Delta Modulation :-

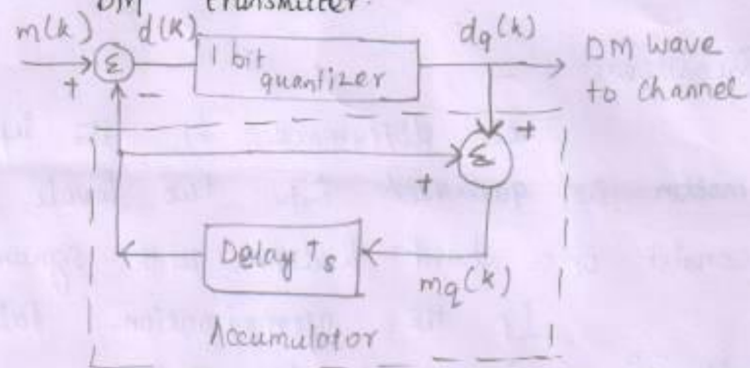
Principle of DM :-

In Delta Modulation, an incoming message signal is over sampled purposely to increase the correlation b/w adjacent samples of the signal.

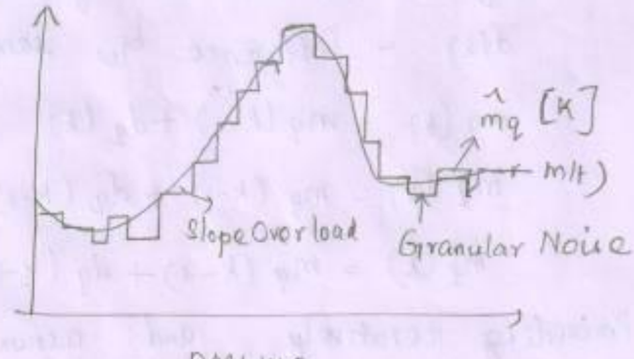
Sampling Rate = 4 x Nyquist rate

If the correlation b/w adjacent sample increases, as a result prediction error decreases.

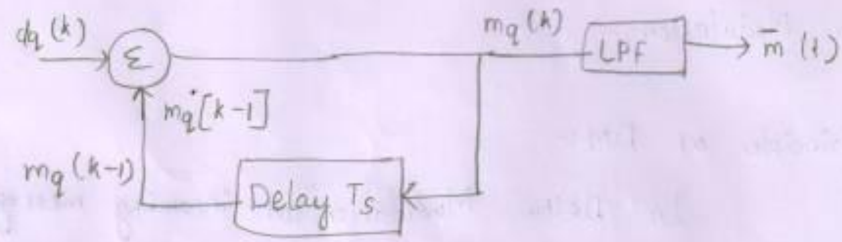
DM transmitter.



DM provides staircase approximation to the over sampled version of the message sgl is shown.



DM wave



DM Receiver.

Operation of DM transmitter:-

The DM transmitter consists of quantize one bit comparator, a 1 bit delay unit and an accu

Quantizer:-

The difference b/w the input and the appr mation is quantized into two levels namely $\pm \Delta$. It consist of a hard limiter with signum function.

If the approximation falls below the sig it is increased by Δ. If on the other hand, approximation lies above the signal it is decrea by Δ.

$m_q(k) \rightarrow$ Quantised signal.

$d(k)$ - Difference b/w samples.

$$m_q(k) = m_q(k-1) + d_q(k)$$

$$m_q(k) = m_q(k-1) + d_q(k-1)$$

$$m_q(k) = m_q(k-2) + d_q(k-1) + d_q(k)$$

Proceeding iteratively and assuming zero initia conditions $m_q(0) = 0$

$$m_q(k) = \sum_{n=0}^{k} d_q(k-n)$$

DM carries the. different signal, Which can be modeled as differentiation.

So, integration recovers the signal. As a result, the feedback loop in the transmitter is realized by an RC integrator.

Comparator:- It computes the difference b/w its two inputs.

Accumulator:- It tracks the input samples by one Step at a time.

DM receiver:- The staircase approximation $m_q(k)$ is ~~re~~ reconstructed by using accumulator.

