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Performance Parameters Trade-Off in Digital Communication Applications

[1]Pankaj Preet Singh Dhaliwal, [2]Mahendra Kumar
[1]Research Scholar, ECE Department, Guru Kashi University, Talwandi Sabo, Bathinda
[2]Dy dean Research, Guru Kashi University, Talwandi Sabo, Bathinda

Abstract: Digital communication is preferred over analog communication due to better noise immunity. For Digital Communication between two geographical apart stations the quality of received signals at receiver end is an important factor. To achieve desired quality of received signal at receiver end we can have a trade-off between major parameters i.e. bandwidth, bit width and hardware complexity. If we go for less complex hardware technique such as Pulse Code Modulation (PCM) and utilize more bandwidth, we can accommodate higher bit rate which will reduce quantization distortion being inverse function of bit rate. Alternatively we can use complex and costly hardware i.e. higher variates of Pulse Code Modulation such as Differential Pulse Code Modulation (DPCM) and Adaptive Differential Code Modulation (ADPCM) which can ensure less quantization distortion at low bit rate or less bandwidth. This paper carries out analysis for optimum use of bandwidth, bit width and hardware complexity for achieving the desired signal quality at receiver end of Digital Communication System.

Keywords: Bandwidth, Digital Communication, Hardware complexity, Trade-off.

1. INTRODUCTION

For transmission of speech signals, digital communication is preferred over analog communication due to certain advantages. Brief introduction to digital communication is given in succeeding paragraphs.

In digital communication, the information in the form of analog electrical signals is converted to discrete or digital signals for communication.

1.1 Digital Communication

The Digital communication system has two hardware subsystems i.e. transmitter and receiver .

a. Transmitter: The block diagram of digital communication system transmitter is given in Fig 1. Analog signal is converted to digital signal by precoder. Very high frequency analog carrier is modulated by digital signal and amplified by power amplifier and fed to transmission media. Buffers are used in the circuit to avoid loading.

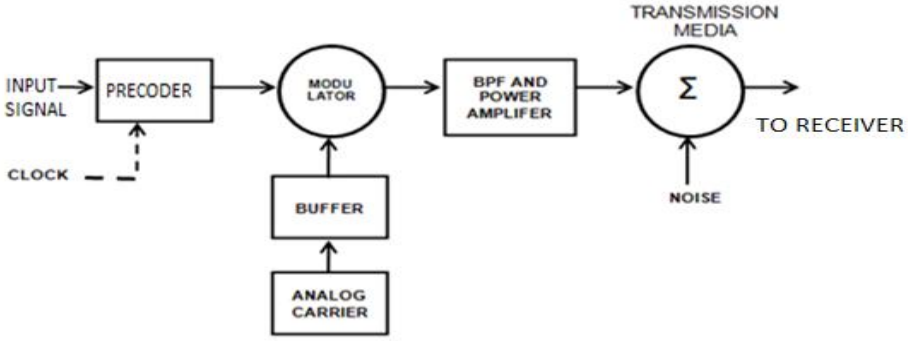

Figure 1 – Digital Communication Transmitter

b. Receiver: The block diagram of digital communication system receiver is given in Fig 2.The modulated signal received from transmission media is amplified by amplifier and fed to demodulator and decoder. Carrier and clock recovered from received signal is also given as input to demodulator and decoder. The output of demodulator and decoder is required audio signal.

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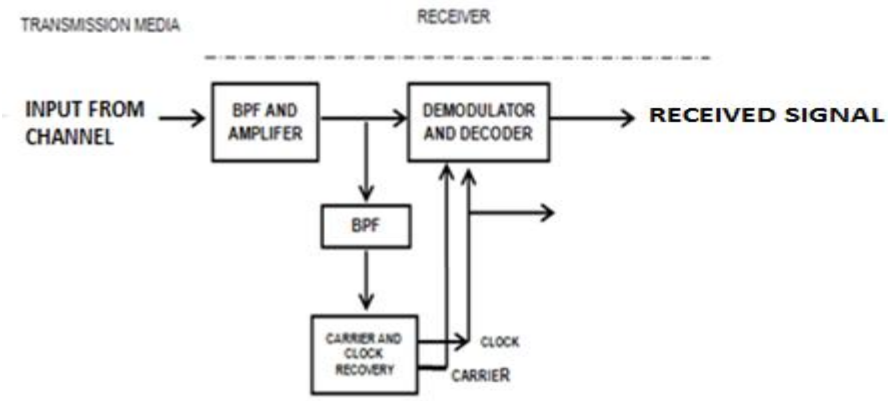


