1.1 Advantages of Digital Communication System

Presently most of the communication is digital. For example cellular phone) communication, satellite communication, radar and sonar signals, Fax data transmission over internet etc all use digital communication. Paractically 20 years, analog communication will be totally replaced by digital communication

Why digital communication is so popular ?

There are few reasons due to which people are prefering digital communication over analog communication.

1. Due to advancements in VLSI technology, it is possible to manufacture high speed embedded circuits. Such circuits are used in digital communications.
2. High speed computers and powerful software design tools are available. It make the development of digital communication systems feasible.
3. Internet is spread almost in every city and towns. The compatibility of digital communication systems with internet has opened new area of applications.

Advantages and Disadvantages of Digital Communication

Advantages :

1. Because of the advances in digital IC technologies and high speed computers digital communication systems are simpler and cheaper compared to analog systems.
2. Using data encryption, only permitted receivers can be allowed to detect the transmitted data. This is very useful in military applications.
3. Wide dynamic range is possible since the data is converted to the digital form.
4. Using multiplexing, the speech, video and other data can be merged and transmitted over common channel.
5. Since the transmission is digital and channel encoding is used, the noise does not accumulate from repeater to repeater in long distance communication.
6. Since the transmitted signal is digital, a large amount of noise interference can be tolerated.
7. Since channel coding is used, the errors can be detected and corrected in receivers.
8. Digital communication is adaptive to other advanced branches of data processing such as digital signal processing, image processing and data compression etc.

Disadvantages :

Eventhough digital communication offer many advantages as given above, it has some drawbacks also. But the advantages of digital communication outweigh disadvantages. They are as follows -

1. Because of analog to digital conversion, the data rate becomes high. Hence more transmission bandwidth is required for digital communication.
2. Digital communication needs synchronization in case of synchronous modulation.

1.2 Elements of Digital Communication System

Fig. 1.2.1 shows the basic operations in digital communication system. The source and the destination are the two physically separate points. When the signal travels in the communication channel, noise interferes with it. Because of this interference, the smeared or disturbed version of the input signal is received at the receiver. Therefore the signal received may not be correct. That is errors are introduced in the received signal. Thus the effects of noise due to the communication channel limit the rate at which signal can be transmitted. The probability of error in the received signal and transmission rate are normally used as performance measures of the digital communication system.

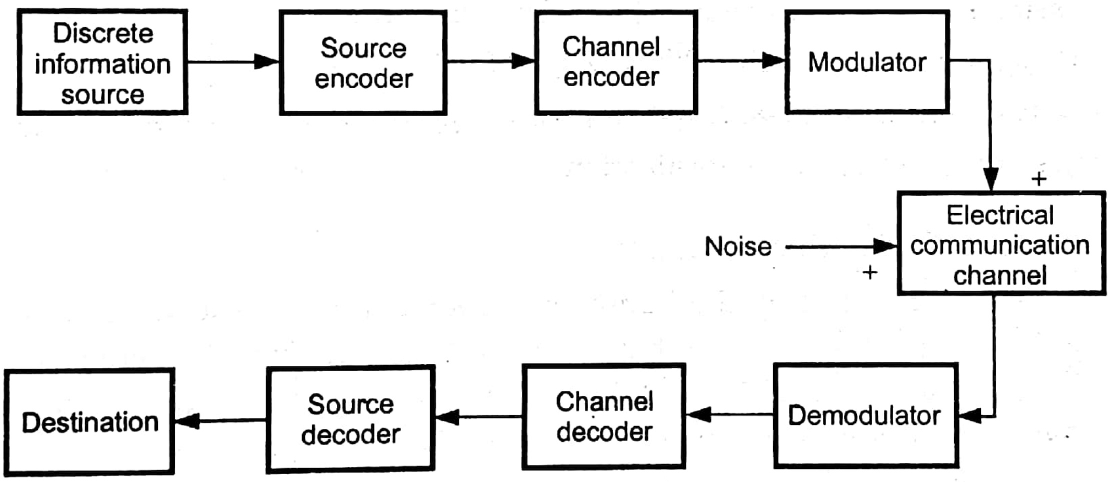


Fig. 1.2.1 Basic digital communication system

1.2.1 Information Source

The information source generates the message signal to be transmitted. In case of analog communication, the information source is analog. In case of digital communication, the information source produces a message signal which is not continuously varying with time. Rather the message signal is intermittent with respect to time. The examples of discrete information sources are data from computers,

1.4 Pulse Amplitude Modulation (PAM)

- **Definition** : The amplitude of the pulse change according to amplitude of modulation signal at the sampling instant.
- **Types of PAM** : Depending upon the shape of the pulse of PAM, there are three types of PAM :

(i) Ideally or instantaneously sampled PAM.

(ii) Naturally sampled PAM.

(iii) Flat top sampled PAM.

1.4.1 Ideal Sampling or Instantaneous Sampling or Impulse Sampling

Basic Principle

Ideal sampling is same as instantaneous sampling. Fig. 1.4.1 (a) shows the switching sampler. If closing time 't' of the switch approaches zero the output $x_\delta(t)$ gives only instantaneous value. The waveforms are shown in Fig. 1.4.1 (b). Since the width of the pulse approaches zero, the instantaneous sampling gives train of impulses in $x_\delta(t)$. The area of each impulse in the sampled version is equal to instantaneous value of input signal $x(t)$.

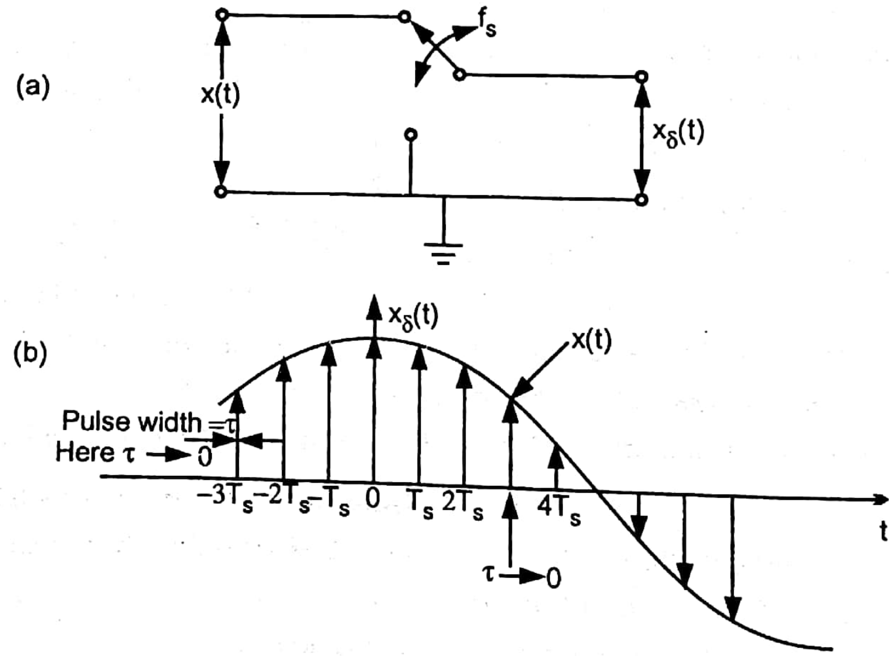


Fig. 1.4.1 **(a) Functional diagram of a switching sampler**
(b) Waveforms of $x(t)$ and its sampled version $x_\delta(t)$ giving instantaneous sampling

Explanation

- We know that the train of impulses can be represented mathematically as,

$$s_\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \quad \text{... (1.4.1)}$$

- This is called sampling function. The sampled signal $x_\delta(t)$ is given by multiplication of $x(t)$ and $s_\delta(t)$.

