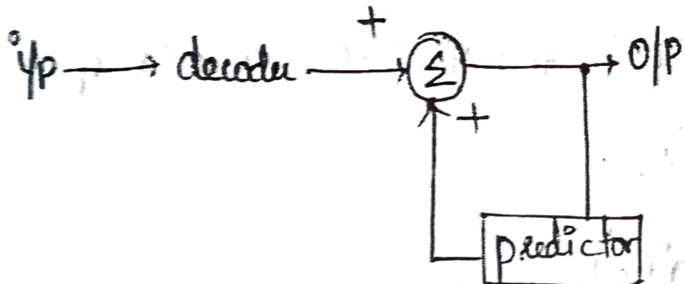
$$\therefore\ x_q(nT_s) = x(nT_s) + q(nT_s)$$

↓

Quantized signal is the sum of original sample value & quantization error

└→ +ve / -ve & does not depend on the prediction filter characteristics.

Rx.



→ Consists of decoder to reconstruct the original signal

→ Quantized signal is reconstructed from the o/p using the same predictor as in Tx.

→ absence of noise → rx i/p is identical to the txr o/p.

→ the rx o/p is equal to $x_q(nT_s)$ that differs from $x(nT_s)$ only by the quantizer o/p $q(nT_s)$.

SNR. $SNR_o = \dfrac{\sigma^2 x}{\sigma^2 Q}$ $\sigma^2 x$ → variance of i/p $x(nT_s)$ with 0 mean

$\sigma^2 Q$ → variance of quantization error $q(nT_s)$

Rewritten as

$$SNR_o = \frac{\sigma^2 x}{\sigma^2 E} \cdot \frac{\sigma^2 E}{\sigma^2 Q}$$

$\sigma^2 E$ → Variance of prediction error $e(nT_s)$

prediction gain $G_p = \frac{\sigma^2_x}{\sigma^2_E}$ $G_p > 1 \rightarrow$ gain of SNR.

→ baseband signal variance $\sigma^2 x$ is fixed so G_p is max by min $\sigma^2 E$ of the prediction error $e(nT_s)$.

Explain the principle of ADPCM with neat diagram.

<u>ADPCM</u> (Adaptive differential pulse code mod)

→ digital coding scheme that uses both adaptive quantization & adaptive prediction is called ADPCM.

→ adaptive quantization → quantiser that operates with time varying step size (ΔnT_s).

T_s → sampling period.

Δ → is varied so as to match the variance $\sigma^2 x$ of the i/p signal $x(nT_s)$.

$$\Delta(nT_s) = \phi \, \hat{\sigma}_x(nT_s)$$

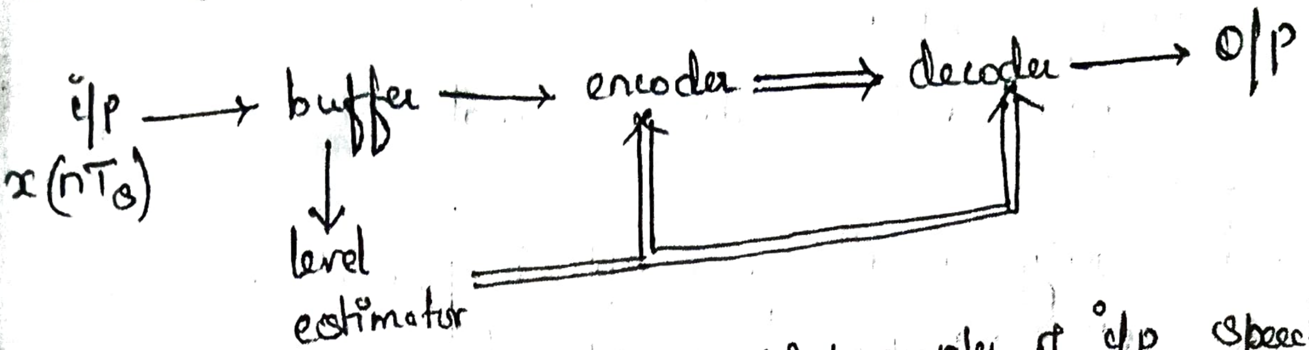
ϕ → const, $\hat{\sigma}_x(nT_s)$ → estimate of SD $\sigma_x(nT_s)$.