Figure 2 – Digital Communication Receiver

It can be seen from above diagrams and explanation that circuitry of digital communication system is complex as compared to analog communication system due to additional circuitry required for speech coding and decoding. Even then digital communication system is preferred due to certain inherent advantages.

1.2 Advantages of Digital Communication

For transmission of signals, digital communication is preferred over analog communication due to following advantages:

(a) Better channel noise immunity
(b) Suitable for time division multiplexing.
(c) Suitable for digital signal processing
(d) Can be transmitted on same channel along with data after multiplexing
(e) Can be stored/delayed for short periods, hence suitable for more efficient communication utilizing statistical multiplexing and asynchronization mode.
(f) Secrecy can be introduced by scrambling and cryptography.

1.3 Disadvantages Of Digital Communication

Digital communication has following disadvantages as compared to analog communication

(a) More band width requirement.
(b) Additional circuitry for digitizing signals.
(c) Addition of quantization distortion to transmitted signals.

1.4 Optimization of Digital Communication System

Digital communication system can be optimized by encashing on its advantages and at the same time reducing the effect of its disadvantages. It can be achieved by:

(a) Increasing digital signal immunity to transmission-media(channel) noise hence improving speech quality at receiver end.
(b) Reducing band width requirement.
(c) Using simple circuit for digitizing signals to reduce system cost.
(d) Reducing quantization distortion in transmitted signals.

2. DIGITIZING TECHNIQUES

For further discussion on optimization of digital communication systems there is requirement of studying various digitizing techniques or different codecs. Popular digitizing techniques are:

(a) Delta Modulation(DM)
(b) Adaptive Delta Modulation(ADM)
(c) Pulse Code Modulation(PCM)
(d) Differential Pulse Code Modulation(DPCM)
(e) Adaptive Differential Pulse Code Modulation (ADPCM)

2.1 Delta Modulation(DM)

The block diagram of delta modulation system is given in Fig 3. The comparator compares the present sample of input signal with feedback from accumulator which is one sampling period delayed version of previous sample.

If present sample is larger ,one bit quantizer output is 1 and otherwise 0 which is further fed to encoder and accumulator. Encoder output is transmitted to distance station as sampled output.
On receiver side sampled channel output is decoded by decoder and given as input to accumulator. Accumulator output will be incremented if input to it is 1 and decreased if input to it is 0. Accumulator output is given to low pass filter for reconstructing analog signal.
Delta modulation has constant increment/decrement during one sampling period hence suffers due to slope overloading at higher signal slopes.

Salient Points about Delta Modulation(DM): Very less cost, very simple circuit, high slope overloading leading to high quantization distortion. Speech quality at receiver end is poor. Not suitable for speech communication.

