Voice Transmission
--Basic Concepts--

❑ **Voice---is analog in character and moves in the form of waves.**

❑ **<u>3-important wave-characteristics</u>:**

  ❑ Amplitude
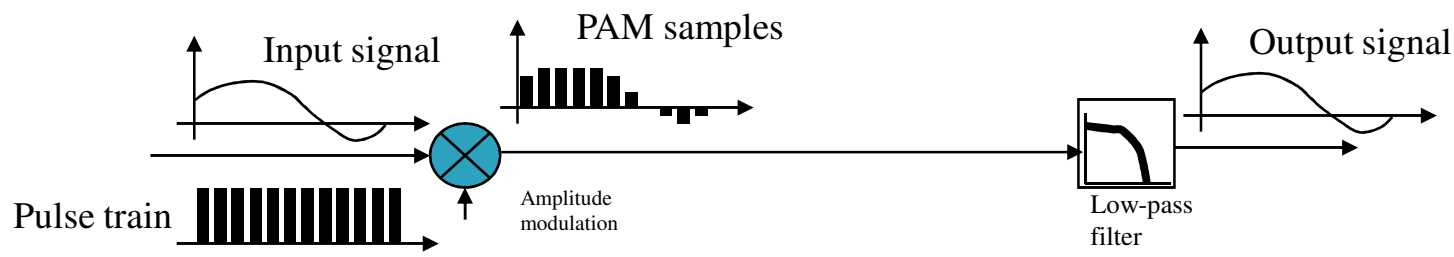
  ❑ Frequency

  ❑ Phase

Voice Digitization in the POTS

❑ Traditional POTS was analog.

❑ The current telephone system of the POTS combines both analog and digital transmission technologies.

❑ **Why Voice digitization:**

❑ Ensures better quality (than analog)

❑ Provides higher capacity (than analog)

❑ Deals with longer distance (than analog)

Voice Digitization—How!

❑ Digitization is just a discrete electrical voltage.

❑ Electrical pulses can be varied to represent characteristics of an analog voice signal.

❑ **<u>5-different VD-techniques:</u>**

 ❑ **PAM = pulse amplitude modulation**

 ❑ **PDM = pulse duration modulation**

 ❑ **PPM = Pulse position modulation**

 ❑ **PCM = Pulse code modulation**
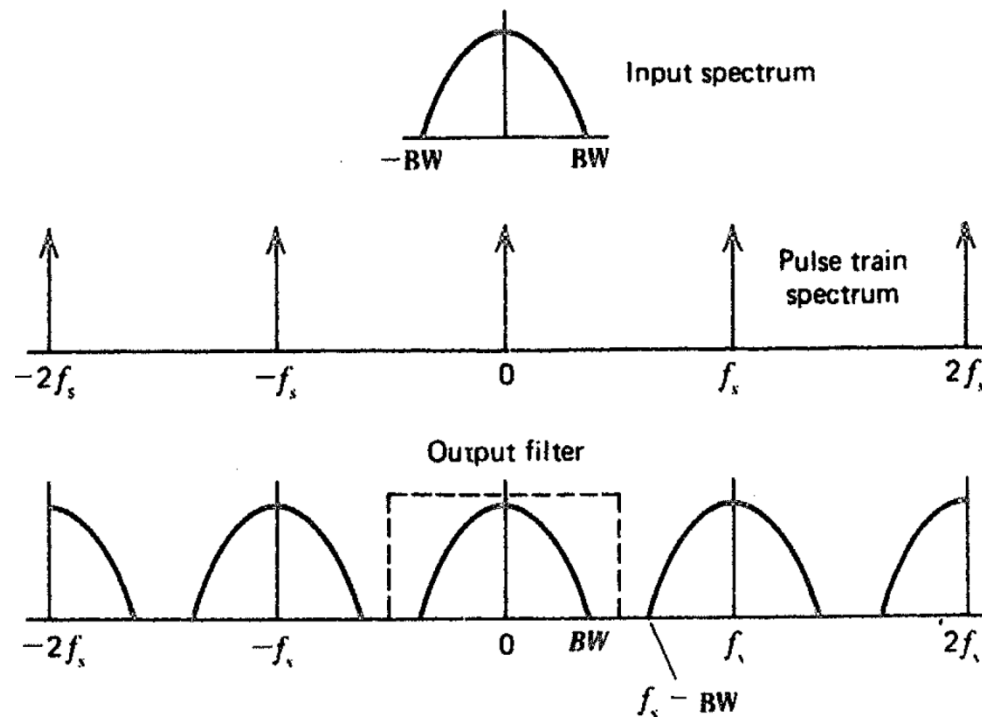
 ❑ **ADPCM = Adaptive differential PCM**