Therefore,
$$x_\delta(t) = x(t)\, s_\delta(t)$$
$$= x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$
$$= \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t - nT_s) \quad \text{... (1.4.2)}$$

The above equation we have directly written previously as equation 1.3.1.

The Fourier transform of the ideally sampled signal given by above equation can be written as,

$$\boxed{\textbf{Spectrum of Ideally Sampled Signal} : X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)} \quad \text{... (1.4.3)}$$

Comments

i) $X(f)$ is periodic in f_s and weighed by f_s.

ii) Instantaneous sampling is possible only in theory because it is not possible to have a pulse whose width approaches zero

1.4.2 Natural Sampling or Chopper Sampling

- **Basic Principle**

In natural sampling the pulse has a finite width τ. Natural sampling is some times called chopper sampling because the waveform of the sampled signal appears to be chopped off from the original signal waveform.

- **Explanation**

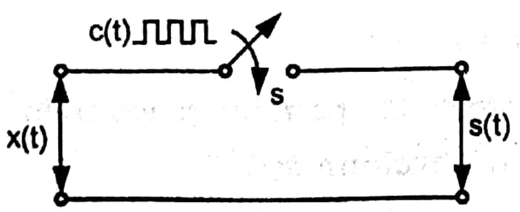


Fig. 1.4.2 Natural sampler

Let us consider an analog continuous time signal $x(t)$ to be sampled at the rate of f_s Hz and f_s is higher than Nyquist rate such that sampling theorem is satisfied. A sampled signal

1.4.3 Flat Top Sampling or Rectangular Pulse Sampling

Basic Principle

This is also a practically possible sampling method. Natural sampling is little complex, but it is very easy to get flat top samples. The top of the samples remains constant and equal to instantaneous value of baseband signal $x(t)$ at the start of sampling. The duration of each sample is τ and sampling rate is equal to $f_s = \frac{1}{T_s}$.

Generation of flat top samples

Fig.1.4.5 (a) shows the functional diagram of sample and hold circuit generating flat top samples and Fig. 1.4.5 (b) shows waveforms.

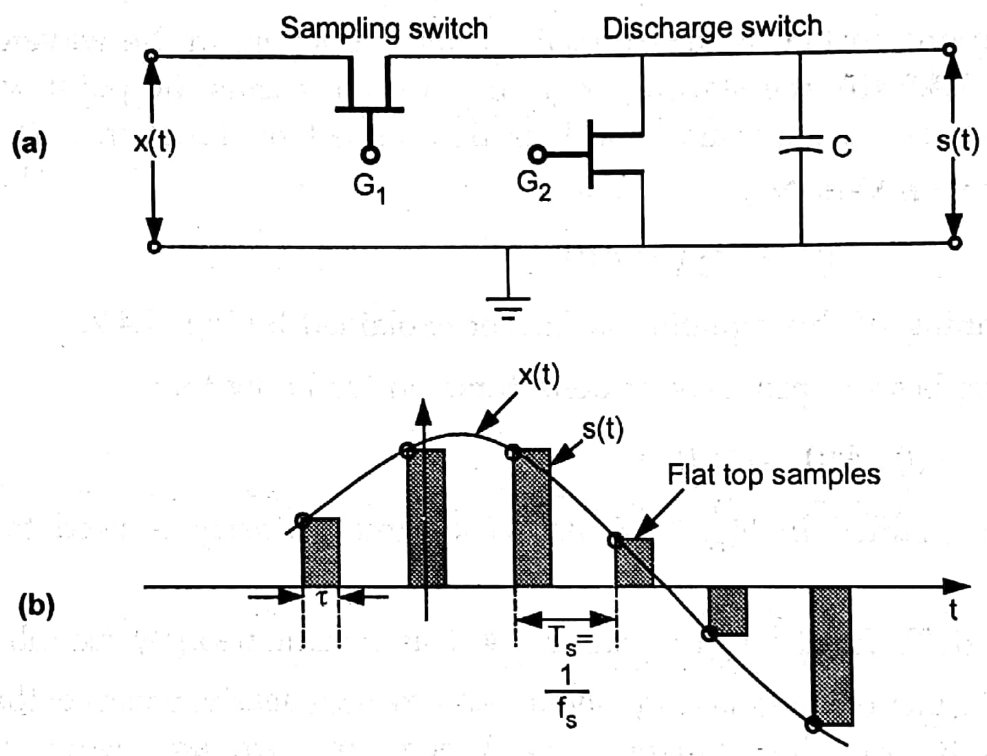
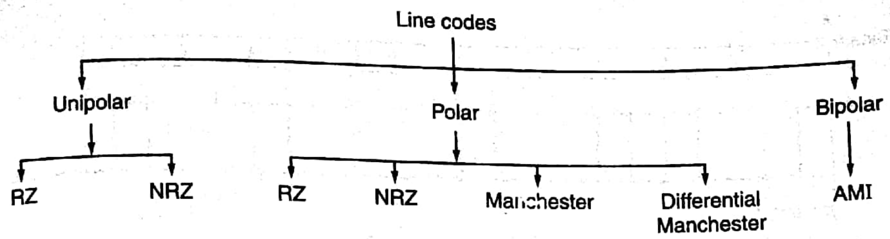


Fig. 1.4.5 (a) Sample and hold circuit generating flat top sampling (b) Waveforms of flat top sampling

Normally the width of the pulse in flat top sampling and natural sampling is increased as far as possible to reduce the transmission bandwidth.

Explanation of Flat top Sampled PAM

Here we can see from Fig. 1.4.5 (b) that only starting edge of the pulse represents instantaneous value of the baseband signal $x(t)$. The flat top pulse of $s(t)$ is mathematically equivalent to the convolution of instantaneous sample and pulse $h(t)$ as shown in Fig. 1.4.6.



Basically, the line codes are divided into following three categories:

1. Unipolar codes
2. Polar codes
3. Bipolar codes

1. Unipolar Codes

Basically, unipolar codes use only one voltage level other than zero. Hence, the encoded signal will have either +A volts value or 0. These codes are very simple and primitive and are almost absolute now-a-days.

2. Polar Codes

Polar coding uses two voltage levels other than zero such as + A/2 and – A/2 volts. This brings the dc level for some codes to zero which is a desired characteristics.

3. Bipolar Codes

Basically, bipolar coding uses three voltage levels positive, negative and zero which is similar to polar codes. However, here, the zero level is always used for representing the '0' in the data stream at the input.

13.9. VARIOUS PAM FORMATS OR LINE CODES

Some of the important PAM formats or line coding techniques are as under:

(i) Non-return to zero (NRZ) and return to zero (RZ) unipolar format.

(ii) NRZ and RZ polar format.

(iii) Non-return to zero bipolar format.

(iv) Manchester format.

(v) Polar quaternary NRZ format.