High freq quantization noise is rejected by passing it thro' a LPF.

The filter has the bandwidth equal to the ~~ori~~ original bandwidth.

SNR Calculation of DM:-

Delta Modulation is subjected to two types of quantization error.

Slope overload distortion.

Granular Noise

If the step size Δ is too small to follow a Steep segment of input $m(t)$ then slope overload distortion occurs.

If the step size Δ is too large to follow a local slope of input $m(t)$ then granular noise occurs.

Noise power N_g is given by

$$N_g = \int_{-W}^{W} G_E(f)\,df = \frac{W\Delta^2}{3f_s}.$$

SNR for granular noise is

$$(SNR)_g = \frac{S_m}{N_g} = \frac{S_m}{W\Delta^2/3f_s}$$

$$= \frac{3f_s S_m}{W\Delta^2}$$

$S_m \rightarrow$ power in the time signal $m(t)$

The Slope loading factor $S = \dfrac{\text{Maximum DM slope}}{\text{rms signal slope}}$.

Physically, a large S ensures small slope overload.

$$SNR = \frac{3f_s}{\Delta^2 W} S_m.$$

Advantages:

1. Simpler transmitter and receiver ckts are needed.
2. For bit rate less than 40kbps, DM is preferred.

Drawbacks:-

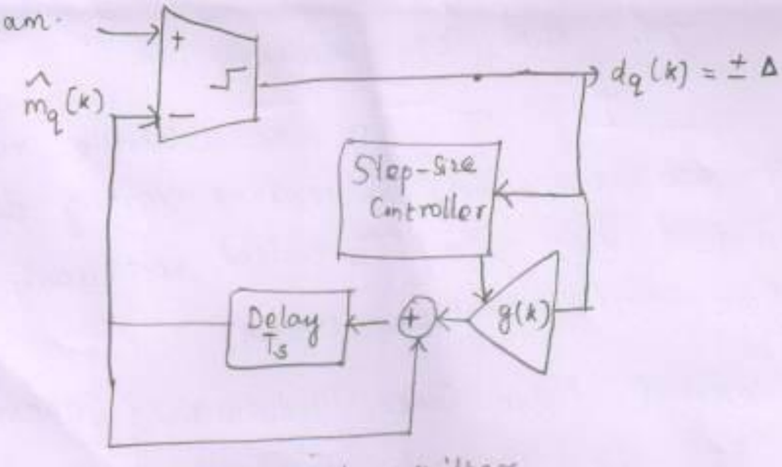
1. DM requires a large transmission BW than PCM to achieve the same SNR.

2. Speech signal requires large dynamic range, but to avoid slope overload DM has small dynamic range. So, DM is not suitable for high dynamic range speech.

Adaptive Delta Modulation (ADM)

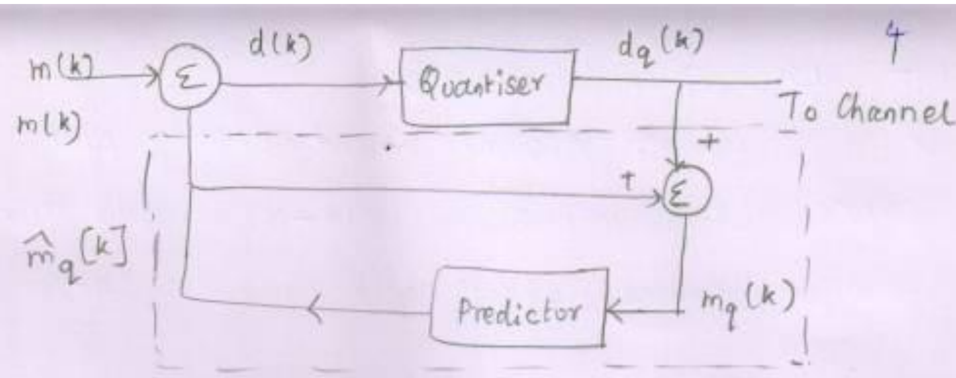
In Adaptive DM, additional hardware is kept for providing variable step size. This helps in tackling slope overload noise without increasing granular noise.