Problem → continuously find $\sigma_x(nT_s)$

To proceed this soln is 2 ways

- forward → unquantized samples of i/p
- backward → quantized samples of o/p.

AQF. (adap quanti with forward estimation). Define APF



→ $x(nT_s)$ buffered by the unquantized samples of i/p speech signal.

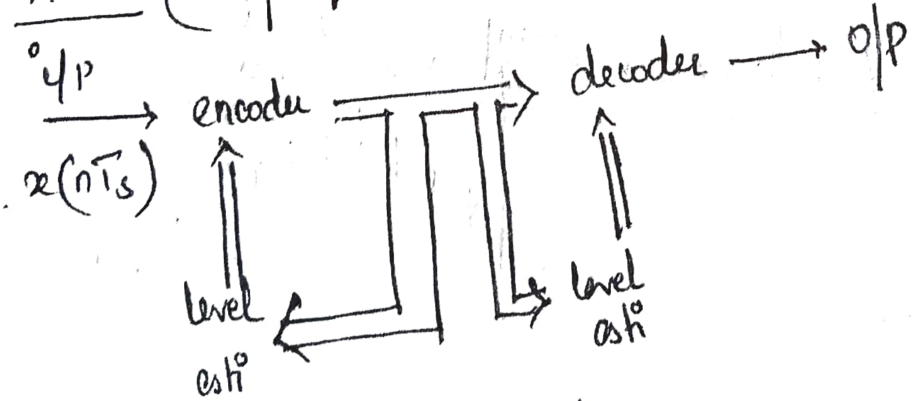
→ samples are released after estimate $\hat{\sigma}_x(nT_s)$. Independent of quan noise.

→ step size is reliable the AQB.

→ problem of level txn, buffering & delay intrinsic are avoided in AQB.

→ Quantizer o/p to extract info for the computation of stepsize $\Delta(nT_s)$.

AQB. (adap quanti with backward estimation). Define APB.



→ Non linear feedback system.

→ Stable if the quantizer i/p $x(nT_s)$ is bounded ∀ $\hat{\sigma}_x(nT_s)$ estimate the $\Delta(nT_s)$.

DM (delta modulation). What is DM?

Explain delta modulation in detail with neat diagram.

→ 1 bit version of DPCM

→ Staircase approximation to the over sampled version of i/p baseband signal.

→ transmits only one bit per sample. ie present sample value is compared with the previous sample value and indication whether the amp is increased | decreased.

→ i/p signal $x(t)$ is approximated to step signal by DM. Step size is fixed.

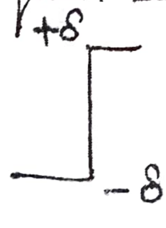
→ diff b/w i/p & approximation is quantized to 2 levels ie $\pm\delta$

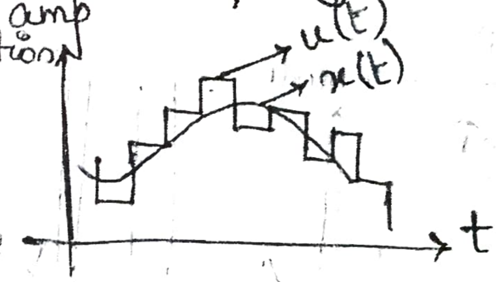
$+\delta \to +ve \to$ step size $\uparrow \to$ 1 sent

$-\delta \to -ve \to$ step size $\downarrow \to$ 0 sent

→ Staircase approxi remains within $\pm\delta$ the i/p signal.