All the formats have been shown for a binary message 10110100. Figure 13.8 shows various PAM formats or line codes.

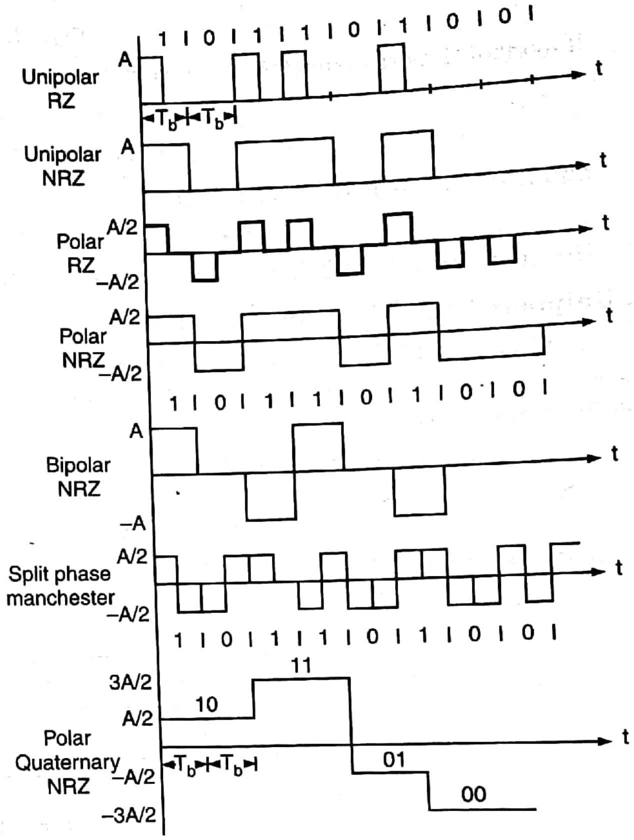


Fig. 13.8. Various digital PAM signals formats, (a) Unipolar RZ, (b) Unipolar NRZ, (c) Polar RZ, (d) Polar NRZ, (e) Bipolar NRZ, (f) Split phase Manchester, (g) Polar quaternary NRZ

13.10. UNIPOLAR RZ AND NRZ

1. Definition

In unipolar format, the waveform has a single polarity. The waveform can have +5 or +12 volts when high. The waveform is simple on-off.

2. Unipolar RZ* : Waveform and Expression

In the unipolar RZ form, the waveform has zero value when symbol '0' is transmitted and waveform has 'A' volts when '1' is transmitted. In RZ form, the 'A' volts is present for $T_b/2$ period if symbol '1' is transmitted and for remaining $T_b/2$, waveform returns to zero value, i.e., for unipolar RZ form, we have

*The RZ waveforms consist of unipolar-RZ, bipolar RZ, and RZ-AMI. These codes find application in baseband data transmission and in magnetic recording. With unipolar-RZ, a one is represented by a half-bit-wide pulse, and a zero is represented by the absence of a pulse. With bipolar RZ, the ones and zeros are represented by opposite-level pulses that are one-half bit wide.

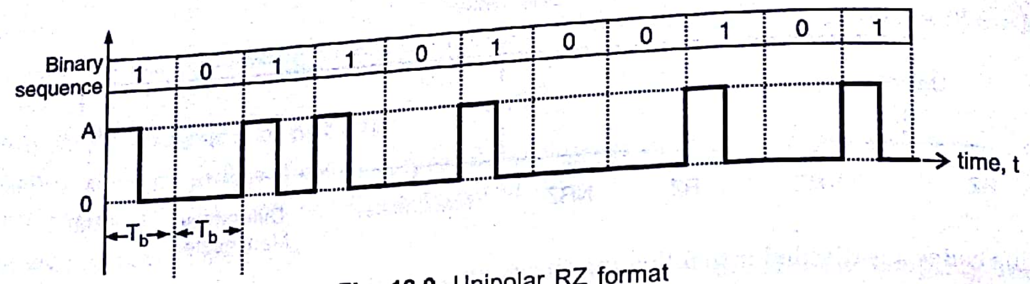


Fig. 13.9. Unipolar RZ format

If symbol '1' is transmitted, then we have

$$x(t) = \begin{cases} A \text{ for } 0 \le t < T_b/2 & \text{(Half interval)} \\ 0 \text{ for } T_b/2 \le t < T_b & \text{(Half interval)} \end{cases} \quad ...(13.7)$$

and if symbol '0' is transmitted, then

$$x(t) = 0 \quad \text{for} \quad 0 \le t < T_b \quad \text{(complete interval)} \quad ...(13.8)$$

Hence, in unipolar RZ format, each pulse returns to a zero value. Figure 13.9 shows this signal format.

3. Unipolar NRZ*: Waveform and Expression

A unipolar NRZ (i.e., not return to zero) format is shown in figure 13.10. When symbol '1' is to be transmitted, the signal has 'A' volts for full duration. When symbol '0' is to be transmitted, the signal has zero volts (i.e., no signal) for complete symbol duration.

Thus, for unipolar NRZ format,

If symbol '1' is transmitted, we have

$$x(t) = A \quad \text{for} \quad 0 \le t < T_b \quad \text{(complete interval)} \quad ...(13.9)$$

If symbol '0' is transmitted, we have

$$x(t) = 0 \quad \text{for} \quad 0 \le t < T_b \quad \text{(complete interval)} \quad ...(13.10)$$

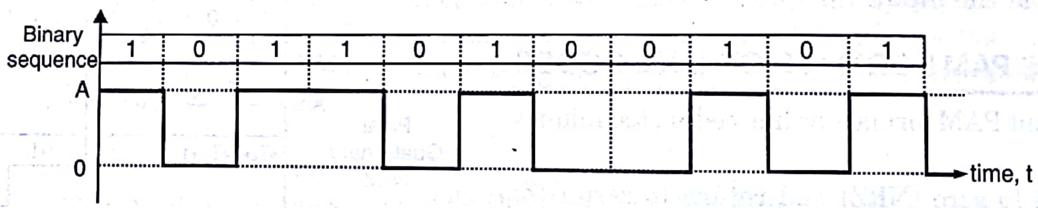


Fig. 13.10. Unipolar NRZ format

4. Important Points

(i) For NRZ format, it may be observed that the pulse does not return to zero on its own. If symbol '0' is to be transmitted, then pulse becomes zero.

(ii) Internal computer waveforms are usually of unipolar NRZ type.

(iii) Because, there is no separation between the pulses, therefore, the receiver needs synchronization to detect unipolar NRZ pulse.

(iv) As compared to RZ format, NRZ pulse width (pulse to pulse interval is same) is more. Thus, energy of the pulse is more.

(v) However, unipolar format has some average DC value. This DC value does not carry any information.

13.11. POLAR RZ AND NRZ**

1. Polar RZ : Waveform and Expression

In the polar RZ format, symbol '1' is represented by positive voltage polarity whereas symbol '0' is represented by negative voltage polarity. Because this is RZ format, the pulse is transmitted only for half duration. Thus, for polar RZ, if symbol '1' is transmitted, then

* With NRZ-M, the one, or mark, is represented by a change in level, and the zero, or space, is represented by no change in level. This is often referred to as different encoding. NRZ-M is used primarily in magnetic tap recording. NRZ-S is the complement of NRZ-M: A one is represented by no change in level, and a zero is represented by a change in level.