A careful examination of the DM signal $d_q(t)$ in fig shows that under slope-overloading, it is a sequence of pulses having the same polarity, whereas under granular noise, it is an alternating pulse stream.

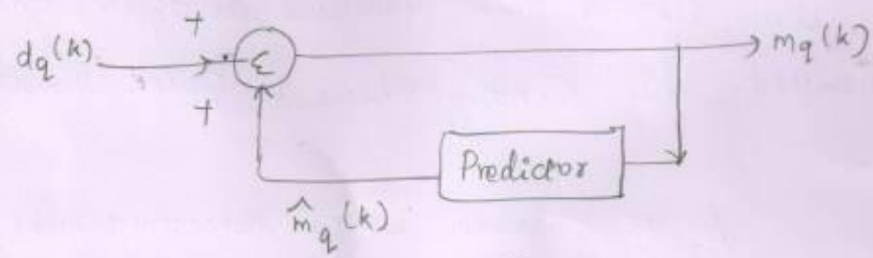


ADM transmitter

In transmitter, the ~~seto~~ step size in the feedback loop is adjusted by a variable gain $g(k)$ such that $\hat{m}_q(k) = \hat{m}_q(k-1) + g(k-1)\, d_q(k-1)$



DPCM Transmitter



DPCM Receiver

DPCM system:-

If in a communication system, instead of transmitting the input signal $m(k)$, one transmits the difference b/w the successive sample values, $d(k)$

$$d(k) = m(k) - m(k-1)$$

The new difference $d(k) = m(k) - \hat{m}(k)$

Definitely, the new $d(k)$ is smaller than the $d(k)$ of unestimated case.

If this $d(k)$ is transmitted and the receiver also estimates the signal $m(k)$ by generating $\hat{m}(k)$ locally and adds $d(k)$ received from the transmitter with it, then the signal reconstruction is achieved

The step size controller carries out the adjustment algorithm.

$$g(k) = \begin{cases} g(k-1) \times \beta & \text{if } dq(k) = dq(k-1) \\ g(k-1) \times 1/\beta & \text{if } dq(k) \neq dq(k-1) \end{cases}$$

Where β is a constant taken to be in the range $1 < \beta < 2$.

Initially the step size is kept small to minimise the granular noise and gradually increased until the slope overload noise begins to dominate.

One should note that quantisation in ADM is non-uniform and it is achieved by varying the stepsize. The SNR of ADM is typically 8-14dB better than that of ordinary DM.

If the step size controller varies the step size continuously, instead of varying directly, the resultant ADM variety is called continuously variable slope delta modulation (CVSDM).

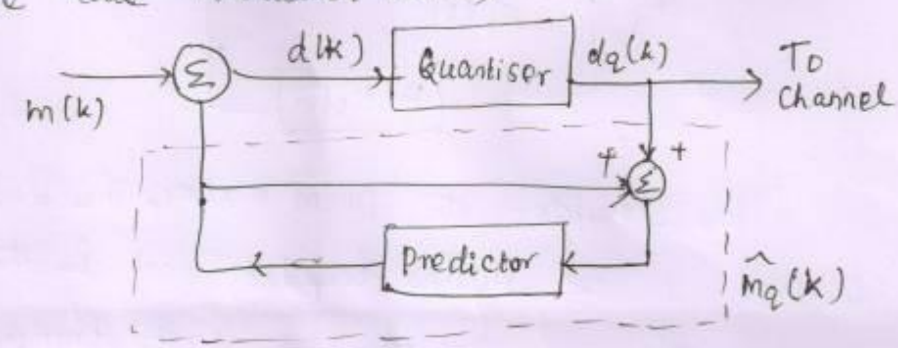
Differential Pulse Code Modulation - (DPCM)

In PCM, each sample of the W/F is encoded independently of all the other samples. If the consecutive samples of a source O/P are independent of each other.