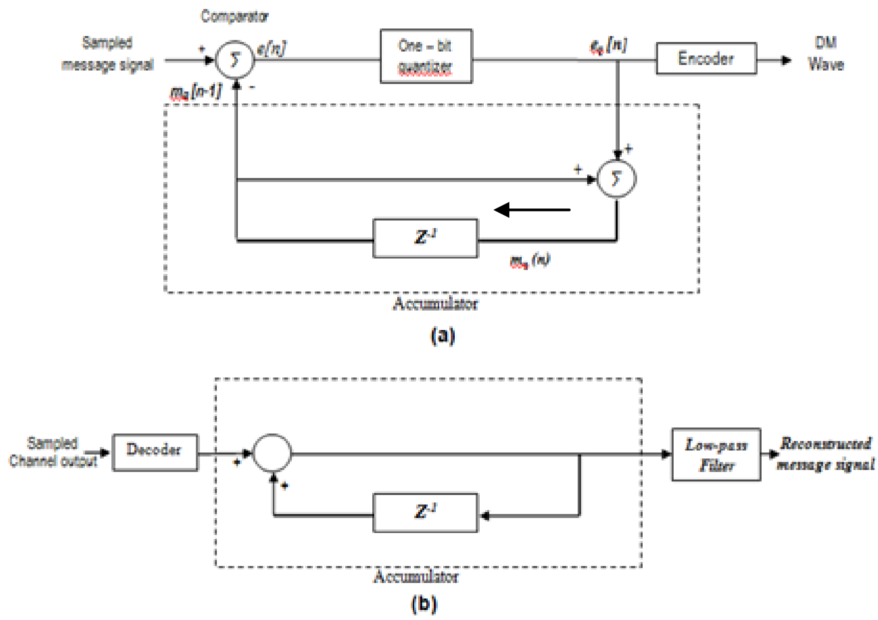
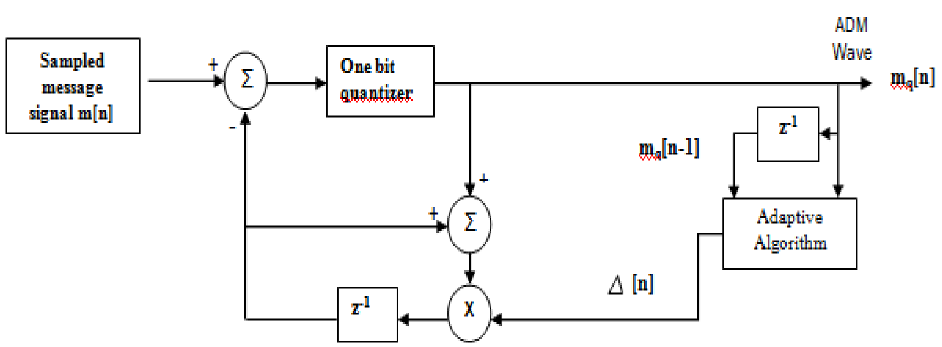


Fig. 3 DM System (a) Transmitter. (b) Receiver

2.2 Adaptive Delta Modulation(ADM)

The block diagram of adaptive delta modulation system is given at Fig 4. The functional working is similar to delta modulation except that adaptive algorithm block and multiplier circuit are added for adapting the step size as per slope of the signal and for avoiding the slope over loading.



(a)

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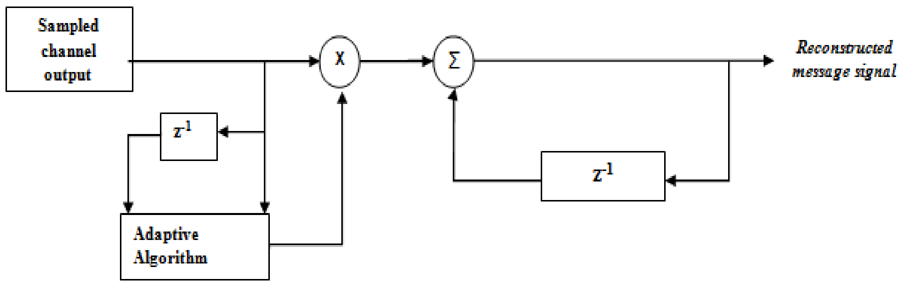

(b)

Figure 4 Adaptive Delta Modulation System (a) Transmitter (b) Receiver

Salient Points about Adaptive Delta Modulation(ADM): Simple circuit, less cost, less slope over loading and quantization distortion as compared to Delta Modulation(DM), satisfactory speech quality at receiver end. May be used for speech communication where satisfactory speech quality is acceptable.

2.3 Pulse Code Modulation(PCM)

The block diagram of pulse code modulation system is given at Fig 5. Input signal is passed through a band pass filter and given to sample and hold circuit which samples the signal at sampling frequency and holds the present sample for analog to digital conversion by next block till next sample arrives. Parallel output of analog to digital converter is changed to serial output by next block and transmitted on channel to distance receiver.

In between transmitter and receiver there can be regenerative repeaters for regenerating of pulses. At receiver end serial input is converted to parallel and given to digital to analog converter. Output of digital to analog converter is given to hold circuit. Output of hold circuit is given to low pass filter for generating of analog signal.

