Module 5

Syllabus:

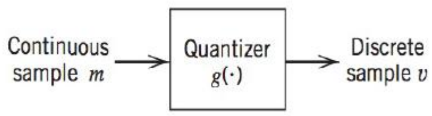
Module-5	
SAMPLING AND QUANTIZATION (Contd): The Quantization Random Process, Quantization Noise, Pulse–Code Modulation: Sampling, Quantization, Encoding, Regeneration, Decoding, Filtering, Multiplexing; Delta Modulation **(7.8 – 7.10 in Text)**, Application examples - (a) Video + MPEG **(7.11 in Text)** and (b) Vocoders(**refer Section 6.8 of Reference Book 1**).	**L1, L2,L3**

THE QUANTIZATION PROCESS

A continuous signal, such as voice, has a continuous range of amplitudes and therefore its samples have a continuous amplitude range. In other words, within the finite amplitude range of the signal, we find an infinite number of amplitude levels.

This means that the original continuous signal may be *approximated* by a signal constructed of discrete amplitudes selected on a minimum error basis from an available set. Clearly, if we assign the discrete amplitude levels with sufficiently close spacing, we may make the approximated signal practically indistinguishable from the original continuous signal.

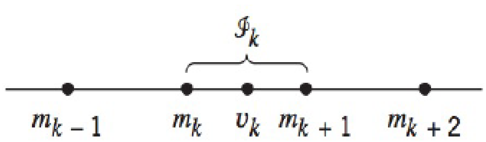
Definition: "Amplitude *quantization* is defined as the process of transforming the sample amplitude $m(nT_s)$ of a message signal $m(t)$ at time $t = nT_s$ into a discrete amplitude $v(nT_s)$ taken from a *finite* set of possible amplitudes."



The quantization process is *memoryless* and *instantaneous*.
Consider a signal amplitude 'm' specified in a range $\mathcal{J}_k$, defined by

$$\mathcal{J}_k: \{m_k < m \leq m_{k+1}\}, \quad k = 1, 2, \dots \dots \dots L$$

L = Number of amplitude levels used in the quantizer.



The quantizer output v equals v_k if the input signal sample m belongs to the interval $\mathcal{J}_k$.

Then, quantizer characteristic can be represented as

$$v = q(m)$$

<u>Transfer characteristics of Quantizer:</u>

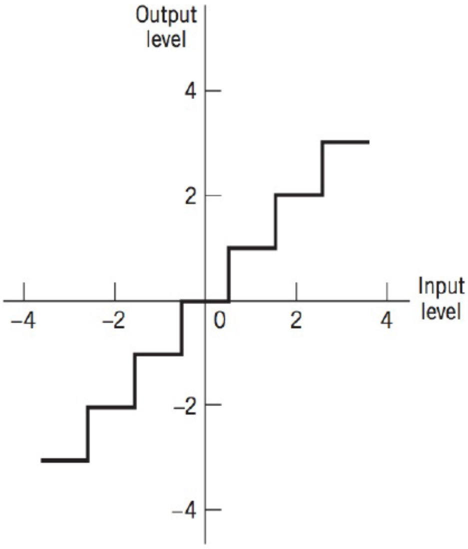
Two types:

1. Mid-tread quantizer

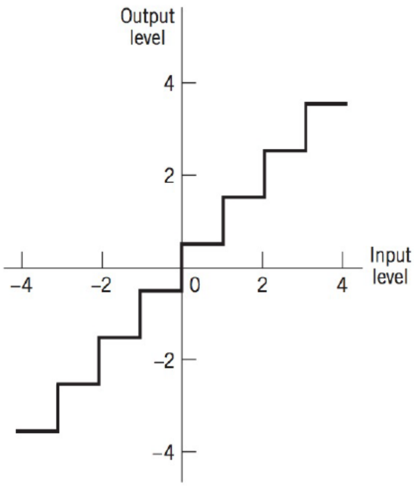
2. Mid-riser quantizer

Mid-tread quantizer:

- The origin lies in the middle of a tread of the staircase-like graph
- The input amplitudes, m_k are called *decision levels* or *decision thresholds.* These 'decision levels' are used to truncate or round-off input sample amplitude.
- Based on the decision level, m_k, the quantizer transforms sample amplitude to output value v_k, represented by one of the levels called *representation levels* or r*econstruction levels*, and the spacing between two adjacent representation levels is called a *quantum* or *step-size.*



Mid-riser quantizer:

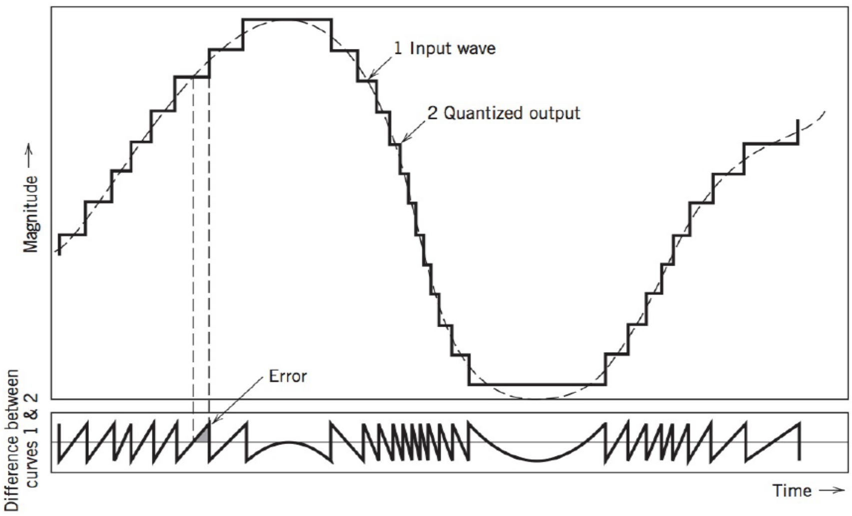


- the origin lies in the middle of a rising part of the staircase like graph.