** The duty cycle of a binary pulse can be used to categorize the type of transmission. If the binary pulse is maintained for the entire bit time, this is called non return to zero (NRZ). If the active time of the binary pulse is less than 100% of the bit time, this is called return to zero (RZ).

$$x(t) = \begin{cases} +\dfrac{A}{2} & \text{for } 0 \le t < T_b/2 \\ 0 & \text{for } T_b/2 \le t < T_b \end{cases} \qquad ...(13.11)$$

and if symbol '0' is transmitted, then

$$x(t) = \begin{cases} -\dfrac{A}{2} & \text{for } 0 \le t < T_b/2 \\ 0 & \text{for } T_b/2 \le t < T_b \end{cases} \qquad ...(13.12)$$

Polar RZ waveform has been shown in figure 13.11.

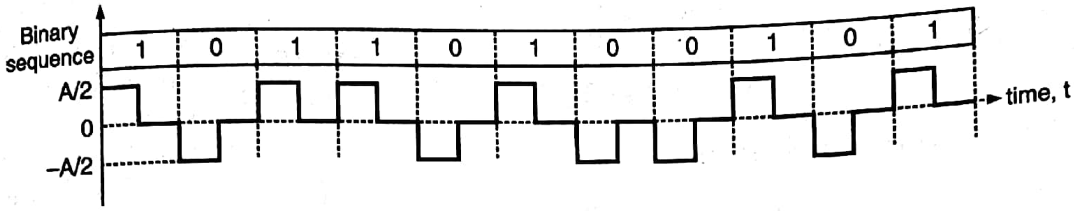


Fig. 13.11. Polar RZ format

1. Polar NRZ : Waveform and Expression

The polar NRZ is shown in figure 13.12. In polar NRZ format, symbol '1' is represented by positive polarity whereas symbol '0' is represented by negative polarity. These polarities are maintained over the complete pulse duration i.e., for polar NRZ, we have

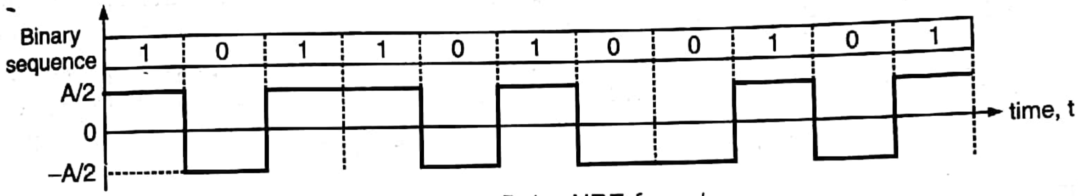


Fig. 13.12. Polar NRZ format

If symbol '1' is transmitted, then

$$x(t) = +\frac{A}{2} \qquad \text{for} \qquad 0 \le t < T_b \qquad ...(13.13)$$

and if symbol '0' is transmitted, then

$$x(t) = -\frac{A}{2} \qquad \text{for} \qquad 0 \le t < T_b \qquad ...(13.14)$$

Important Points

(i) Since polar RZ and NRZ formats are bipolar, therefore, the average DC value is minimum in these waveforms.

(ii) If probabilities of occurrence of symbols '1' and '0' are same, then average DC components of the waveform would be zero.

13.12. BIPOLAR NRZ [ALTERNATE MARK INVERSION (AMI)]

1. Definition

In this format, the successive '1's are represented by pulses with alternate polarity and '0's are represented by no pulses.

2. Waveform

Figure 13.13 illustrates the bipolar NRZ or AMI waveform. If there are even number of 1's, the DC component of the waveform would be zero. The advantage of this format is that the ambiguities due to transmission sign inversion are eliminated.

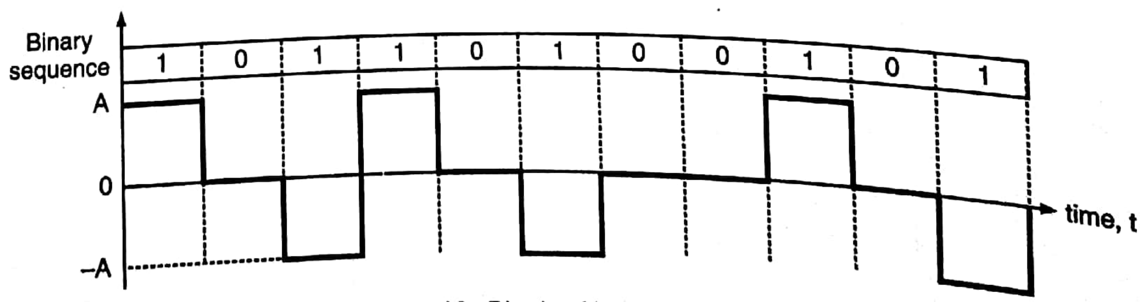


Fig. 13.13. Bipolar NRZ format (AMI)

13.13. SPLIT PHASE MANCHESTER FORMAT

1. Definition and Waveform

This type of waveform is shown in figure 13.14. In this case, if symbol '1' is to be transmitted, then a positive half interval pulse is followed by a negative half interval pulse. If symbol '0' is to be transmitted, then a negative half interval pulse is followed by a positive half interval pulse. Hence, for any symbol the pulse takes positive as well as negative value.

2. Mathematical Expressions

If symbol '1' is to be transmitted, then

$$x(t) = \begin{cases} \dfrac{A}{2} & \text{for } 0 \le t < \dfrac{T_b}{2} \\ -\dfrac{A}{2} & \text{for } \dfrac{T_b}{2} \le t < T_b \end{cases} \quad \text{...(13.15)}$$

and if symbol '0' is to be transmitted, then

$$x(t) = \begin{cases} -\dfrac{A}{2} & \text{for } 0 \le t < \dfrac{T_b}{2} \\ \dfrac{A}{2} & \text{for } \dfrac{T_b}{2} \le t < T_b \end{cases} \quad \text{...(13.16)}$$

DO YOU KNOW?

The various line codes are also known by other names. For example, polar NRZ is also called NRZ-L, where L denotes the normal logical level assignment. Bipolar RZ is also called RZ-AMI where AMI denotes alternate mark (binary 1) inversion.

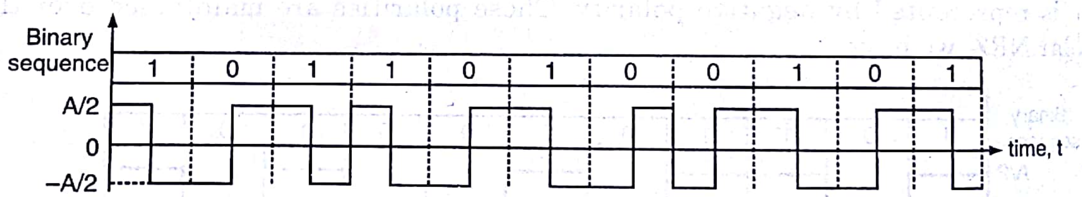


Fig. 13.14. Split phase manchester format

3. Important Points

(i) The primary advantage of this format is that irrespective of the probability of occurrence of symbol '1' and '0' the waveform has zero average value. Therefore, by this mode, the power saving is quite more.