Salient Points about Pulse Code Modulation(PCM): Average circuit complexity, moderate cost, less quantization distortion. Good speech quality at receiver end. Very suitable for digital communication of speech.

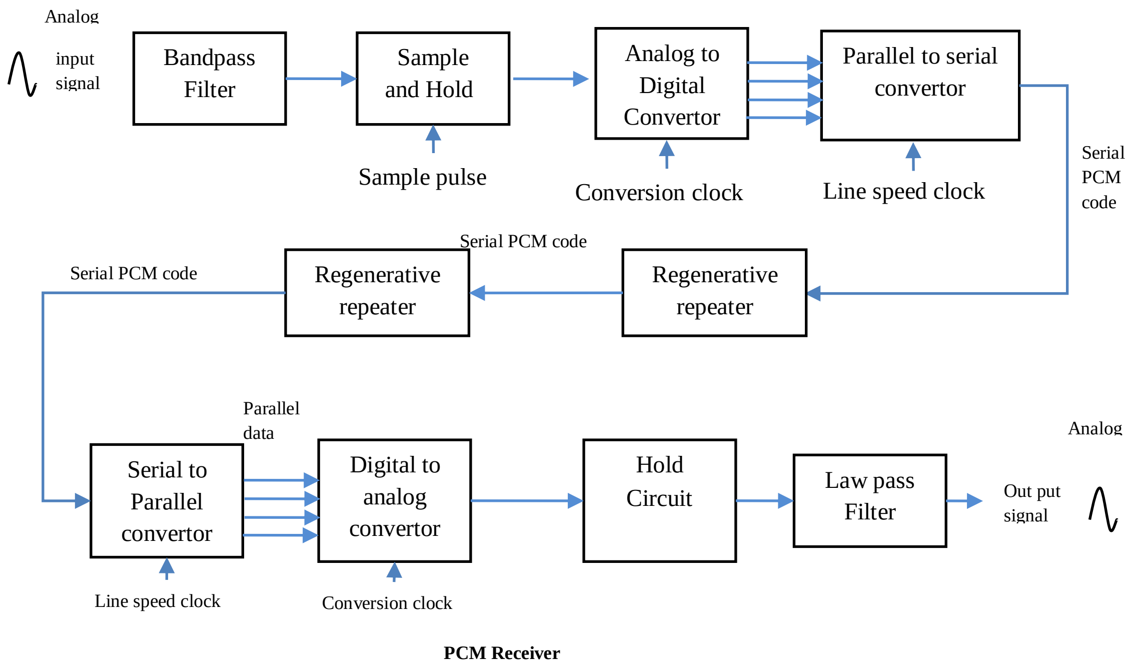

Figure 5 Pulse Code Modulation System

Performance Comparison Between ADM and PCM: Quantization distortion in a system is inversely proportional to output bit rate of that system. Quantization distortion in a ADM system is less than a PCM system up to an output bit rate of 32Kb/S. At 32kb/S both systems give an approximate quantization distortion of -22dB which is not acceptable for good speech quality at receiver end. For further reducing quantization distortion for improving the speech quality at receiver end we have to increase the out put bit rate beyond 32Kb/S. For output bit rate range of greater than 32Kb/S PCM gives better performance and less quantization distortion as compared to ADM. Ref Fig 6.

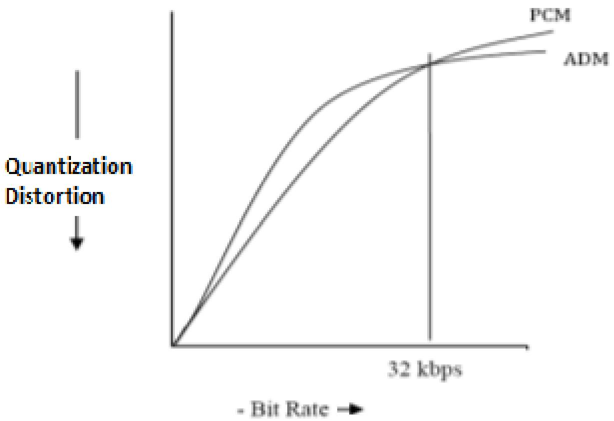

Figure 6 Performance Comparison Between ADM and PCM

2.4 Differential Pulse Code Modulation(DPCM)

Block diagram of differential pulse code modulation system is given at Fig7. Sampled input signal is fed to the comparator which gets another predicted input from prediction filter. The output of comparator is prediction error which is input to the quantizer. The output of quantizer is given to encoder for encoding and further transmission to distance station. Output of quantizer is also given to prediction filter through summing circuit for predicting the amplitude of next input sample.