→ δ → absolute value of 2 representation levels of one bit quantizer





→ step size $\Delta = 2\delta$

→ error b/w the sampled value of $x(t)$ and staircase approximation $u(t)$

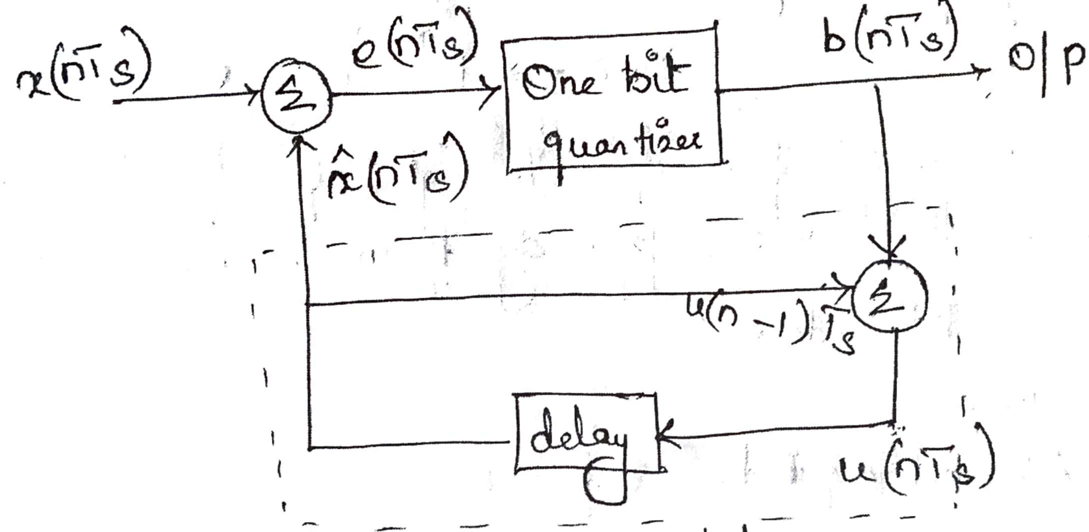
$$e(nT_s) = x(nT_s) - \hat{x}(nT_s)$$

$e(nT_s)$ ↓ error at present sample; $x(nT_s)$ ↓ sampled signal; $\hat{x}(nT_s)$ ↓ last sample i.e. $e(nT_s) = x(nT_s) - \underbrace{u(nT_s - T_s)}_{\text{present sample approximation}}$

$$b(nT_s) = \delta \, \mathrm{sgn}\left[e(nT_s)\right]$$

$$u(nT_s) = u(nT_s - T_s) + b(nT_s).$$

Tx.



→ a summer, quantizer & accumlator → initially set to 0.

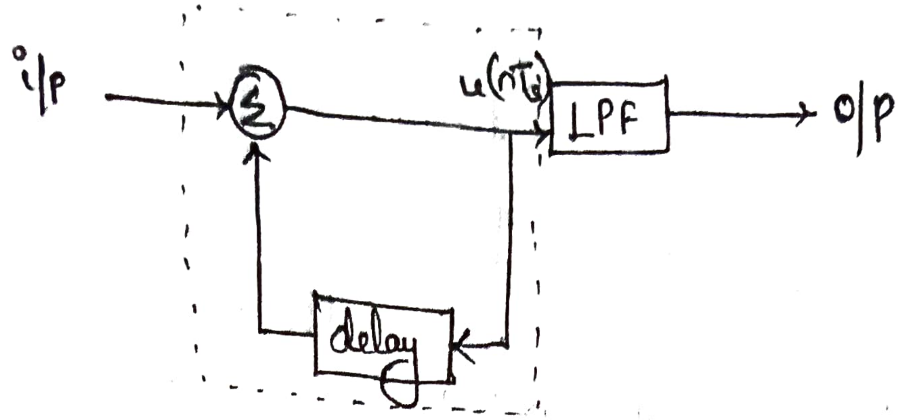
→ accumlator adds quantizer o/p with the previous sample approxi

$$u(nT_s) = u(nT_s - T_s) + b(nT_s)$$

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

$$b(nT_s) = \begin{cases} +\delta & x(nT_s) > \hat{x}(nT_s) \\ -\delta & x(nT_s) < \hat{x}(nT_s) \end{cases}$$