(ii) However, the drawback of this format is that it requires absolute sense of polarity at the receiver end*.

DO YOU KNOW?

The Manchester NRZ line code has the advantage of always having a 0-dc value, regardless of the data sequence, but it has twice the bandwidth of the unipolar NRZ or polar NRZ code because the pulses are half the width.

13.14. POLAR QUATERNARY NRZ FORMAT

$$2 \quad \overline{2T_b} \qquad ...(13.17)$$

13.15. HIGH DENSITY BIPOLAR (HDB) SIGNALLING

1. Definition

In case of bipolar NRZ or AMI signal, the transmitted signal is equal to zero when a binary 0 is to be transmitted. This is true even for the unipolar RZ and unipolar NRZ signals. The absence of transmitted signal can cause problems in synchronization at the receiver, if long sequence of binary 0's are being transmitted. This problem can be solved by adding (transmitting) pulses when long strings of 0's exceeding a number n are being transmitted. This type of coding is called as High Density Bipolar coding. It is denoted by HDBN. Here, N = 1, 2, 3, ... The most widely used HDB format is with N = 3 i.e., HDB3.

2. Waveforms

In the string of message bits, when (N + 1) or minimum number of zeros occur, they are replaced by special binary sequences of (N + 1) length. As shown in figure 13.16, these sequences contain some binary 1's which are necessary for synchronization at the receiver end. The (N + 1) long special sequences for the HDB3 coding are 000V and B00V where B and V both are considered to be binary 1's. When the number of consecutive zeros exceed (N + 1) i.e., 4 in case of HDB3, the above mentioned special sequences are inserted as shown in figure 13.16.

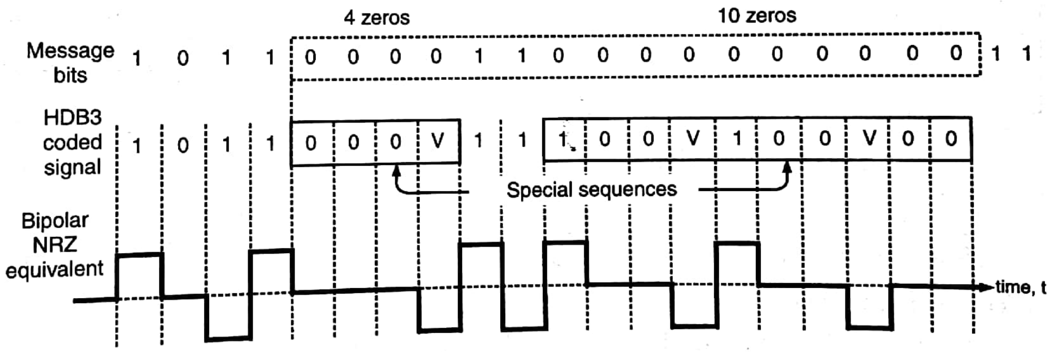
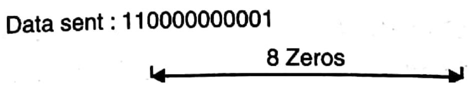


Fig. 13.16.

13.16. B8ZS LINE CODE



removed and data rate can be reduced.

1.11 Differential Pulse Code Modulation

1.11.1 Redundant Information in PCM

The samples of a signal are highly corrected with each other. This is because any signal does not change fast. That is its value from present sample to next sample does not differ by large amount. The adjacent samples of the signal carry the same information with little difference. When these samples are encoded by standard PCM system, the resulting encoded signal contains redundant information. Fig. 1.11.1 illustrates this.

Fig. 1.11.1 shows a continuous time signal $x(t)$ by dotted line. This signal is sampled by flat top sampling at intervals $T_s, 2T_s, 3T_s \dots nT_s$. The sampling frequency is selected to be higher than nyquist rate. The samples are encoded by using 3 bit (7 levels) PCM. The sample is quantized to the nearest digital level as shown by small

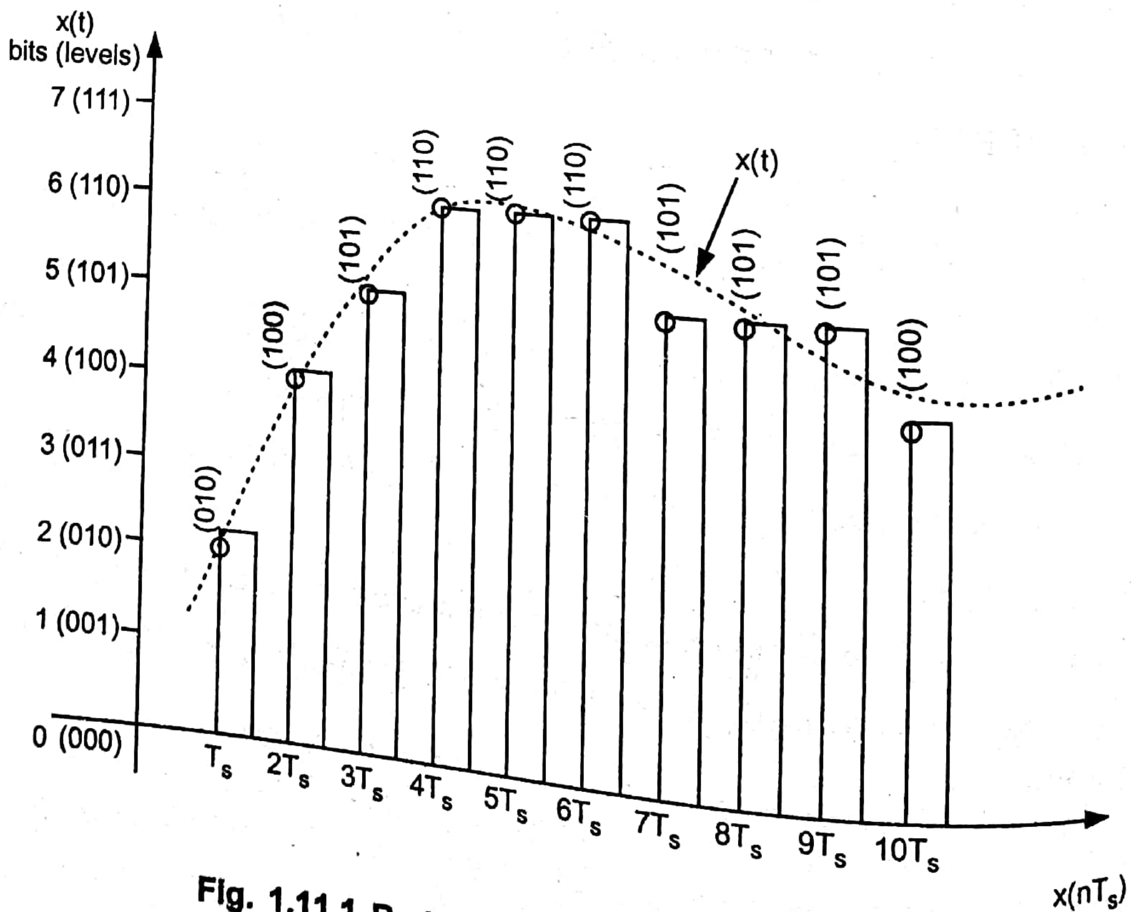


Fig. 1.11.1 Redundant Infor...

circles in the diagram. The encoded binary value of each sample is written on the top of the samples. We can see from Fig. 1.11.1 that the samples taken at $4T_s, 5T_s$ and $6T_s$ are encoded to same value of (110). This information can be carried only by one sample. But three samples are carrying the same information means it is redundant. Consider another example of samples taken at $9T_s$ and $10\,T_s$. The difference between these samples is only due to last bit and first two bits are redundant, since they do not change.