QUANTIZATION NOISE

The use of quantization introduces an error defined as the difference between the input signal *m* and the output signal v. This error is called *quantization noise.*

$$q_e = m - v$$



Let the quantizer input m be the sample value of a zero-mean random variable *M.*
A quantizer q(.) maps the input random variable *M* of continuous amplitude into a discrete random variable V; their respective sample values m and *v* are related by

$$v = q(m)$$

Let the quantization error be denoted by the random variable Q of sample value q_e. We may thus write

$$q_e = m - v$$

or, correspondingly

$$Q = M - V$$

Signal – to – Noise ratio (SNR) of Quantizer:

Let the quantizer input m be the sample value of a zero-mean random variable *M.*
A quantizer q(.) maps the input random variable *M* of continuous amplitude into a discrete random variable V; their respective sample values m and *v* are related by

$$v = q(m)$$

Let the quantization error be denoted by the random variable Q of sample value q_e. We may thus write

$$q_e = m - v$$

We may thus express the probability density function of the quantization error Q as follows:

Pdf: $$f_Q(q_e) = \begin{cases} \frac{1}{\Delta}, & -\frac{\Delta}{2} < q_e \leq \frac{\Delta}{2} \\ 0, & elsewhere \end{cases}$$

Mean: $E[Q] = 0$

Variance: $E[Q^2] = \sigma_Q^2$

$$\sigma_Q^2 = \int_{-\Delta/2}^{\Delta/2} q_e^2 \, f_Q(q_e) \, dq$$

$$\sigma_Q^2 = \int_{-\Delta/2}^{\Delta/2} q_e^2 \, \frac{1}{\Delta} \, dq$$

$$\sigma_Q^2 = \frac{1}{\Delta} \int_{-\Delta/2}^{\Delta/2} q_e^2 \; dq = \frac{1}{\Delta} \left[\frac{q_e^3}{3} \right]_{-\Delta/2}^{\Delta/2}$$

$$\sigma_Q^2 = \frac{1}{\Delta} \left[\frac{(\Delta/2)^3}{3} - \frac{(-\Delta/2)^3}{3} \right] = \frac{1}{3\Delta} \left[\frac{\Delta^3}{8} + \frac{\Delta^3}{8} \right]$$

$$\boxed{\sigma_Q^2 = \frac{\Delta^2}{12}}$$

Consider then an input *m* of continuous amplitude in the range $(-m_{max},\ m_{max})$. Assuming a uniform quantizer of the midrise type, we find that the step-size of the quantizer is given by

$$\boxed{\Delta = \frac{2m_{max}}{L}}$$

where L = number of representation levels

If R = number of *bits per sample* used in the construction of the binary code,

Therefore, $$\boxed{L = 2^R}$$

$$\Delta = \frac{2m_{max}}{2^R}$$

Now, quantization noise variance becomes

$$\sigma_Q^2 = \frac{\Delta^2}{12} = \frac{(2m_{max}/2^R)^2}{12}$$

$$\sigma_Q^2 = \frac{m_{max}^2}{3\,(2^{2R})}$$

Let P denote the average power of the message signal $m(t)$.

We may then express the *output signal-to-noise ratio* of a uniform quantizer as

$$\boxed{(SNR)_o = \frac{Average\ signal\ power}{Quantization\ noise\ variance} = \frac{P}{\sigma_Q^2}}$$

$$(SNR)_o = \frac{P}{m_{max}^2/3\left(2^{2R}\right)}$$

$$\boxed{(SNR)_o = \left(\frac{3P}{m_{max}^2}\right) 2^{2R}}$$

<u>Sinusoidal Modulating Signal:</u> Let $m(t) = A_m \sin 2\pi f_m t$

Then, signal power P = $\frac{A_m^2}{2}$ and $m_{max} = A_m$

$$(SNR)_o = \left(\frac{3P}{m_{max}^2}\right) 2^{2R}$$

$$(SNR)_o = \left(\frac{3A_m^2/2}{A_m^2}\right) 2^{2R}$$

$$(SNR)_o = \frac{3}{2}(2^{2R})$$

Expressing the signal-to-noise ratio in decibels, we get

$$(SNR)_{o,dB} = 10 \ \log_{10}\left[\frac{3}{2}(2^{2R})\right]$$

$$(SNR)_{o,dB} = 10 \ \log_{10}\left[\frac{3}{2}\right] + 10 \ \log_{10}(2^{2R})$$

$$\boxed{(SNR)_{o,dB} = 1.8 + 6R}$$

This SNR is called "6-dB rule" as the SNR increases by 6-dB for every rise in 1 bit per sample.