So, the receiver instead of having past samples of $m(k-1); m(k-2), \ldots m(k-N)$ has only their quantised versions $m_q(k-1), m_q(k-2) \ldots m_q(k-N)$.

$$d(k) = m(k) - \hat{m_q}(k)$$

This $d(k)$ is then quantised as $d_q(k)$ and transmitted. The encoding scheme is called Differential Pulse Code Modulation (DPCM)



In the fig the predictor is implemented with a feedback loop around the quantiser. At the transmitter the ~~pr~~ predictor input is $m_q(k)$. So, predictor assume that the predictor o/p is $m_q(k)$. Hence the quantiser input is

$$d(k) = m(k) - \hat{m}(k).$$

After the quantiser block, the quantisation error $q(k)$ gets added, so that the o/p becomes

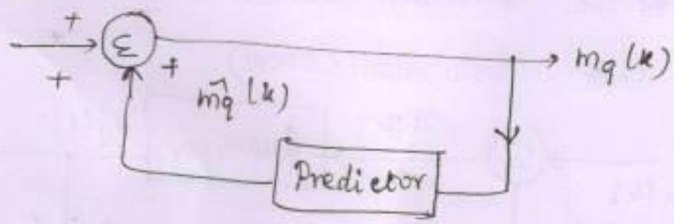
$$d_q(k) = d(k) + q(k)$$

Consider predictor o/p $\hat{m_q}(k)$ is fed back to the predictor input side.

So, input to predictor is

$$\hat{m_q}(k) + d_q(k) = m(k) - d(k) + d_q(k)$$
$$= m(k) + \{d_q(k) - d(k)\}$$
$$= m(k) + q(k)$$
$$= m_q(k)$$

DPCM Receiver



Now consider the DPCM receiver. This block is identical to the dashed-bounded block in the transmitter. The input to this block is the transmitted $d_q(k)$. So, the predictor input is $m_q(k)$ and output is $\hat{m}_q(k)$. Hence, the receiver o/p is $m_q(k) = m(k) + q(k)$.

At the receiver o/p, we get the desired signal $m(k)$ plus the quantisation noise $q(k)$ associated with the quantisation of the difference signal $d(k)$ which is much smaller than $m(k)$.

SNR improvement in DPCM :-

This improvement factor come due to the employment of the feedback loop and the predictor block in the DPCM system.

This G_p is given by

$$G_p = \left(\frac{m_p}{d_p}\right)^2.$$

So, the processing gain can also be written as

$$G_p = \frac{\overline{m}^2}{d^2}$$

$$SNR_{DPCM} = G_p \; SNR_{PCM}$$

$$= G_p(dB) + (4.77 + 6l)\,dB.$$

Predictor gain depends on statistical properties of the input signal. The predictor gain of speech is $G_p = 5.6\,dB$. Usually for speech is $G_p = 5.6\,dB$, whereas for TV video $G_p = 12\,dB$.

Adaptive Differential Pulse Code Modulation (ADPCM)

The encoding methods like PCM, APCM & DPCM demand channel bandwidths of the order of 40-64 kHz for its transmission. In certain applications, involving speech commonication, such as, secure transmission over low bandwidth radio channel, there is a pressing need for reducing transmission BW further.

The design philosophy of all these coders is

1. To remove redundancies from speech signal as far as possible
2. To encode the speech signal devoid of

~~red~~ redundancy in a perceptually efficient manner by constantly adapting to the speech statistics of the input signal.

The adaptive predictors are classified according to whether the estimate is derived from the previous samples of the input signal itself. [Adaptive prediction with forward estimation (APF) or from the previous predictor O/ps as well as corresponding prediction errors (APB). Adaptive prediction with backward Estimation.

An ADPCM system, apart from predically refreshing the the predictor co-efficients also varies the step size of the quantiser $\Delta(kT_s)$ dynamically according to the signal's standard deviation $\hat{\sigma}_m(kT_s)$

$$\Delta[kT_s] \propto \hat{\sigma}_m[kT_s]$$

Adaptive quantisers also of two broad types.

- Adaptive Quantisation with forward Estimation (AQF): Unquantised samples of the input signal
- Adaptive Quantisation with backward Estimation. Samples of the quantiser o/p are used for estimation of $\hat{\sigma}_m[kT_s]$.

AQF requires.