Rx.



accumlator

2 parts
- accumlator → generates staircase approxi signal & delay by 1 bit sampling period.
  - $+\delta \to 1$ $\quad -\delta \to 0$
- LPR
  - acts as cut off freq equal to highest freq $x(t)$.
  - Smoothen the staircase appro signal and reconstructed to $x(t)$

Features
- → One bit codeword.
- → simple
- → ADC not required.

Quan Noise What is slope overload distortion? How is it reduced

(1) slope overload distortion.

Let $q(nT_s)$ be the error signal

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

prediction error signal

$$e(nT_s) = x(nT_s) - x(nT_s - T_s) + q(nT_s - T_s)$$

→ Error arises bcoz of large dynamic range of i/p signal.

→ Staircase signal cannot approximate step size so there is a large diff b/w the staircase signal & ori signal, error is called slope overload distortion.

② Granular Noise. What is granular noise?

→ step size is too large when compared to i/p signal, the staircase signal is changed by large amount (δ) bcoz of large step size.

→ i/p signal is flat → $u(t)$ is oscillating by $\pm\delta$.

→ error b/w the i/p & approx signal is called granular noise.

→ to ↓ step size → granular = error b/w step size & i/p.

What is ADM? Explain ADM in detail with neat dia. Compare DM & ADM.

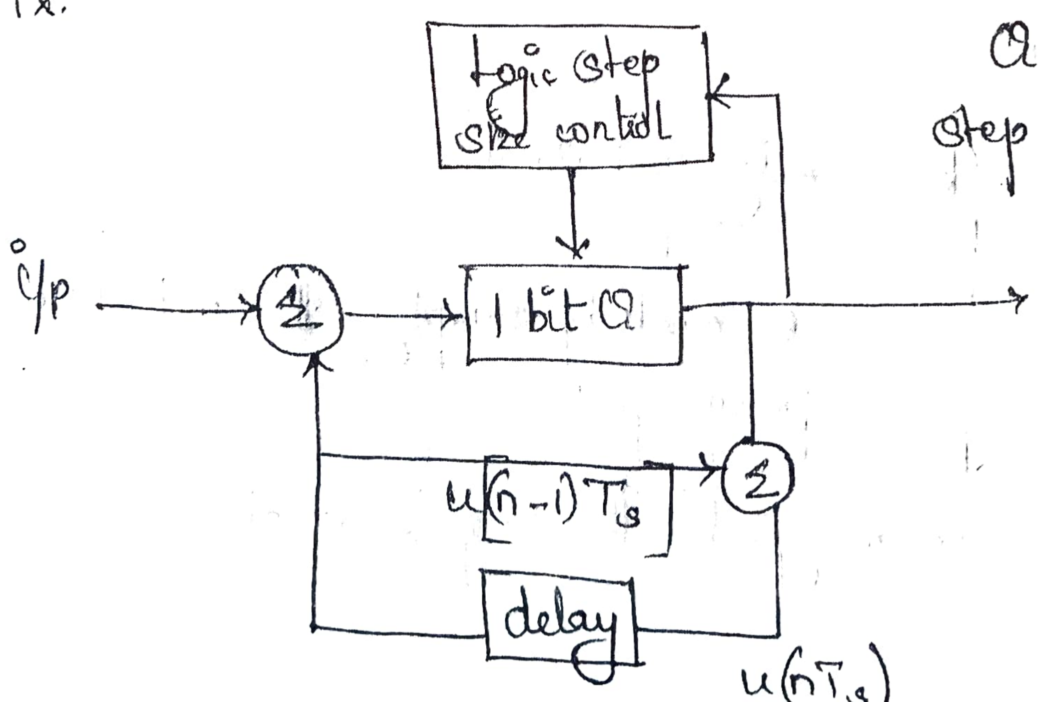
ADM. (adaptive delta modulation)

→ Performance of DM ↑ by making stepsize of modulator be a time varying form.