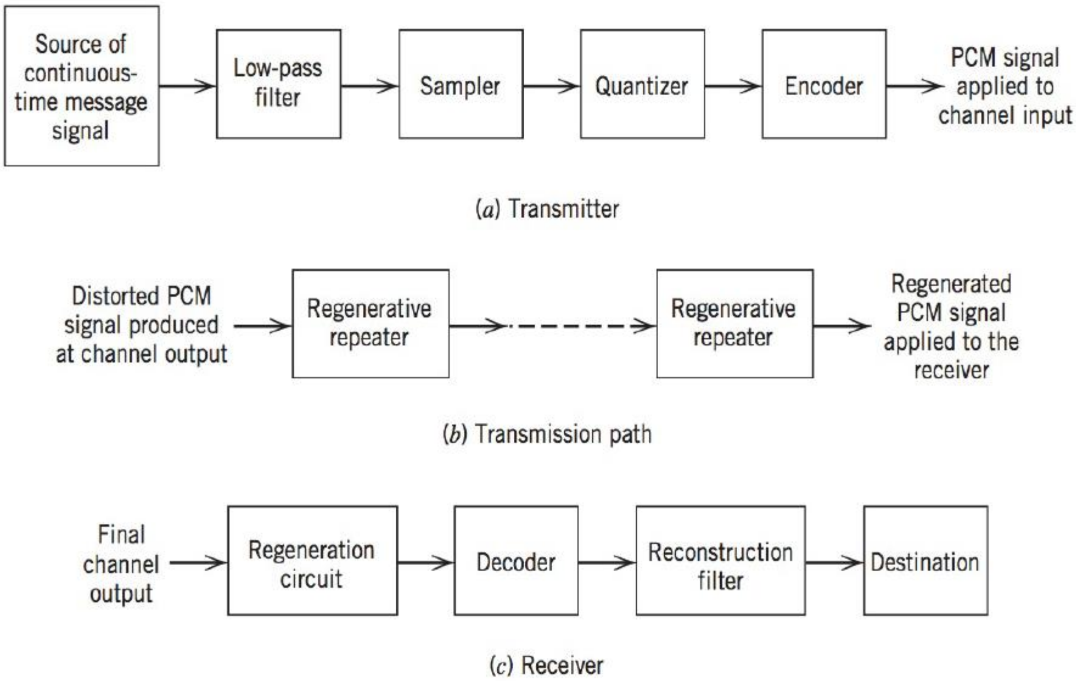
PULSE-CODE MODULATION

- In *pulse-code modulation* (*PCM*) *a message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form in both time and amplitude.*
- The basic operations performed in the transmitter of a PCM system are:
  1. *sampling*,
  2. *quantizing*, and
  3. *encoding*.

  A low-pass filter prior to sampling is included to prevent aliasing of the message signal. The quantizing and encoding operations are usually performed in the same circuit, which is called an *analog-to-digital converter*.

- The basic operations in the receiver are
  1. *regeneration* of impaired signals,
  2. *decoding*, and
  3. *reconstruction* of the train of quantized samples.
- Regeneration also occurs at intermediate points along the transmission path as necessary

- When time-division multiplexing is used, it becomes necessary to synchronize the receiver to the transmitter for the overall system to operate satisfactorily.



Basic operations of PCM system:

<u>SAMPLING</u>**:** The incoming message signal is sampled with a train of narrow rectangular pulses. In order to ensure perfect reconstruction of the message signal at the receiver, the sampling rate must be greater than twice the highest frequency component *W* of the message signal.

$$f_s > 2W$$

<u>QUANTIZATION:</u> The sampled version of the message signal is then quantized, either uniform or non-uniform. The use of a nonuniform quantizer is equivalent to passing the baseband signal through a *compressor* and then applying the compressed signal to a uniform quantizer.
A particular form of compression law that is used in practice is the so called *μ-law* defined By

$$|v| = \frac{\log(1+\mu|m|)}{\log(1+\mu)}$$

where *m* and *v* are the normalized input and output voltages, and *p* is a positive constant.
Another compression law that is used in practice is the so-called *A-law* defined by

$$|v| = \begin{cases} \frac{A|m|}{1+\log A}, & 0 \le |m| \le \frac{1}{A} \\ \frac{1+\log(A|m|)}{1+\log A}, & \frac{1}{A} \le |m| \le 1 \end{cases}$$

<u>ENCODING:</u> To make the transmitted signal more robust to noise, interference, and other channel degradations, we require the use of an *encoding process* to translate the discrete set of sample values to a more appropriate form of signal. Any plan for representing each

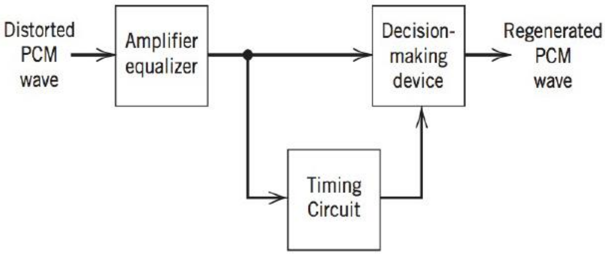
of this discrete set of values as a particular arrangement of discrete events is called a *code*. One of the discrete events in a code is called a *code element* or *symbol*.

For example, the presence or absence of a pulse is a symbol. A particular arrangement of symbols used in a code to represent a single value of the discrete set is called a *code word* or *character*.

In a *binary code*, each symbol may be either of two distinct values or kinds, such as the presence or absence of a pulse. The two symbols of a binary code are customarily denoted as 0 and 1. In a *ternary code*, each symbol may be one of three distinct values or kinds, and so on for other codes.