Pulse Amplitude Modulation

❑ First step in digitizing an analog waveform

❑ Establishes a set of discrete times at which the input signal waveform is sampled.

❑ The sampling process is equivalent to amplitude modulation of a constant amplitude pulse train, thus, PAM.

❑ Nyquist Sampling Rate : The minimum sampling frequency required to extract all information in a continuous, time-varying waveform.

   ❑ Nyquist Criterion: $f_s > 2 \times BW$

   ❑ f_s: sampling rate, BW: bandwidth of the input signal

Input signal   PAM samples   Output signal

Pulse train   Amplitude modulation   Low-pass filter

Spectrum of PAM Signal

❑ The PAM spectrum can be derived by observing that a continuous train of impulses has a frequency spectrum consisting of discrete terms at multiples of the sampling frequency.

Input spectrum

$-BW$ BW

Pulse train spectrum

$-2f_s$ $-f_s$ 0 f_s $2f_s$

Output filter

$-2f_s$ $-f_s$ 0 BW f_s $2f_s$

$f_s - BW$

Source: Digital Telephony by Bellamy

Foldover Distortion

❑ If the input is undersampled ($f_s<2BW$), the original waveform cannot be recovered without distortion

❑ Another term for this impairment is "aliasing". Remember the old Westerns in which the wheels of stagecoaches appear to move backward.
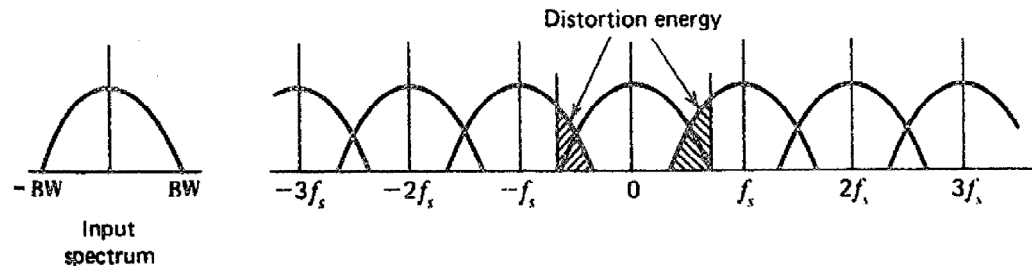
Aliasing

❑ The example below demonstrates an aliasing process occurring in speech if a 5.5 kHz signal is sampled at an 8-KHz rate. Notice that the sample values are identical to those obtained from a 2.5 KHz signal.