1.11.2 Principle of DPCM

If this redundancy is reduced, then overall bit rate will decrease and number of bits required to transmit one sample will also be reduced. This type of digital pulse modulation scheme is called Differential Pulse Code Modulation.

1.11.3 DPCM Transmitter

The differential pulse code modulation works on the principle of prediction. The value of the present sample is predicted from the past samples. The prediction may not be exact but it is very close to the actual sample value. Fig. 1.11.2 shows the transmitter of Differential Pulse Code Modulation (DPCM) system. The sampled signal is denoted by $x(n\,T_s)$ and the predicted signal is denoted by $\hat{x}(n\,T_s)$. The comparator finds out the difference between the actual sample value $x(n\,T_s)$ and predicted sample value $\hat{x}(n\,T_s)$. This is called error and it is denoted by $e(n\,T_s)$. It can be defined as,

$$e(n\,T_s) = x(n\,T_s) - \hat{x}(n\,T_s) \quad \text{... (1.11.1)}$$

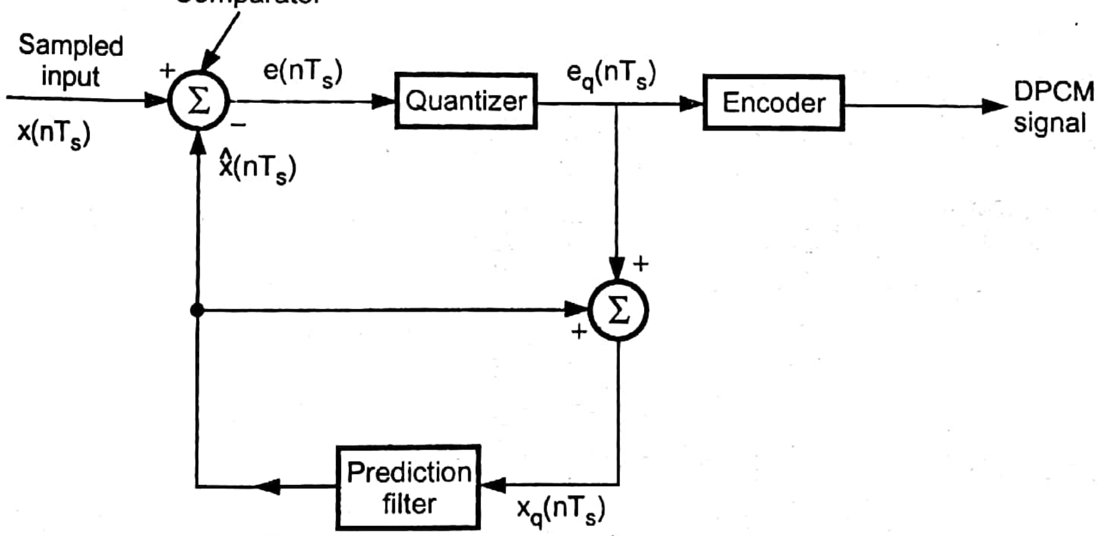


Fig. 1.11.2 Differential pulse code modulation transmitter

Thus error is the difference between unquantized input sample $x(n\,T_s)$ and prediction of it $\hat{x}(n\,T_s)$. The predicted value is produced by using a prediction fi... The quantizer output signal $e_q\,(n\,T_s)$ and previous prediction is added and gi...

input to the prediction filter. This signal is called $x_q(nT_s)$. This makes the prediction more and more close to the actual sampled signal. We can see that the quantized error signal $e_q(nT_s)$ is very small and can be encoded by using small number of bits. Thus number of bits per sample are reduced in DPCM.

The quantizer output can be written as,

$$e_q(nT_s) = e(nT_s) + q(nT_s) \quad \text{... (1.11.2)}$$

Here $q(nT_s)$ is the quantization error. As shown in Fig. 1.11.2, the prediction filter input $x_q(nT_s)$ is obtained by sum $\hat{x}(nT_s)$ and quantizer output i.e.,

$$x_q(nT_s) = \hat{x}(nT_s) + e_q(nT_s) \quad \text{... (1.11.3)}$$

Putting the value of $e_q(nT_s)$ from equation 1.11.2 in the above equation we get,

$$x_q(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s) \quad \text{... (1.11.4)}$$

Equation 1.11.1 is written as,

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

$$\therefore \quad e(nT_s) + \hat{x}(nT_s) = x(nT_s) \quad \text{... (1.11.5)}$$

$\therefore$ Putting the value of $e(nT_s) + \hat{x}(nT_s)$ from above equation into equation 1.11.4 we get,

$$x_q(nT_s) = x(nT_s) + q(nT_s) \quad \text{... (1.11.6)}$$

Thus the quantized version of the signal $x_q(nT_s)$ is the sum of original sample value and quantization error $q(nT_s)$. The quantization error can be positive or negative. Thus equation 1.11.6 does not depend on the prediction filter characteristics.

1.11.4 Reconstruction of DPCM Signal

Fig. 1.11.3 shows the block diagram of DPCM receiver.

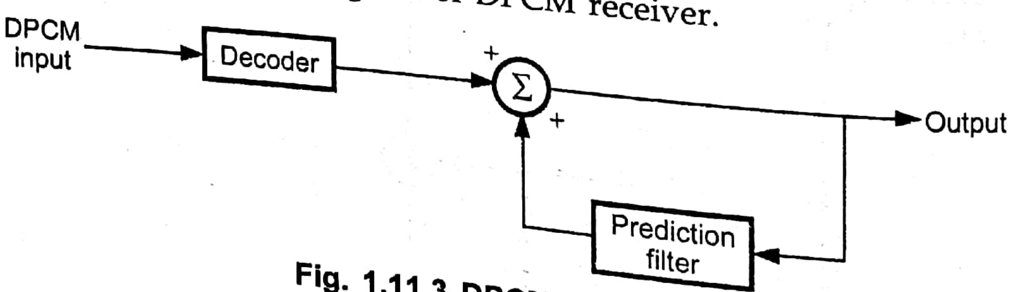


Fig. 1.11.3 DPCM receiver

The decoder first reconstructs the quantized error signal from incoming binary signal. The prediction filter output and quantized error signals are summed up to give the quantized version of the original signal. Thus the signal at the receiver differs from actual signal by quantization error $q(nT_s)$, which is introduced permanently in the reconstructed signal.

2 Delta Modulation

We have seen in PCM that, it transmits all the bits which are used to code the sample. Hence signaling rate and transmission channel bandwidth are large in PCM. To overcome this problem Delta Modulation is used.