However, *the maximum advantage over the effects of noise in a transmission medium is obtained by using a binary code, because a binary symbol withstands a relatively high level of noise and is easy to regenerate.*

<u>REGENERATION:</u> The ability to control the effects of distortion and noise produced by transmitting a digital signal through a channel is accomplished by reconstructing the signal by means of a chain of *regenerative repeaters* located at sufficiently close spacing along the transmission route.



Three basic functions are performed by a regenerative repeater:

1. *equalization*,
2. *timing*, and
3. *decision making*.

- The equalizer shapes the received pulses so as to compensate for the effects of amplitude and phase distortions produced by the transmission characteristics of the channel.
- The timing circuitry provides a periodic pulse train, derived from the received pulses, for sampling the equalized pulses at the instants of time where the signal-to-noise ratio is a maximum.
- The sample so extracted is compared to a predetermined *threshold* in the decision-making device. In each bit interval a decision is then made whether the received symbol is a 1 or a 0 on the basis of whether the threshold is exceeded or not.

<u>DECODING:</u> The first operation in the receiver is to regenerate (i.e., reshape and clean up) the received pulses one last time. These clean pulses are then regrouped into code words and decoded (i.e., mapped back) into a quantized PAM signal. The *decoding* process involves generating a pulse the amplitude of which is the linear sum of all the pulses in the

code word, with each pulse being weighted by its place value $(2^0, 2^1, 2^2 \ldots .\, 2^{R-1})$ in the code, where R is the number of bits per sample.

FILTERING: The final operation in the receiver is to recover the message signal wave by passing the decoder output through a low-pass reconstruction filter whose cut-off frequency is equal to the message bandwidth *W*. Assuming that the transmission path is error free, the recovered signal includes no noise except for the initial distortion introduced by the quantization process.

Non-uniform Quantization

In certain applications, however, it is preferable to use a variable separation between the representation levels. By using a *nonuniform quantizer* with the feature that the step-size increases as the separation from the origin of the input-output amplitude characteristic is increased, the large end step of the quantizer can take care of possible excursions of the voice signal into the large amplitude ranges that occur relatively infrequently.
In other words, the weak passages, which need more protection, are favored at the expense of the loud passages. In this way, a nearly uniform percentage precision is achieved throughout the greater part of the amplitude range of the input signal, with the result that fewer steps are needed than would be the case if a uniform quantizer were used.

The use of a nonuniform quantizer is equivalent to passing the baseband signal through a *compressor* and then applying the compressed signal to a uniform quantizer.

There are two types of compression techniques:

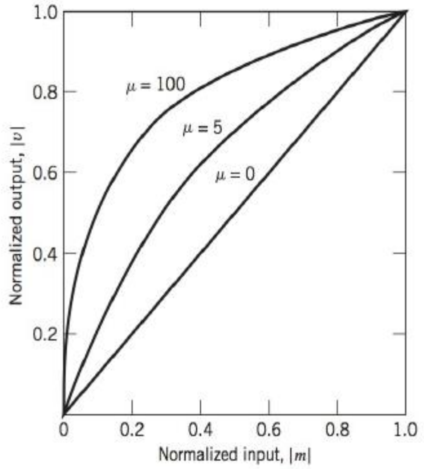
1. $\mu -$ law companding
2. $A -$ law companding

$\mu -$ law:

The μ *-law* is defined by

$$|v| = \frac{\log(1+\mu|m|)}{\log(1+\mu)}$$

where *m* and *v* are the normalized input and output voltages, and *p* is a positive constant.
the μ law for varying μ.



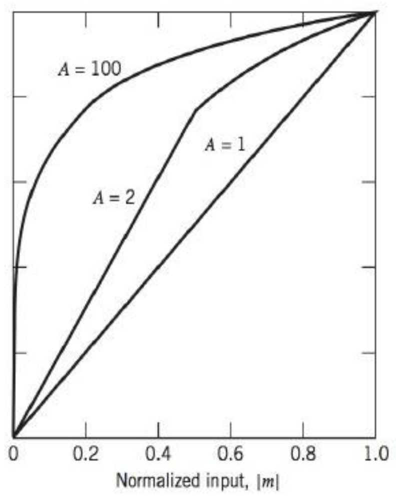
For a given value of μ, the reciprocal slope of the compression curve, which defines the quantum steps, is given by the derivative of $|m|$ with respect to $|v|$; that is,

$$\frac{d|m|}{d|v|} = \frac{\log(1+\mu)}{\mu}\left(1 + \mu|m|\right)$$

Therefore, that the μ -law is neither strictly linear nor strictly logarithmic, but it is approximately linear at low input levels corresponding to $\mu|m| \ll 1$, and approximately logarithmic at high input levels corresponding to $\mu|m| \gg 1$.

A-law: *The A-law* defined by

$$|v| = \begin{cases} \dfrac{A|m|}{1+\log A}, & 0 \le |m| \le \dfrac{1}{A} \\ \dfrac{1+\log(A|m|)}{1+\log A}, & \dfrac{1}{A} \le |m| \le 1 \end{cases}$$



Thus, the quantum steps over the central linear segment, which have the dominant effect on small signals, are diminished by the factor A/(l + log A). This is typically about 25 dB in practice, as compared with uniform quantization.

In order to restore the signal samples to their correct relative level, we must, of course, use a device in the receiver with a characteristic complementary to the compressor. Such a device is called an *expander*. Ideally, the compression and expansion laws are exactly inverse so that, except for the effect of quantization, the expander output is equal to the compressor input. The combination of a compressor and an *expander* is called a *compander*.