❑ Thus, after the sampled signal passes through the 4 KHz output filter, a 2.5 KHz signal arises that did not come from the source.
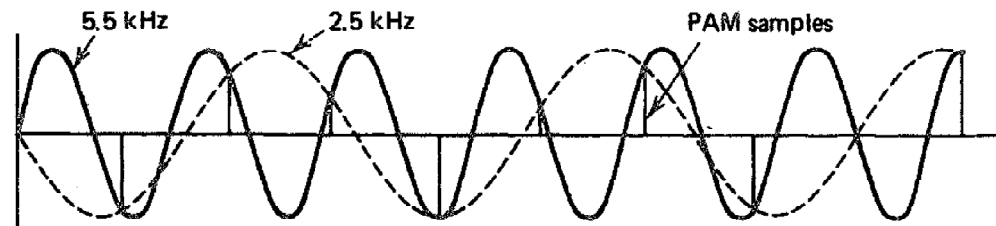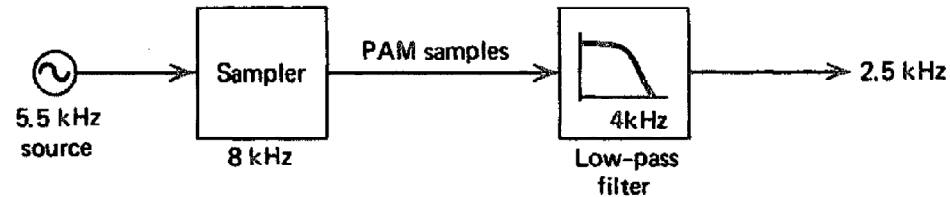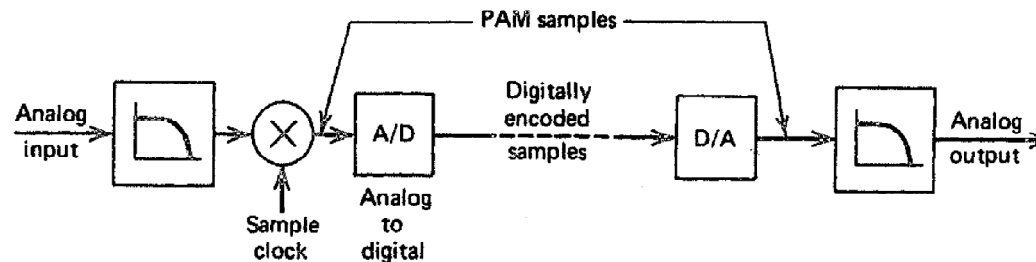
PAM System

❑ Complete PAM system includes a bandlimiting filter before sampling to ensure that no source related signals get folded back into the desired signal bandwidth.

❑ End-to-End PAM system
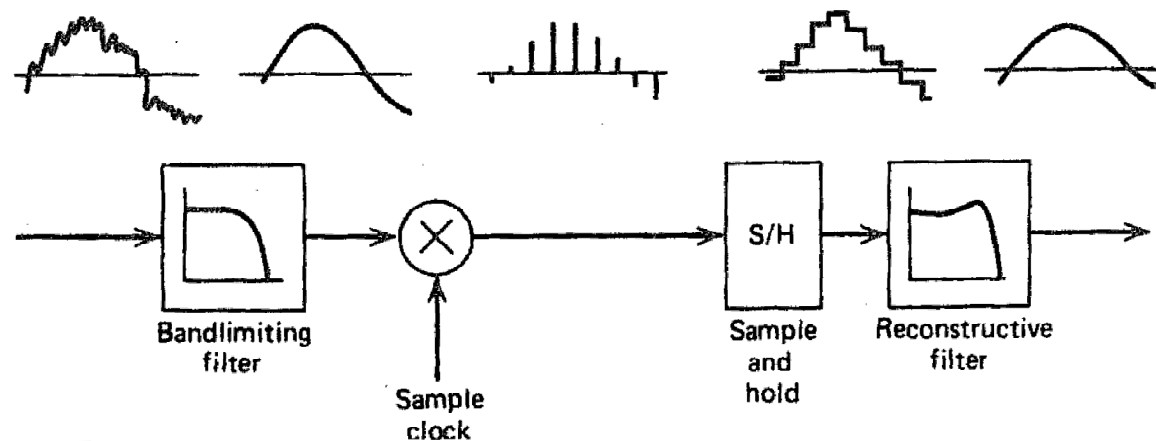
Pulse Coded Modulation (PCM)

❑ PCM is an extension of PAM wherein each analog sample is quantized into a discrete value for representation as a digital codeword

❑ PAM system can be converted to PCM if we add ADC at the source and DAC at the destination.

Quantization

❑ Quantization process has a set of quantization intervals associated in a one-to-one fashion with a binary codeword.