① Step size is ↑ ② i/p sig varies slowly stepsize is ↓.

→ 2 scheme to adjust step size
- → discrete set of values are provided.
- → con. range of step size variation is provided.

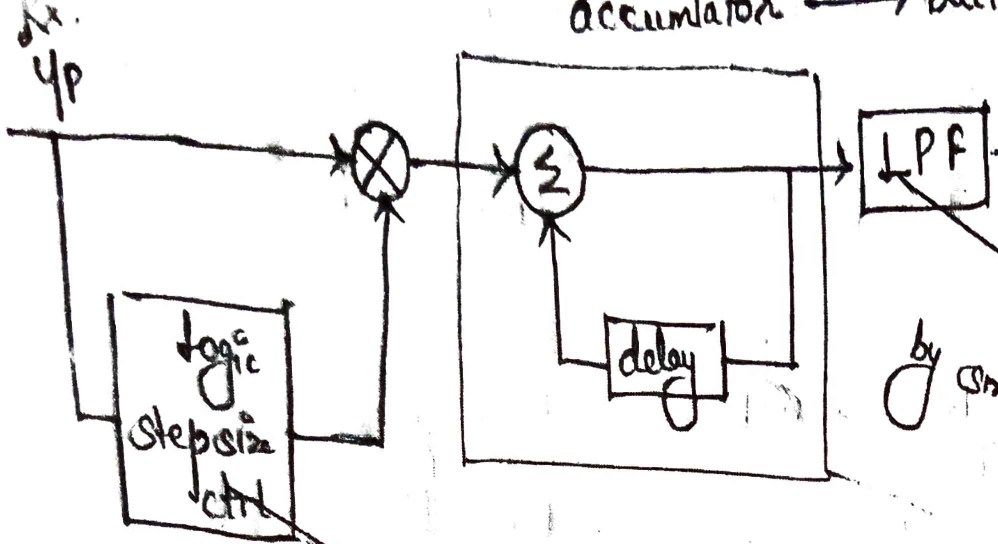
Tx.



Q o/p → 1 → step size is doubled

o/p → 0 → o/p stepsize is reduced by 1.

accumulator → builds staircase approx.



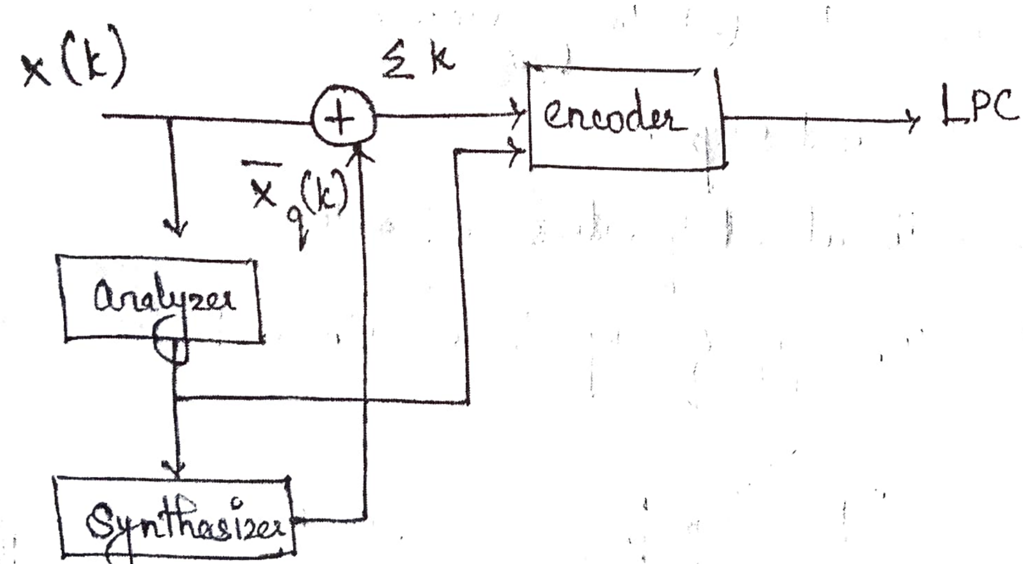
reconstruct the signal by smoothens the staircase.

generates a step size from each i/p signal & present i/p decides the step size.