2.1 Delta Modulation

2.1.1 Operating Principle of DM

Delta modulation transmits only one bit per sample. That is the present sample value is compared with the previous sample value and the indication, whether the amplitude is increased or decreased is sent. Input signal $x(t)$ is approximated to step signal by the delta modulator. This step size is fixed. The difference between the input signal $x(t)$ and staircase approximated signal confined to two levels, i.e. $+\delta$ and $-\delta$. If the difference is positive, then approximated signal is increased by one step i.e. 'δ'. If the difference is negative, then approximated signal is reduced by 'δ'. When the step is reduced, '0' is transmitted and if the step is increased, '1' is transmitted. Thus for each sample, only one binary bit is transmitted. Fig. 2.1.1 shows the analog signal $x(t)$ and its staircase approximated signal by the delta modulator.

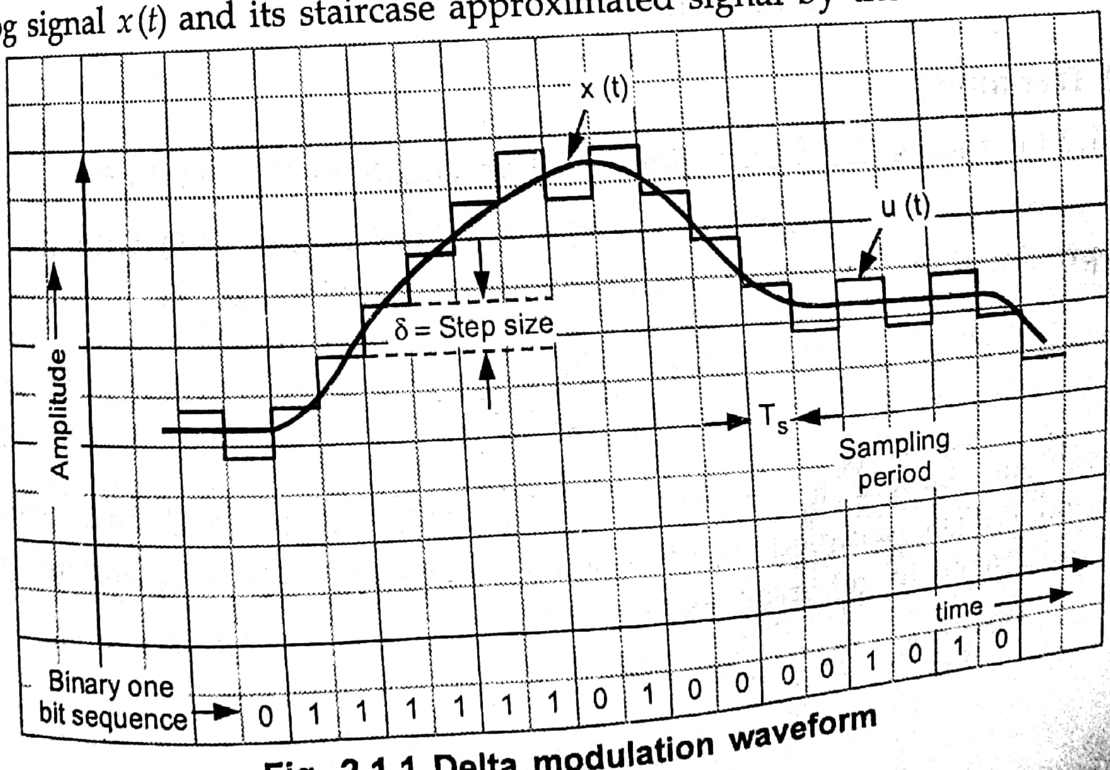


Fig. 2.1.1 Delta modulation waveform

The principle of delta modulation can be explained by the following set of equations. The error between the sampled value of $x(t)$ and last approximated sample is given as,

$$e(nT_s) = x(nT_s) - \hat{x}(nT_s) \qquad \dots (2.1.1)$$

Here, $e(nT_s)$ = Error at present sample

$x(nT_s)$ = Sampled signal of $x(t)$

$\hat{x}(nT_s)$ = Last sample approximation of the staircase waveform.

We can call $u(nT_s)$ as the present sample approximation of staircase output.

Then, $u[(n-1)T_s] = \hat{x}(nT_s)$ $\qquad \dots (2.1.2)$

= Last sample approximation of staircase waveform.

Let the quantity $b(nT_s)$ be defined as,

$$b(nT_s) = \delta \, sgn[e(nT_s)] \qquad \dots (2.1.3)$$

That is depending on the sign of error $e(nT_s)$ the sign of step size δ will be decided. In other words,

$$\begin{aligned} b(nT_s) &= +\delta \quad \text{if} \quad x(nT_s) \geq \hat{x}(nT_s) \\ &= -\delta \quad \text{if} \quad x(nT_s) < \hat{x}(nT_s) \end{aligned} \qquad \dots (2.1.4)$$

If $b(nT_s) = +\delta$; binary '1' is transmitted

and if $b(nT_s) = -\delta$; binary '0' is transmitted.

T_s = Sampling interval.

2.1.2 DM Transmitter

Fig. 2.1.2 (a) shows the transmitter based on equations 2.1.3 to 2.1.5.

The summer in the accumulator adds quantizer output ($\pm\delta$) with the previous sample approximation. This gives present sample approximation. i.e.,

$$u(nT_s) = u(nT_s - T_s) + [\pm\delta] \quad \text{or}$$

$$= u[(n-1)T_s] + b(nT_s) \qquad \dots (2.1.5)$$

The previous sample approximation $u[(n-1)T_s]$ is restored by delaying one sample period T_s. The sampled input signal $x(nT_s)$ and staircase approximated signal $\hat{x}(nT_s)$ are subtracted to get error signal $e(nT_s)$.

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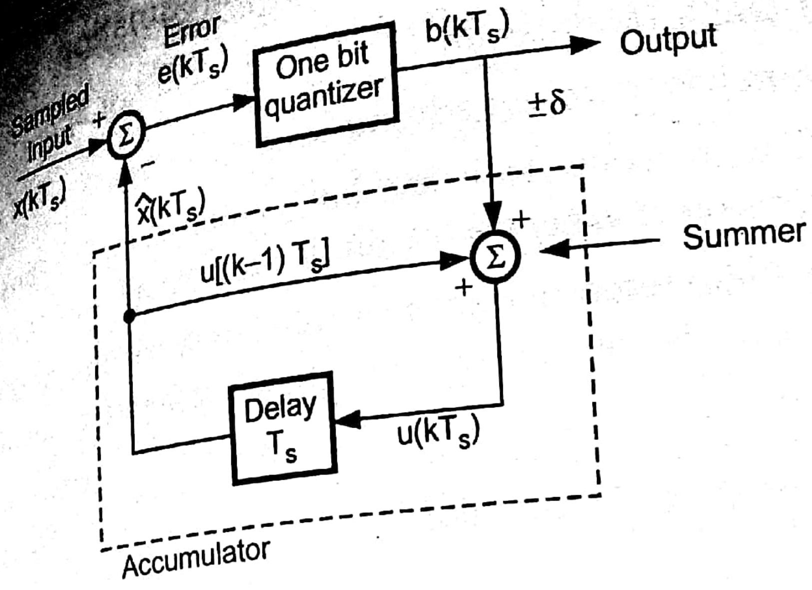
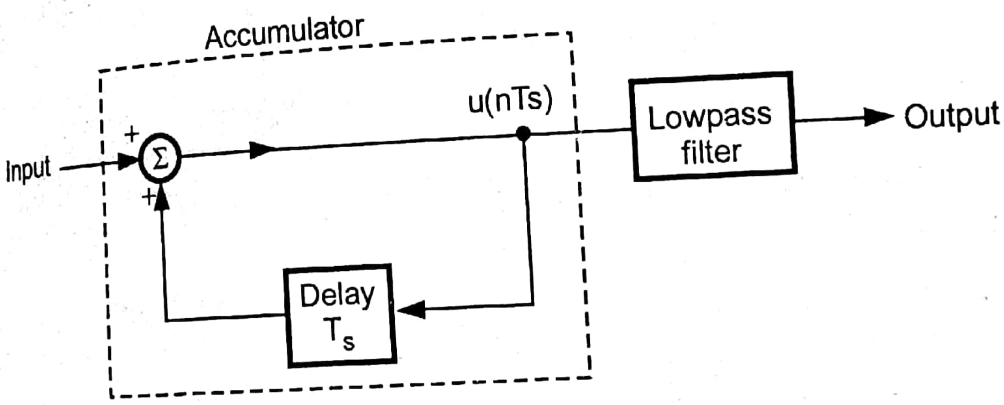