   ❑ Binary codeword corresponds to a discrete amplitude

❑ Quantization process introduces a certain amount of error or distortion into the signal samples.
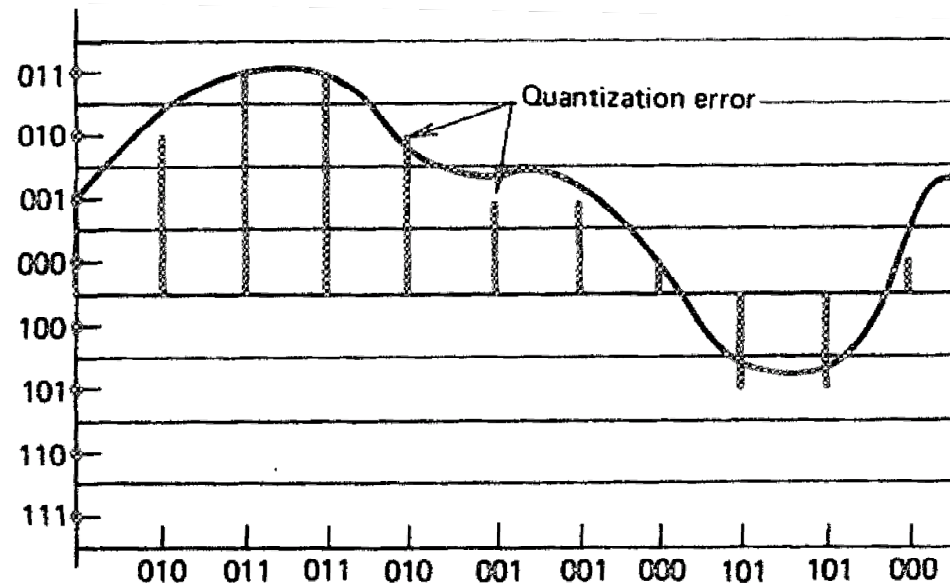
Quantization Noise

❑ Quantization errors of a PCM encoder are generally assumed to be distributed randomly and uncorrelated to each other

❑ Treated as an additive noise

❑ Signal-to-quantizing-noise ratio (SQR) to define amount of quantization distortion.
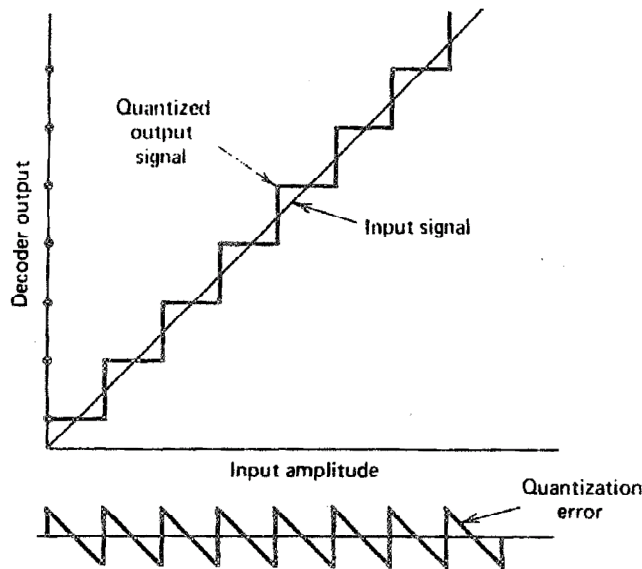
$$SQR = \frac{E\{x^2(t)\}}{E\{[y(t) - x(t)]^2\}}$$

where

$E\{.\} =$ expectation or averaging

$x(t) =$ analog input signal

$y(t) =$ decoded output signal

Quantization Noise

❑ The error e(t)=y(t)-x(t) is limited in amplitude to q/2, where q is the height of the quantization interval.