Explain the principle of LPC.

LPC (Linear Predictive Coding) What is the principle of LPC?

Tx.



Speech signal x(k) → digitized signal.

↓

Obtained by sampling the cont. time speech signal.

↓

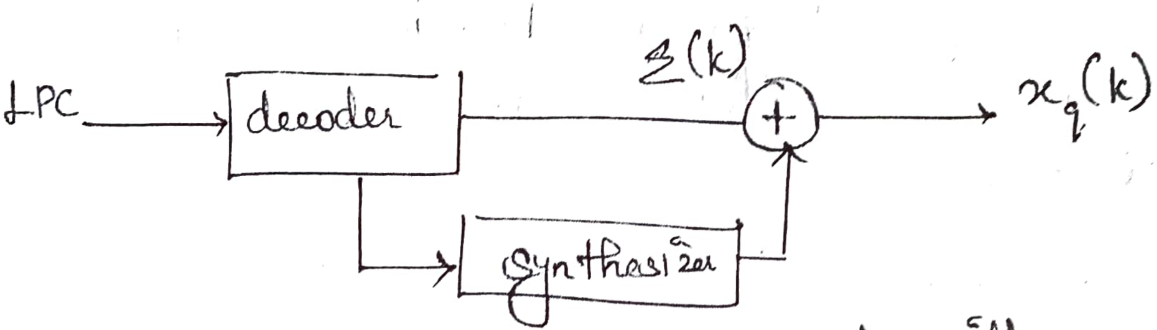
Seq of speech signals that applied to the analyzer

analyzer → determines the synthesizer parameter.
↓
reconstructs the signal $\hat{x}(k)$.

$\hat{x}_q(k)$ and $x(k)$ compared with each other to determine the error $e(k)$.

analyzer param & error signal → multiplexed & txn.
↓
sig called LPC.

Rx.



LPC applied to the decoder → seperates filter parameters and ε_k given to the synthesizer. Synthesizer o/p + ε_k = $x_q(k)$

Explain about line codes types / binary data formats.

Line Codes. What is line coding?

→ digital msg → ordered ser of symbols produced by a discrete info source. M = 2 → symbols 0 & 1.

→ Way of representing each of the discrete set of values in a particular set of discrete events called codes.

→ when digital data are transmitted through a band limited channel, dispersion in the channel causes an overlap in time b/w symbols. form of distortion is called ISI.

→ Line Coding → technique of converting digital to digital signals | method of representing binary info interms of seq of voltage pulses.

Elements of line codes } → Tx, Rx, Spec of line code, BW, noise, implementation cost & performance char.

<u>Need / Objectives</u>. When we need line coding schemes?

→ info are discrete

→ when signal gets dispersed over bandlimited channel.

→ Overlaps & distortions causes

→ distortion is ISI.

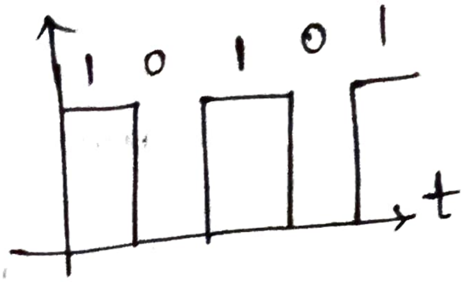
<u>Binary data format.</u>

1. On off signalling / Unipolar.
2. Non Return to Zero
3. Return to Zero
4. Bipolar
5. Manchester
6. Differential Encoder
7. Quaternary

① On off Signalling

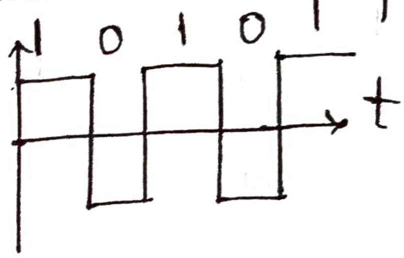
→ Unipolar

→ { 0 → switch off pulse.
  1 → transmitting pulse.