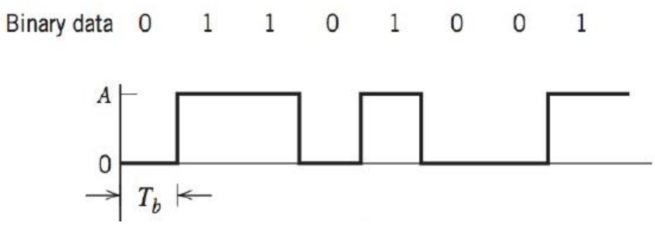
In actual PCM systems, the companding circuitry does not produce an exact replica of the nonlinear compression curves. Rather, it provides a *piecewise linear* approximation to the desired curve.

LINE CODES

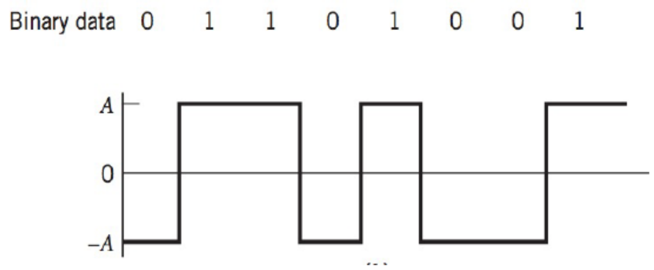
It is in a *line code* that a binary stream of data takes on an electrical representation. Any one of several line codes can be used for the electrical representation of a binary data stream. Some of the important line codes are:

1. Unipolar nonreturn-to-zero (NRZ) signaling
2. Unipolar return-to-zero (RZ) signaling
3. Polar nonreturn-to-zero (NRZ) signaling
4. Polar return-to-zero (RZ) signaling
5. Bipolar nonreturn-to-zero (NRZ) signaling
6. Bipolar return-to-zero (NRZ) signaling
7. Split-phase or Manchester coding

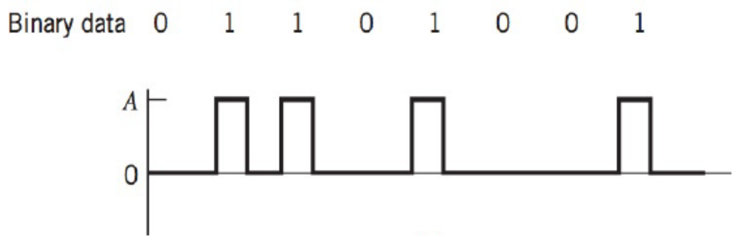
1. **Unipolar Nonreturn-to-Zero (NRZ) Signaling:** In this line code, symbol 1 is represented by transmitting a pulse of amplitude A for the duration of the symbol, and symbol 0 is represented by switching off the pulse. This line code is also referred to as *on-off signaling.* A disadvantage of on-off signaling is the waste of power due to the transmitted DC level.



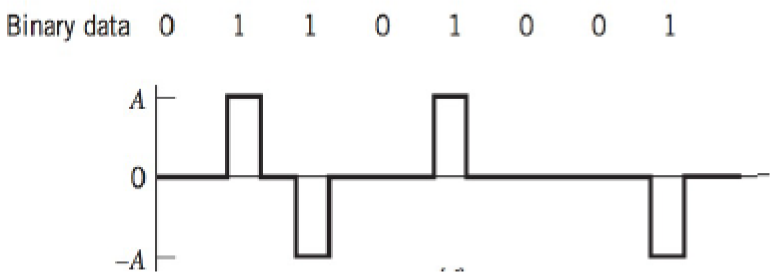
2. **Polar Nonreturn-to-Zero (NRZ) Signaling:** In this second line code, symbols 1 and 0 are represented by transmitting pulses of amplitudes $+A$ and $-A$, respectively. This line code is relatively easy to generate and is more power-efficient than its unipolar counterpart.



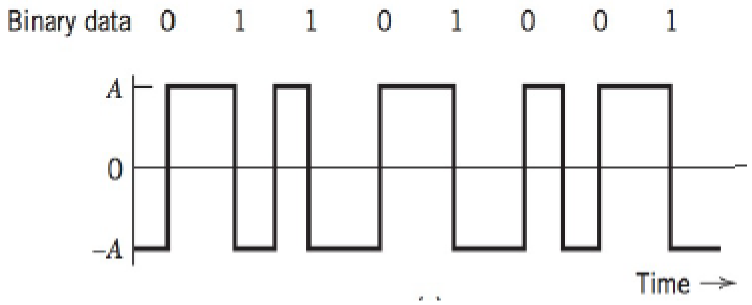
3. **Unipolar Return-to-Zero (RZ) Signaling:** In this other line code, symbol **1** is represented by a rectangular pulse of amplitude A and half-symbol width, and symbol 0 is represented by transmitting no pulse. An attractive feature of this line code is the presence of delta functions at $f = 0, \pm 1/T_b$ in the power spectrum of the transmitted signal, which can be used for bit-timing recovery at the receiver. However, its disadvantage is that it requires 3 dB more power than polar return-to-zero signaling for the same probability of symbol error.