❑ A sample value is equally likely to fall anywhere within a quantization interval implying a uniform probability density of amplitude 1/q.

$$p(\eta) = \begin{cases} \frac{1}{q} & -\frac{2}{2} \leq \eta \leq \frac{1}{q} \\ 0 & \text{otherwise} \end{cases}$$

The average or expected value of noise power is determined as

$$\text{Quantization noise power} = \int_{-q/2}^{q/2} \left(\frac{1}{q}\right) \eta^2 d\eta = \left(\frac{1}{12}\right) q^2$$

$$\text{SQR(dB)} = 10 \log_{10} \left(\frac{v^2}{q^2/12}\right) = 10.8 + 20 \log_{10}\left(\frac{v}{q}\right) \quad v \rightarrow \text{rms amplitude}$$

$$\text{SQR(dB)} = 10 \log_{10} \left(\frac{A^2/2}{q^2/12}\right) = 7.78 + 20 \log_{10}\left(\frac{A}{q}\right) \quad A \rightarrow \text{sinewave amplitude}$$

Idle Channel Noise

❑ The noise may actually be greater than the signal when sample values are in the first quantization interval.

    ❑ Bothersome during speech pauses

    ❑ Midriser Quantizer: It cannot generate constant zero output level

    ❑ Midtread Quantizer:It straddles the origin to generate zero output
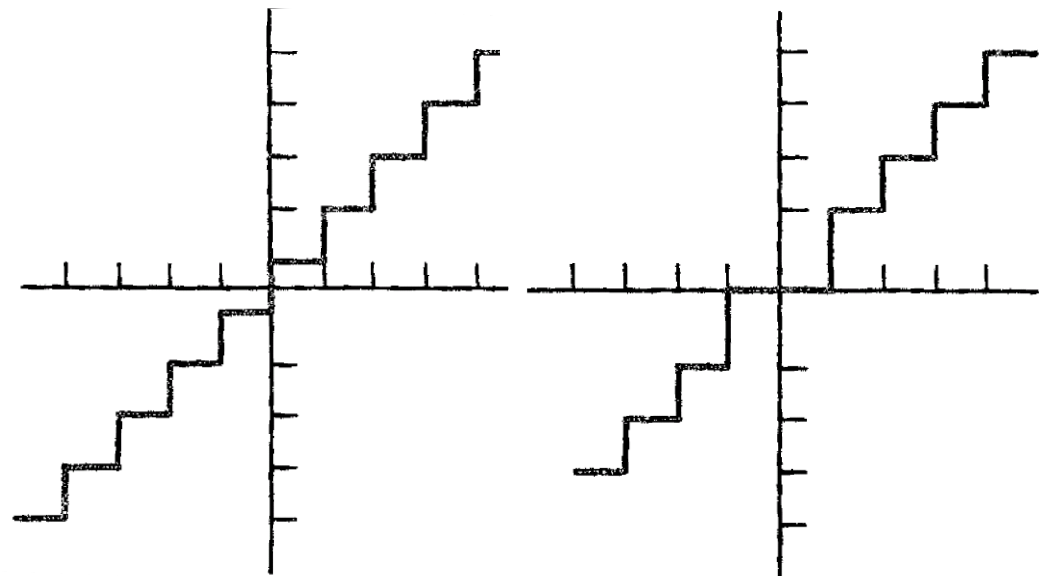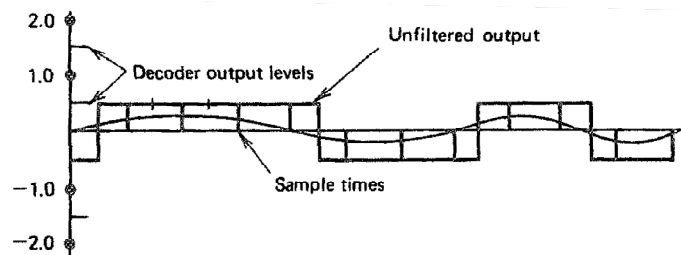

Midriser   Midtread

Uniformly Encoded PCM

❑ Numerical value of each codeword is proportional to the quantized amplitude that it represents

❑ Minimum digitized voice quality requires 26 dB SQR to provide adequate quality for small signals