(a)



(b)

Fig. 2.1.2 (a) Delta modulation transmitter and (b) Delta modulation receiver

Depending on the sign of $e(nT_s)$ one bit quantizer produces an output step of $+\delta$ or $-\delta$. If the step size is $+\delta$, then binary '1' is transmitted and if it is $-\delta$, then binary '0' is transmitted.

2.1.3 DM Receiver

At the receiver shown in Fig. 2.1.2 (b), the accumulator and low-pass filter are used. The accumulator generates the staircase approximated signal output and is delayed by one sampling period T_s. It is then added to the input signal. *If input is* binary '1' then it adds $+\delta$ step to the previous output (which is delayed). *If input is* binary '0' then one step 'δ' is subtracted from the delayed signal. The low-pass filter has the cutoff frequency equal to highest frequency in $x(t)$. This filter smoothen the staircase signal to reconstruct $x(t)$.

2.2 Advantages and Disadvantages of Delta Modulation

2.2.1 Advantages of Delta Modulation

The delta modulation has following advantages over PCM,

1. Delta modulation transmits only one bit for one sample. Thus the signaling rate and transmission channel bandwidth is quite small for delta modulation.
2. The transmitter and receiver implementation is very much simple for delta modulation. There is no analog to digital converter involved in delta modulation.

2.2.2 Disadvantages of Delta Modulation

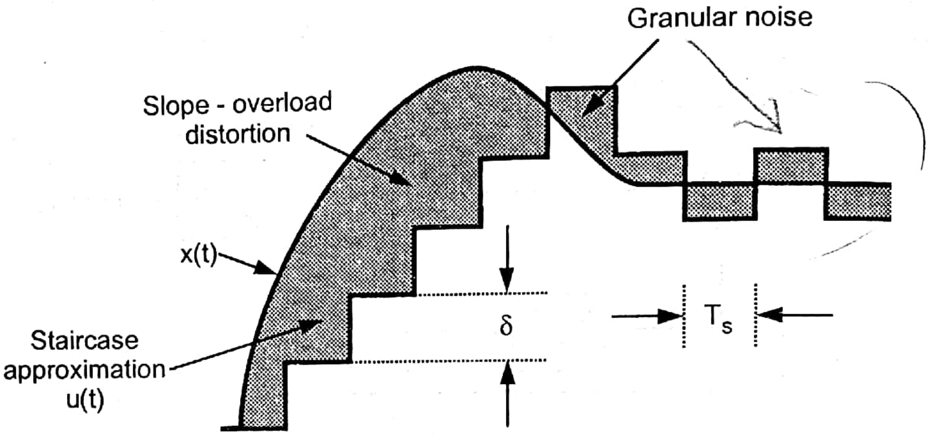


Fig. 2.2.1 Quantization errors in delta modulation

The delta modulation has two drawbacks -

2.2.2.1 Slope Overload Distortion (Startup Error)

This distortion arises because of the large dynamic range of the input signal.

As can be seen from Fig. 2.2.1 the rate of rise of input signal $x(t)$ is so high that the staircase signal cannot approximate it, the step size 'δ' becomes too small for staircase signal $u(t)$ to follow the steep segment of $x(t)$. Thus there is a large error between the staircase approximated signal and the original input signal $x(t)$. This error is called *slope overload distortion*. To reduce this error, the step size should be increased when slope of signal of $x(t)$ is high.

Since the step size of delta modulator remains fixed, its maximum or minimum slopes occur along straight lines. Therefore this modulator is also called Linear Delta Modulator (LDM).

2.2.2.2 Granular Noise (Hunting)

Granular noise occurs when the step size is too large compared to small variations in the input signal. That is for very small variations in the input signal, the staircase signal is changed by large amount (δ) because of large step size. Fig. 2.2.1 shows when the input signal is almost flat, the staircase signal $u(t)$ keeps on oscillating by ± δ around the signal. The error between the input and approximated signal is called granular noise. The solution to this problem is to make step size small.

Thus large step size is required to accommodate wide dynamic range of the input signal (to reduce slope overload distortion) and small steps are required to reduce granular noise. Adaptive delta modulation is the modification to overcome these errors.

Example 2.2.1 : *Using predictability theory, prove that transmission of encoded error signal (rather than encoded signal itself is sufficient for reasonable reconstruction of signal. With the help of block schematic suggest any one technique to transmit and receive encoded errors. What are the limitations and advantages of such techniques with reference to linear or uniform PCM ?*

Solution : Here the technique that uses predictibility theory is basically delta modulation. The output of the accumulator in DM transmitter is given by equation 2.1.5 as,

$$u(nT_s) = u[(n-1)T_s] + b(nT_s) \quad \text{... (2.2.1)}$$

Here $\quad b(nT_s) = \pm\delta \text{ or } \delta \operatorname{sgn}[e(nT_s)]$

Thus $b(nT_s)$ basically represents error signal. Sign of step size 'δ' depends upon whether $e(nT_s)$ is positive or negative.

Now we will show that the signal can be reconstructed only with the help of encoded error signal, i.e. $b(nT_s)$. The accumulator of Fig. 2.1.2(b) acts as a delta modulation receiver. $u(nT_s)$ is the output of accumulator. For simplicity let us drop T_s in equation 2.2.1 Then we get,

$$u(n) = u(n-1) + b(n) \quad \text{... (2.2.2)}$$

Observe that this is recursive equation. Hence $u(n-1)$ can be calculated as,

$$u(n-1) = u(n-2) + b(n-1) \quad \text{... (2.2.3)}$$

Hence equation 2.2.2 becomes,

$$u(n) = u(n-2) + b(n-1) + b(n) \quad \text{... (2.2.4)}$$

From equation 2.2.3 we can calculate $u(n-2)$ as,

$$u(n-2) = u(n-3) + b(n-2)$$

Hence equation 2.2.4 becomes,

$$u(n) = u(n-3) + b(n-2) + b(n-1) + b(n)$$