4. **Bipolar Retum-to-Zero (BRZ) Signaling.** This line code uses three amplitude levels. Specifically, positive and negative pulses of equal amplitude (i.e., $+A$ and $-A$) are used alternately for symbol 1, with each pulse having a half-symbol width. No pulse is always used for symbol 0. A useful property of the BRZ signaling is that the power spectrum of the transmitted signal has no DC component and relatively insignificant low-frequency components for the case when symbols 1 and 0 occur with equal probability. This line code is also called *alternate mark inversion* (AMI) signaling.

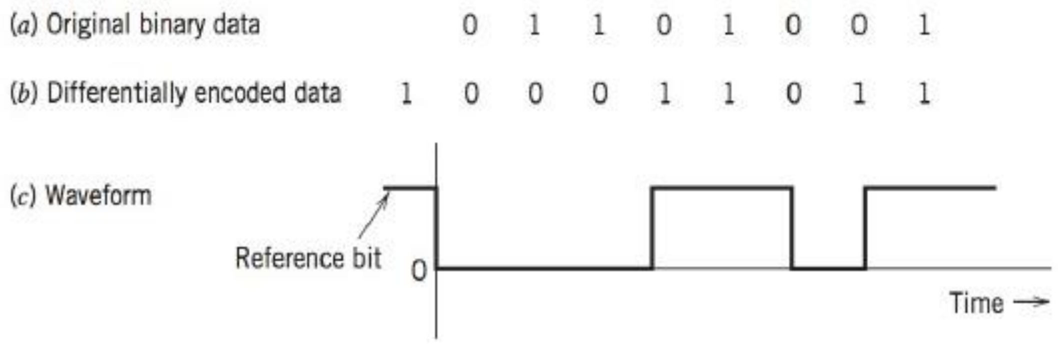


5. **Split-Phase (Manchester Code).** In this method of signaling, symbol 1 is represented by a positive pulse of amplitude A followed by a negative pulse of amplitude -A, with both pulses being a half-symbol wide. For symbol 0, the polarities of these two pulses are reversed. The Manchester code suppresses the DC component and has relatively insignificant low frequency components, regardless of the signal statistics. This property is essential in some applications.



DIFFERENTIAL ENCODING

This method is used to encode information in terms of signal transitions. In particular, a transition is used to designate symbol 0 in the incoming binary data stream, while no transition is used to designate symbol 1. The original binary information is recovered simply by comparing the polarity of adjacent binary symbols to establish whether or not a transition has occurred. Note that differential encoding requires the use of a reference bit before initiating the encoding process.



T1 Carrier System

In this example, we describe the important characteristics of a system known as the T1 carrier system, which is designed to accommodate 24 voice channels, primarily for short-distance, heavy usage in metropolitan areas.

The T1 system was pioneered by the Bell System in the United States in the early 1960s, and with its introduction the shift to digital communications facilities started. The T1 system was adopted for use throughout the United States, Canada, and Japan.

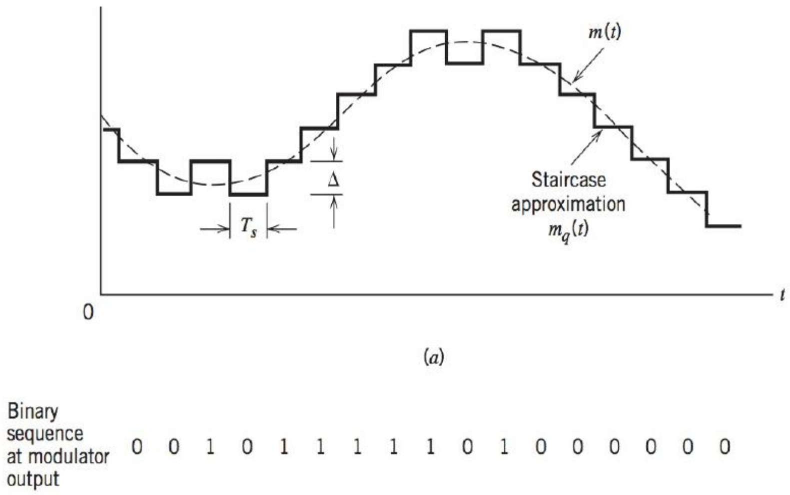
Bandwidth of voice signal	300 Hz to 3.1 kHz
W	3.1 kHz
Nyquist rate	6.2 kHz
Sampling rate	8 kHz
Companding	μ-law
μ	255
No of voice channels	24
Multiplexing	TDM
Binary code	8- bit word of PCM signal
Transmission rate	1.544 Mbps

The different voice channels are combined using a time-division multiplex (TDM) strategy. Each frame of the multiplexed signal consists of twenty-four 8-bit words, one for each voice source, plus a single bit that is added to the end of the frame for synchronization.

Hence, each frame consists of a total of (24×8) + 1 = 193 bits. With a sampling rate of 8 kHz for each voice channel, this implies that each frame has a period of 125 μs. Correspondingly, the duration of each bit is 0.647 μs and the resultant transmission rate is 1.544 megabits per second.

DELTA MODULATION (DM)

In *delta modulation* (DM), an incoming message signal is oversampled (i.e., at a rate much higher than the Nyquist rate) to purposely increase the correlation between adjacent samples of the signal. This is done to permit the use of a simple quantizing strategy for constructing the encoded signal.
In its basic form, DM provides a *staircase approximation* to the oversampled version of the message signal.



The difference between the input signal $m(t)$ and the approximation $m_q(t)$ is quantized into only two levels, namely, ±A, corresponding to positive and negative differences, respectively.