② NRZ

→ pulse occupies the full symbol duration then unipolar called NRZ.



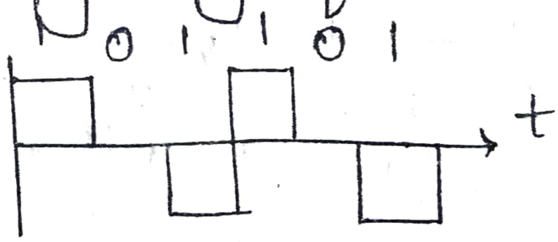
③ RZ

→ pulse occupies the fraction of symbol duration then called RZ.

④ Bipolar

→ pseudoternary signalling 3 amp levels
  → +ve | -ve → alternating 1's
  → no → 0's

→ absence of dc component → contains long strings of 0's & 1's

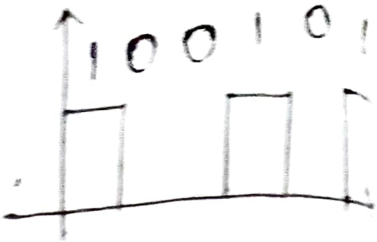


(5) Manchester

→ biphase baseband signalling.

One half of the sym duration → 0.

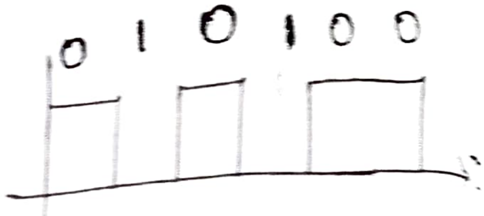
remaining other half → 1.



(6) Differential encoder

→ signal transistion → 0's

signal not transition → 1's

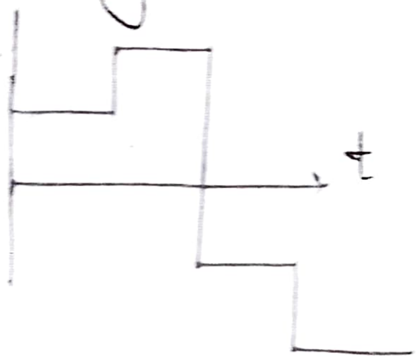


(7) Quaternary

→ grouping the msg bits in blocks of 2 & 4 amp levels

↓

00, 01, 10, 11.



Properties. What are the properties of line codes?

→ txn BW is small

→ power η is small

→ PSD = 1

→ has error ~~capability~~ detection & correction capability

→ prevent long strings of 0's & 1's.

[illegible]

[illegible]

[illegible]

[illegible]

[illegible]

[illegible]

[illegible]

[illegible]

[illegible]

[illegible]

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[illegible]

[illegible]

binary data represented with relation to symbol duration, bit duration is

$$T = T_b \log_2 M$$

auto correlation fn $R_A(n) = E[A_k A_{k-n}]$

PSD of discrete signal is

$$S_x(f) = \frac{1}{T} |v(f)|^2 \sum_{n=-\infty}^{\infty} R_A(n) \exp(-j 2\pi n f T)$$

① NRZ unipolar format.