   ❑ q_{max}=0.123A, and encoding from –A to A, we need 4 bits

❑ Dynamic Range (DR): Capability of transmitting a large range of signal amplitudes

   ❑ Expressed in dB as the ratio of the maximum amplitude signal to the minimum amplitude signal

$$DR = 10 \log_{10}\left(\frac{P_{max}}{P_{min}}\right) = 20 \log_{10}\left(\frac{V_{max}}{V_{min}}\right)$$

   ❑ PCM performance equation for uniform coding

$$SQR = 1.76 + 6.02n + 20 \log_{10}\left(\frac{A}{A_{max}}\right)$$
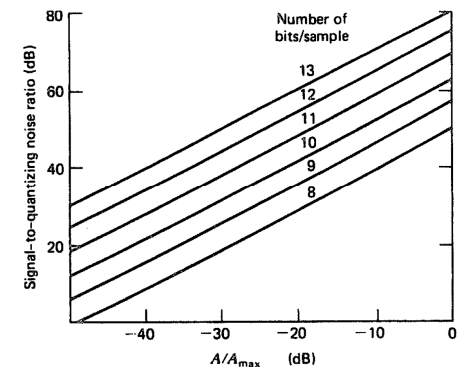
Companding

- ❑ SQR provides the quantization interval for small signals.
- ❑ Large signals are also encoded with the same quantization interval
  - ❑ 26 dB SQR (small signals)+30 dB (DR)=56 dB SQR for large signals
  - ❑ Uniform PCM provides unneeded quality for large signals
  - ❑ Large signals least likely to occur
- ❑ Companding = Compression + Expansion
  - ❑ Compression: More efficient coding procedure is achieved if the quantization intervals are NOT uniform but allowed to increase with sample value
  - ❑ Relationship becomes nonlinear between codeword and sample value
  - ❑ Expansion: Inverse compression is needed at the receiver

Compression

❑ Various compression-expansion characteristics can be chosen to implement a compandor
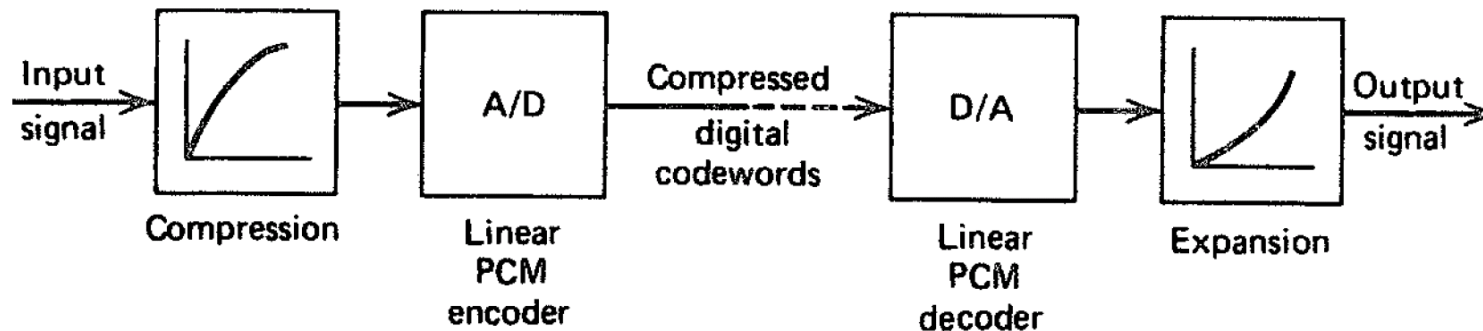
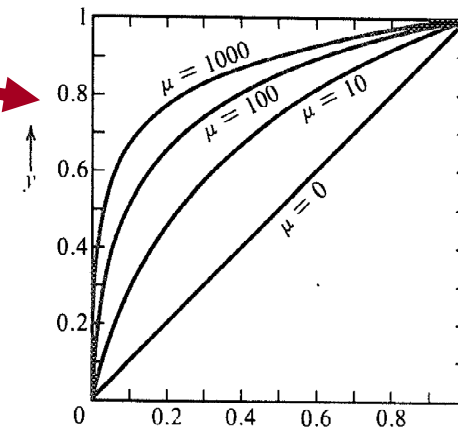❑ Compression used in North America has the μ-law characteristics

$$F_\mu(x) = \text{sgn}(x)\frac{\log(1 + \mu|x|)}{\log(1 + \mu)}$$ ⟵ compressor ⟶