- If the approximation lies above the signal, it is diminished by A.
- If the approximation lies below the signal, it is enhanced by A.

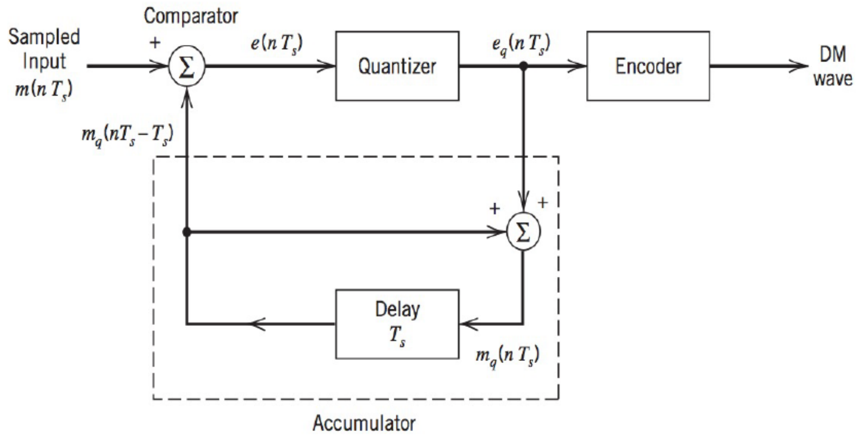
The basic principle of delta modulation may be formalized in the following set of discrete-time relations:

If T_s = the sampling period;

$e(nT_s)$ = an error signal representing the difference between the present sample value $m(nT_s)$ of the input signal and the latest approximation to it, that is, $m_q(nT_s - T_s)$; and

$e_q(nT_s)$ = the quantized version of $e(nT_s)$. The quantizer output $e_q(nT_s)$ is finally coded to produce the desired DM signal.

Analog to digital encoder:



The principal virtue of delta modulation is its simplicity. It may be generated by applying the sampled version of the incoming message signal to a digital encoder that involves a *comparator*, *quantizer*, and *accumulator* interconnected as shown in Figure. The comparator computes the difference between its two inputs. The quantizer consists of a *hard limiter* with an input-output relation that is a scaled version of the signum function.

$$e(nT_s) = m(nT_s) - m_q(nT_s - T_s)$$
$$e_q(nT_s) = m_q(nT_s) - m_q(nT_s - T_s)$$
$$m_q(nT_s) = m_q(nT_s - T_s) + e_q(nT_s)$$
$$e_q(nT_s) = \Delta\,\mathrm{sgn}\big(e(nT_s)\big)$$

- When $m(nT_s) > m_q(nT_s - T_s)$, $e(nT_s)$ is positive,

$$e_q(nT_s) = +\Delta$$

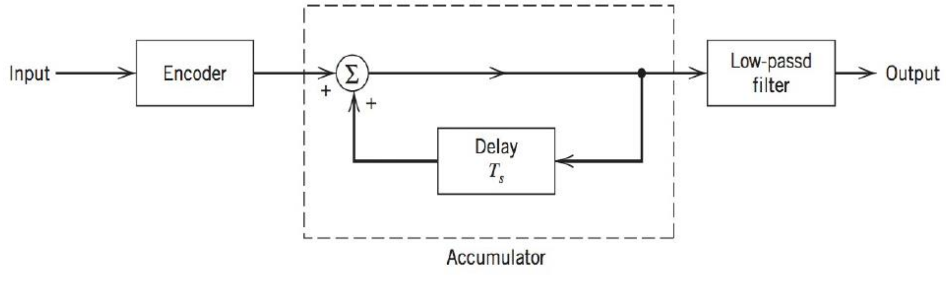
- When $m(nT_s) < m_q(nT_s - T_s)$, $e(nT_s)$ is negative,

$$e_q(nT_s) = -\Delta$$

The quantizer output is then applied to an accumulator, producing the result

$$m_q(nT_s) = \Delta \sum_{i=1}^{n} \mathrm{sgn}[e(iT_s)]$$
$$m_q(nT_s) = \sum_{i=1}^{n} e_q(iT_s)$$

Digital to analog decoder:

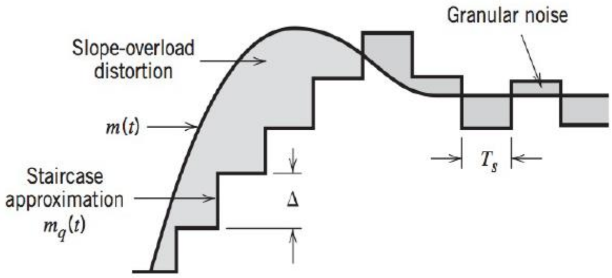


In the receiver, the staircase approximation $m_q(t)$ is reconstructed by passing the sequence of positive and negative pulses, produced at the decoder output, through an accumulator in a manner similar to that used in the transmitter. The out-of-band quantization noise in the high-frequency staircase waveform $m_q(t)$ is rejected by passing it through a low-pass filter with a bandwidth equal to the original message bandwidth.

Errors in DM:

Delta modulation is subject to two types of quantization error:

1. slope overload distortion and
2. granular noise.



Slope – overload distortion:

When the step-size Δ is too small for the staircase approximation $m_q(t)$ to follow a steep segment of the input waveform $m(t)$ with the result that $m_q(t)$ falls behind $m(t)$. This condition is called *slope overload,* and the resulting quantization error is called *slope-overload distortion* (noise).