On receiver side the signal received from distance station is given as input to decoder. The decoder output is given to the summing circuit along with an input from prediction filter. The output of summing circuit is the received signal which is also given as input to prediction filter for prediction of amplitude of next sample.

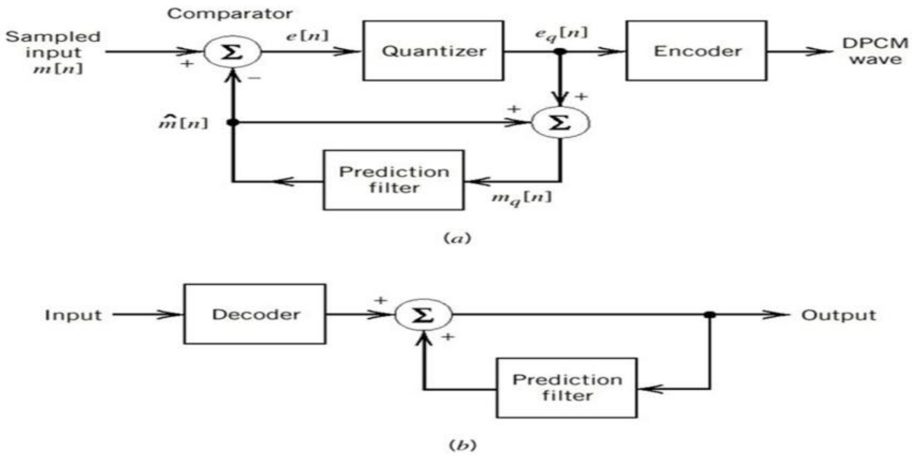

Figure 7- DPCM System (a) Transmitter (b) Receiver

Salient Points about Differential Pulse Code Modulation(DPCM): Complex circuitry, high cost, very less quantization distortion, very good signal quality at receiver end. Not preferred for speech communication due to complex circuitry and high cost of customer equipment. Suitable for video and image transmission.

2.5 Adaptive Differential Pulse Code Modulation

The block diagram of adaptive differential pulse code modulation is given at Fig 8. The working of transmitter and receiver is similar to DPCM except that it has additional circuit known as logic for adaptive prediction. This additional circuit further fine tunes the prediction filter by changing its coefficients as per the varying slope of the input signal. This reduces the prediction error.

Salient Points about Adaptive Differential Pulse Code Modulation(ADPCM): very complex circuit, very high cost, negligible quantization distortion, excellent signal quality at receiver end. Not preferred for speech communication due to very complex circuit and very high cost of customer equipment. Most suitable for applications where excellent received signal quality is required and available band width is less.