- memory buffer to store unquantised samples of input
- explicit transmission of step-size samples as side information.
- more processing delay [$\simeq$ 16ms for speech which is unacceptable in certain applications)

All the three drawbacks can be avoided in AQB. This makes AQB a popular choice in ADPCM systems.

APF also suffers from the same disadvan[tages] buffering, side information & delay.

Linear Predictor Coding (LPC)

The most widely used Model-~~M~~ based encoding method is called Linear predictor coding(L[PC]). In this scheme, the basic assumption is that it is always possible to simulate the sampled speech sequence $x(n)$, $n = 0, 1, \ldots N-1$. by applying appropriat[e] excitation sequence $v(n)$ to a discrete time linear filter with an all-pole transfer function $H(z)$. representing the human vocal tract.

It has been found that all possible spe[ech] sequences can be generated by applying either an impulse train or a noise sequence as excita[tion]

Human speech can be categorised as either voiced or unvoiced.

On a Short time basis, Voice speech is found to be periodic with a fundamental freq. f_0 Where f_0 is the pitch frequency of the speaker.

Thus Voiced Speech can be simulated by exciting the filter $H(z)$ with a periodic impulse train having its frequency equaling the pitch frequency of the speech. So, for voiced speech, the excitation sequence can be generated by an Impulse signal generator supplied with the pitch frequency of the speech signal to be encoded.

Unvoiced speech sounds are generated by exciting the same filter with White Noise.

The filter transfer function $H(z)$ is described by

$$H(z) = \frac{G}{1 - \sum_{k=1}^{p} a_k z^{-k}}.$$

So, to specify the filter, one needs to define the filter co-efficients $\{a_k\}$ and the gain G.

$$x(n) = \sum_{k=1}^{p} a_k x[n-k] + G\, V(n),$$

$n = 0, 1, 2 \ldots\ldots$

When the input sequence is White noise,

We may form prediction of $x(n)$ from the difference equation as

$$\hat{x}[n] = \sum_{k=1}^{p} a_k x(n-k) \quad , n \geq 0$$

The error in the prediction is given by

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

$$= x(n) - \sum_{k=1}^{p} a_k x[n-k]$$

However to completely specify the transfer function $H(z)$, we must determine the filter gain G.

$$G V(n) = x(n) - \sum_{k=1}^{p} a_k x[n-k] \quad n = 0, 1, 2 \ldots$$

$$= e[n]$$

Squaring and Averaging

$$E[(G V[n])^2 = \overline{e^2[n]}$$

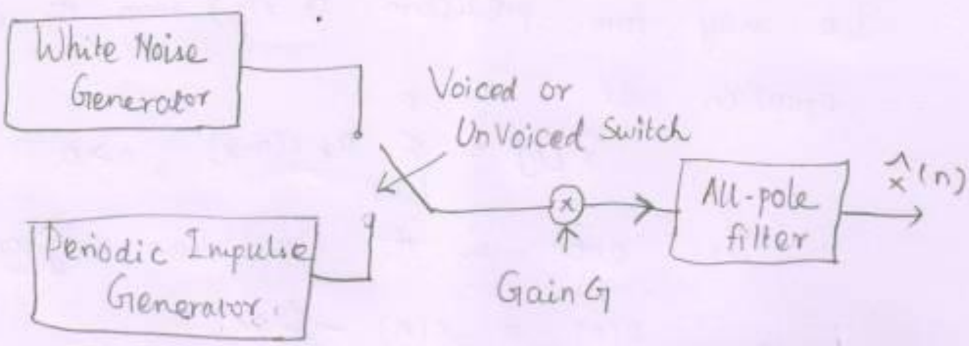
$$E[G V[n]]^2 = G^2 E[V^2[n]]$$

$$= G^2.$$

$$\overline{e^2[n]} = R_{00} - (a_1 R_{01} + a_2 R_{02} + \cdots - a_p R_{0p})$$

$$G^2 = R_{00} - \sum_{k=1}^{p} a_k R_{0k}$$

$$R_{0k} = \frac{1}{N} \sum_{l=0}^{N-k} x_i x_{i+k} \quad k = 0, 1, 2 \ldots p.$$