Suppose 0's & 1's occur with equal probability then

$$P(A_k = 0) = P(A_k = a) = 1/2 \qquad \text{— ①}$$

$n=0 \longrightarrow$ $E(A_k^2) = 0^2 P(A_k = 0) + a^2 P(A_k = a)$

$$= 0 + a^2/2$$

$$E(A_k^2) = a^2/2$$

$n \neq 0 \longrightarrow$ $A_k A_{k-n} = 0, 0, 0, a^2$ having prob $1/4$.

$$E(A_k A_{k-n}) = 3(0)(1/4) + a^2(1/4)$$

$$= 0 + a^2/4$$

$$E(A_k A_{k-n}) = a^2/4$$

autocorrelation fn $R_A(n) = \begin{cases} a^2/2 & n = 0 \\ a^2/4 & n \neq 0 \end{cases}$ — ②

for $v(t)$ basic pulse i.e. rect pulse of amp 1 & duration T_b the fourier transform

$$V(f) = T_b \operatorname{sinc}(f T_b) \quad — ③$$

$$\operatorname{sinc}(\lambda) = \sin(\pi \lambda)/\pi\lambda .$$

from ② & ③ with $T = T_b$ P.S.D of unipolar NRZ is

$$S_X(f) = \frac{a^2 T_b}{4} \operatorname{sinc}^2(f T_b) + \frac{a^2 T_b}{4} \operatorname{sinc}^2(f T_b) \sum_{-\infty}^{\infty} \exp(-j 2\pi n f T_b) \quad — ④$$

$\downarrow$

Poisson formula i.e.

$$= \frac{1}{T_b} \sum_{-\infty}^{\infty} \delta\left[f - m/T_b\right] \quad — ⑤$$

equ ④ can be written as

$$S_X(f) = \frac{a^2 T_b}{4} \operatorname{sinc}^2(f T_b) + \frac{a^2}{4} \delta(f) \quad — ⑥$$

$S_X(f) \rightarrow$ normalized with respect to $a^2 T_b$ and $f \rightarrow$ normalized with respect to $1/T_b$.

NRZ Bipolar format.

$\rightarrow$ unipolar transformed to bipolar bymeans of special precoder having 3 levels $a, 0, -a$. assume 0's, 1's occur with equal prob then

$P(A_k = -a)$

$$= a^2\left(\tfrac{1}{4}\right) + 0^2\left(\tfrac{1}{2}\right) + a^2\left(\tfrac{1}{4}\right)$$

$$E\left(A_k^2\right) = \frac{2a^2}{4} = \frac{a^2}{2}$$

$n \neq 0 \longrightarrow$ $A_k A_{k-n} = 0, 0, 0, -a^2$ with prob $1/4$.

$$E\left(A_k A_{k-1}\right) = 3(0)\frac{1}{4} + -a^2\frac{1}{4}$$

$$= -a^2/4.$$

$n > 1 \longrightarrow E\left(A_k A_{k-n}\right) = 0$

$$\therefore R_A(n) = \begin{cases} \dfrac{a^2}{2} & n = 0 \\ -\dfrac{a^2}{4} & n = \pm 1 \\ 0 & \text{other} \end{cases}$$

$V(f) = T_b \operatorname{sinc}(f T_b) \longrightarrow$ fourier of $v(t)$

PSD of of NRZ binary format is

$$S_x(f) = a^2 T_b \operatorname{sinc}^2(f T_b) \sin^2(\pi f T_b).$$

content of the format is small around zero frequency.

Manchester format.

→ i/p binary data consists of independent, equally likely symbols.

→ auto corre fn $R_A(n) = \begin{cases} a^2 & n = 0 \\ 0 & n \neq 0 \end{cases}$

→ v(t) consists of doublet pulse of unit amp & T_b, the fourier transform

$$V(f) = j T_b \,\text{sinc}\left(\frac{f T_b}{2}\right) \sin\left(\frac{\pi f T_b}{2}\right)$$

PSD of manchester

$$S_x(f) = a^2 T_b \,\text{sinc}^2\left(\frac{1}{2} f T_b\right) \sin^2\left(\frac{\pi f T_b}{2}\right).$$

Power of the format lies inside a B/W $= 2/T_b$.