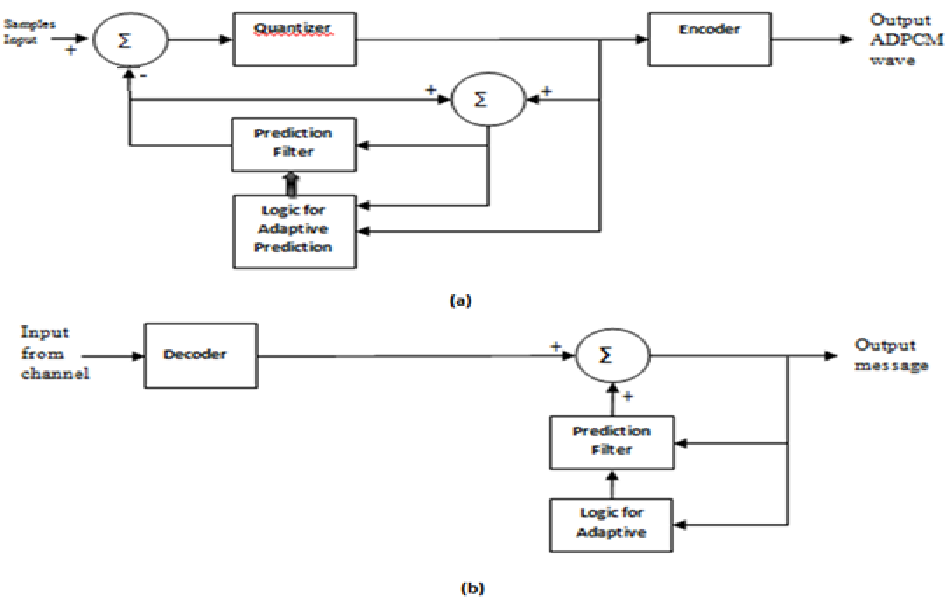

Figure 8 – Adaptive Differential Pulse Code Modulation (a) Transmitter (b) Receiver

3. DISTORTION DUE TO CHANNEL NOISE IN DIGITAL BIT STREAM RECEIVED AT INPUT TO RECEIVER SYSTEM

It is applicable to all digital communication techniques.

Violation in digital bit stream due to channel noise are predicted by probability of error (Pe)

The probability of an error (P_e) taking place with respect to each received bit of digital stream is given by:

$$P_e = \frac{1}{2}\text{erfc}\left[\frac{1}{2}\sqrt{\left(E_{max}/N_{\circ}\right)}\right]$$

Where

erfc (u) is Complementary Error Function u and is given by:

erfc (u) = $2/\sqrt{\pi}\int_u^{\infty} exp(-Z^2)\, dz$,

where,

$$u = \frac{1}{2}\left[\sqrt{\left(E_{max}/N_{\circ}\right)}\right]$$

Emax = Peak signal energy at receiver input.

No = Noise spectral density at receiver input.

And Z= Probabilty variate

Now

Emax = Pmax * Tb

Where,

Pmax = Peak signal power

Tb = Bit Width

Therefore,

$$Pe = 1/\sqrt{\pi}\int_{\frac{1}{2}\sqrt{P_{max}.Tb/N_{\circ}}}^{\infty} \exp(-Z^2)\, dz$$

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Under given environment i.e. the given transmission signal power and expected channel noise, the expression ½ $\sqrt{(P_{max}/N_o)}$ at receiver input will be a constant, say 'C'.

Then

$P_e = 1/\sqrt{\pi} \int_{c*\sqrt{Tb}}^{\infty} exp\,(-Z^2)\, dz$

Hence,

P_e is a decreasing function of $\sqrt{Tb}$

It can be seen that if bit duration is increased then Pe decreases hence distortion due to channel noise is reduced, but bit rate which is inversely proportioned to bit duration is reduced. This reduces bits/sample , hence quantization distortion increases. Hence an effort for decreasing quantization distortion by increasing bit rate results in increase in distortion due to channel noise and vice versa. Therefore, for optimizing the system a trade off between quantization distortion and distortion due to channel noise is required. The bit rate selected should be such that the total distortion at the receiver input due to quantization at transmitter and due to channel noise should be minimized as shown in Fig 9 .