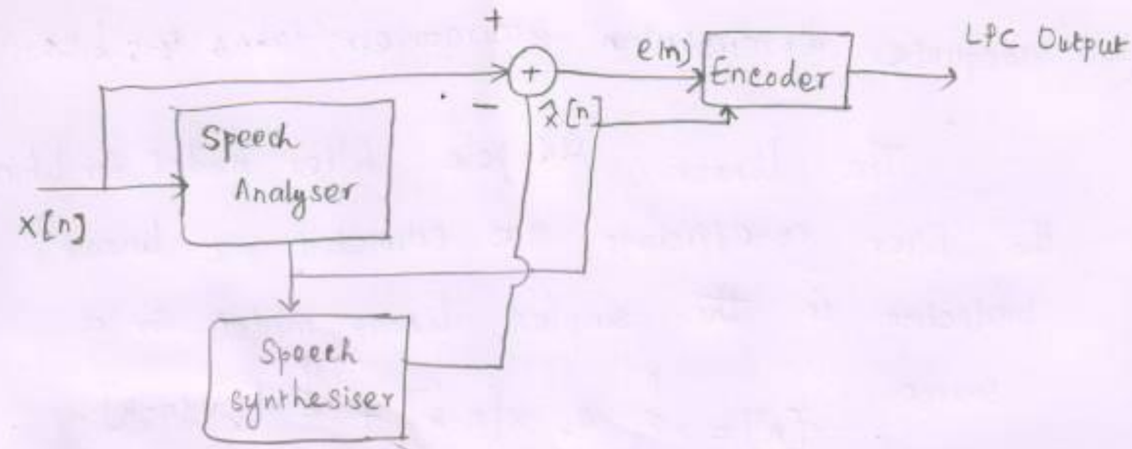


· Speech synthesizer in LPC

LPC has been successfully used in the encoding of speech. From the input speech signal observed over a short period, say 20ms. first the transmitter determines whether the speech so observed is voiced or unvoiced. ~~then~~

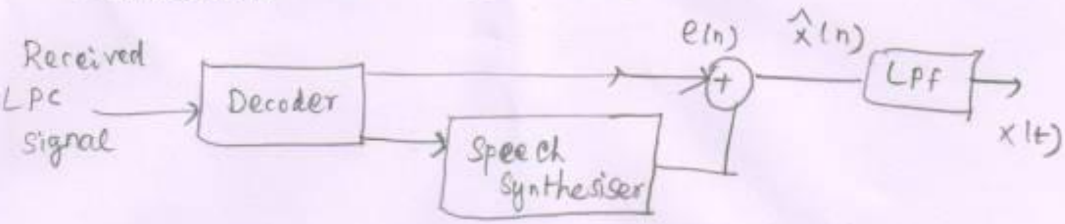
Next the predictor co-efficients and gain G are estimated from the speech sequence $x[n]$. Once all these parameters of the speech. i.e type, pitch, filter co-efficient and gain are determined.

The voiced/unvoiced type information bit activates the voiced/unvoiced switch, thereby connecting either of the impulse or noise generator. to the prediction filter The gain and filter co-efficient information are utilised to implement the prediction filter $H(z)$.



LPC transmitter

The speech Analyser is a sgl processing block whose o/p is the set of all the parameter values required to characterise the speech under consideration the parameters are fed to an encoder for onward transmission to the receiver. The error b/w the ~~act~~ actual speech input and locally synthesised speech is also fed to the encoder for onward transmission to the receiver.



Typically the voiced and unvoiced information requires 1 bit, the pitch period is by 6 bits. and gain parameter by 5 bits.

Filter Co-efficients requires 8-10 bits for adequate representation accuracy. e(n) takes 4-5 bits.

The linear all-pole filter model, for which the filter co-efficients are estimated via linear prediction is the simplest linear model for a source.

$$x[n] = \sum_{k=1}^{p} a_k \, x[n-k] + \sum_{k=0}^{q} b_k v[n-k].$$

$x[n]$ - Output sequence

a_k, b_k - Coefficients by MMSE Criteria.