The condition for no slope-overload error is

$$(Slope\ of\ stairecase\ approximation) \geq Maximum\ slope\ of\ m(t)$$

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

Granular error:

In contrast to slope-overload distortion, *granular noise* occurs when the step-size Δ is too large relative to the local slope characteristics of the input waveform $m(t)$, thereby causing the staircase approximation $m(t)$ to hunt around a relatively flat segment of the input waveform. Granular noise is analogous to quantization noise in a PCM system.

DIGITIZATION OF VIDEO AND MPEG

Video can be represented by three dimensions.

- Two dimensions are spatial and represent a still image,
- the third dimension is temporal and represents how the image evolves with time.

In practice, the still image is actually represented further by three dimensions referred to as the luminance (brightness) and two chrominance (color) components [similar to the three red-green-blue (RGB) components of an analog video signal]. The MPEG standard takes advantage of the high degree of spatial and temporal correlation that is expected in a video signal in order to reduce the number of bits required to represent the signal.

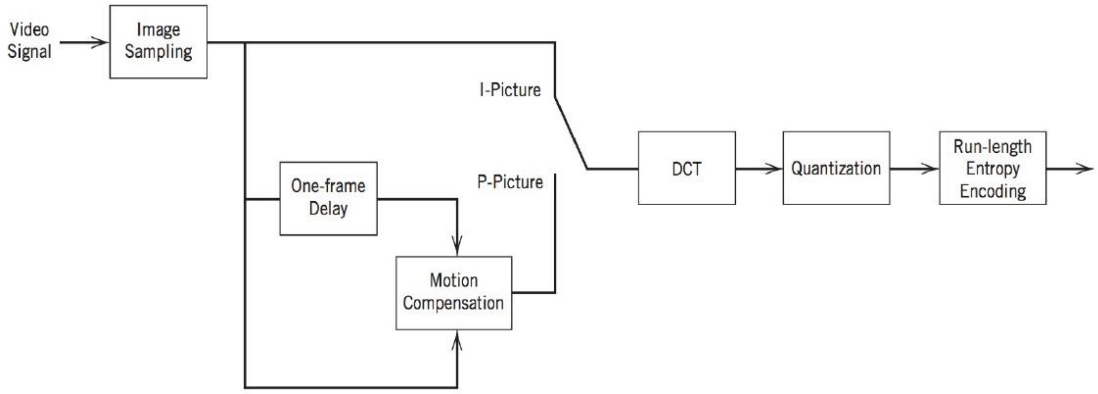


FIGURE Simplified block diagram of video signal processing

<u>The complex processing that occurs in the MPEG-1 video compression standard:</u>

- The first step is sampling the video signal. A sample of a video signal is an *N* X *M* matrix of picture elements corresponding to a complete still image. This matrix sample is referred to as a *video frame*. In fact, three such matrices must be obtained, one for each of the luminance and two chrominance components.
- The MPEG standard takes advantage of the fact that the human eye is less sensitive to changes in chrominance than in luminance and uses a lower frame rate for chrominance signals. Typical frame rates for luminance signals may range 15 to 60 per second and those for chrominance signals may be one-quarter of this value.
- At the receiver, a decoder uses interpolation to construct the missing chrominance samples and recreate the video signal.

The MPEG-encoding algorithm encodes the first frame in a video sequence in the *intraframe coding mode* (I-picture). Each subsequent frame is coded using *interframe prediction* (P-picture)— where data from the previously coded I- or P-frame are used for prediction.

For the first frame (I-picture), the *discrete cosine transform* (DCT) is applied to each 8 X 8 block of pels. Mathematically, this two-dimensional transform is defined by

$$X(k_1,k_2)=\sum_{m_1=0}^{M_1-1}\sum_{m_2=0}^{M_2-1}x(m_1,m_2)\cos\left(\frac{k_1\pi}{M_1}(m_1+0.5)\right)\cos\left(\frac{k_2\pi}{M_2}(m_2+0.5)\right)$$

Where (m_1, m_2) are the coordinates in the spatial domain, and (k_1, k_2) are the coordinates in the transform domain.

For an 8 X 8 transform, M_1 = M_2 = 8 and, k_1 & k_2 range from 0 to 7.

Due to the high spatial correlation, only a small number of the DCT coefficients are significant, typically those near the (0, 0) element. The coefficients for each DCT block are then quantized, and then only the resulting non-zero coefficients (and their position) are transmitted for each 8 X 8 block of the first frame. The method for encoding the quantized coefficients and their position uses an advanced data-compression technique known as *run-length entropy encoding*.

For coding subsequent P-pictures, the previous I- or P-picture frame is stored. The first step in this process is to identify correlation in the time domain. Correlation in the time domain corresponds to motion of a pel or a block of pels in the same direction. To identify this motion, the picture is broken down into 16 X 16 pel *macroblocks*. The stored image and the new image are correlated and a motion vector is identified for each 16 X 16 macroblock. A motion-compensated prediction error is calculated by subtracting each pel in a macroblock from its motion-shifted counterpart in the previous frame. The processing of the prediction errors then follows the same steps as the first picture. A DCT of each 8 X 8 block is performed, and the results are quantized. Since the temporal correlation is expected to be high, the prediction errors are small and most of the quantized DCT coefficients are zero. For the P-pictures, only the motion vectors for each macroblock and the small number of non-zero DCT coefficients have to be transmitted. The transmitted data for P-pictures is typically much less than for I-pictures.