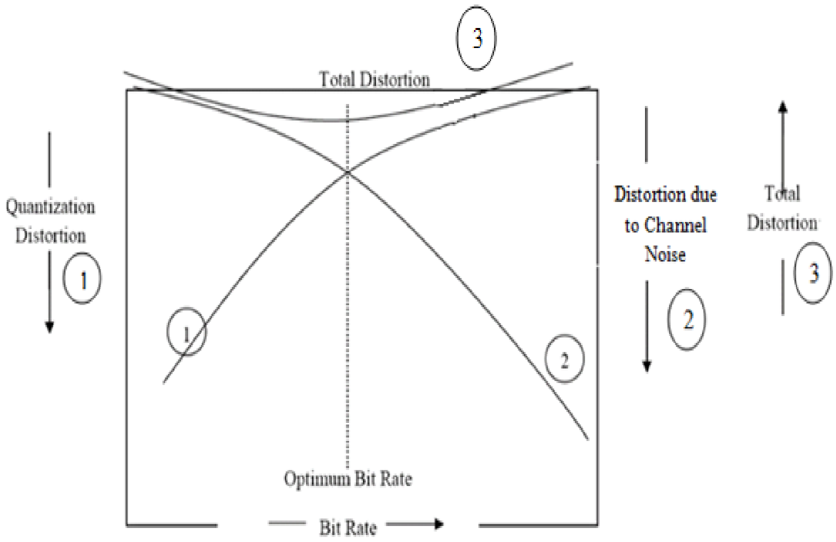

Figure 9 - Total Distortion at the Receiver

4. BAND EXPANSION FACTOR

When an analog signal is converted to digital the bit rate increases many fold as compared to frequency of analog signal. The ratio of resultant digital bit stream to corresponding analog signal frequency is known as "band expansion factor". This happens due to following reason

Conversion of analog signal to digital involves sampling. Number of minimum samples required is given by sampling theorem:

Number of min samples per second of analog signal required =

Twice the Highest Frequency Component Available in the analog signal (Nyquist criteria) = 8K (For analog speech signal band limited to 4 KHz)

Therefore number of min samples/sec itself is numerically equal to bandwidth (BW) of Amplitude Modulated Double Side Band (AMDSB) signal i.e. = 8 KHz.

For PCM: For acceptable dynamic range of input speech signal and reasonably low level of quantization distortion at receiver, at least 256 level quantization of speech signal samples is necessary. Therefore for coding of each speech sample 8 digital bits are required in popular Pulse Code Modulation(PCM) technique

Digital base band bit rate = 8x8 = 64 Kbits/sec

Minimum Band Expansion factor = Digital Signal Bit Rate/BW of AMDSB

(For 64Kbits PCM)

= 64KHz/8KHz = 8

For DPCM and ADPCM: In DPCM and ADPCM only prediction error signals are required to be transmitted, the dynamic range of these error signals is very less as compared to input analog signals. Therefore for coding of prediction error signals samples the number of bits required per sample will be less than 8 for same level of quantization distortion. This results in reduction of bandwidth as compared to PCM due to reduction in band expansion factor.

5. TRADE OFF

As per details in preceding paragraphs for reasonable signal quality at receiver end of a digital communication system, we have to reduce quantization distortion and violations due to channel noise. At the same time we have to take into consideration the system cost. For this the following trade offs are possible.

(a) **Bit Rate or Band Width (BW):** Higher bit rates consume more BW but they reduce quantization distortion. ADM system gives less quantization distortion upto 32Kbits bit rate. At 32Kbits quantization distortion is -22 dB which gives only satisfactory speech quality at receiver end. For further improving speech quality at receiver end by reducing quantization distortion PCM at higher than 32K bits/sec is superior to ADM. Less quantization distortion as compared to PCM can be achieved at the same bit rate by using DPCM or ADPCM.

(b) **Signal Energy/Bit:** Higher bit rate reduces bit width or signal energy/bit. This leads to more bit errors or violations on transmission channels and hence reduces signal quality at receiver end as distortion due to channel noise increases . So we have to make a trade off between acceptable quantization distortion and acceptable distortion due to channel noise while deciding bit rate as shown in . Fig 9 .

(c) **Circuit complexity:** Complex circuit of DPCM and ADPCM reduces quantization distortion even at less bit rates hence less distortion due to channel noise. But cost of user equipment in case of DPCM and ADPCM is very high as compared to PCM.

6. SUMMARY

Taking into consideration the salient points of various digitizing techniques and possible trade offs, the following inferences can be drawn:

(a) For digital communication of speech / digital telephony , customer population is very high , hence the customer equipment i.e. encoder/decoder with moderate cost and a system with reasonably good speech quality at receiver end will be required. Therefore , Pulse Code Modulation system is most suitable for digital communication of speech / digital telephony.

(b) For higher end applications such as image or video transmission very good/excellent received signal quality is required and customer population is not very high. Hence Differential Pulse Code Modulation and Adaptive Differential Pulse Code Modulation systems are suitable.

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