μ defines the amount of compression
$\text{sgn}(x)$ is the polarity of x
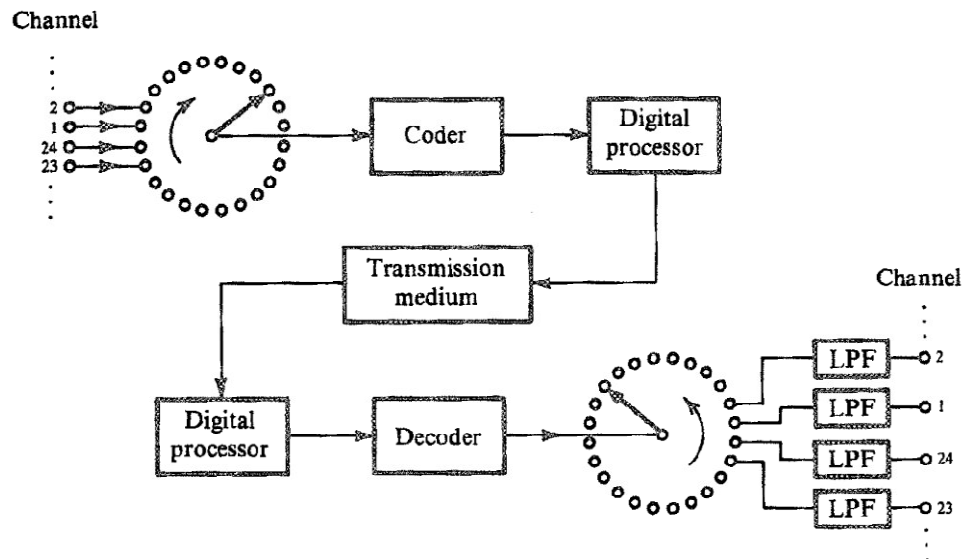x is the normalized input signal amplitude

$$F_\mu^{-1}(y) = \text{sgn}(y)\left(\frac{1}{\mu}\right)[(1 + \mu)^{|y|} - 1]$$ ⟵ expander

T1 Carrier System

❑ Telephone wires were used to transmit one audio signal of bandwidth 4 KHz

❑ After introduction of PCM, the same wires are used to transmit 24 TDM PCM (DS0) telephone signals wit a total BW of 1.544 MHz

❑ Repeaters every at 6000 feet
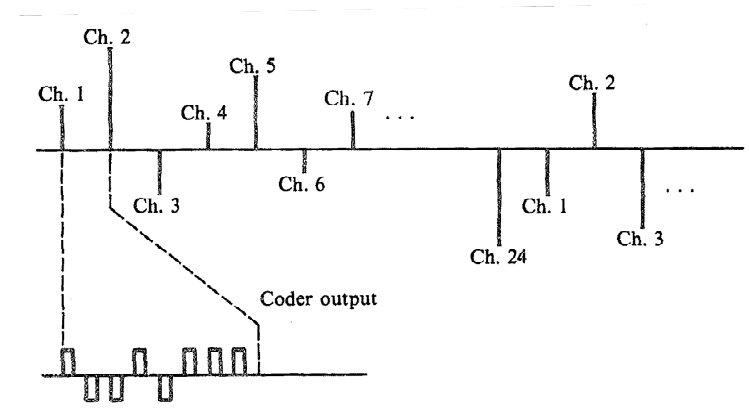
T1 System Signaling Format

❑ Binary code words corresponding to samples of each of the 24 channels are multiplexed in a sequence

  ❑ $\mu=100$, 7 bits are used for data, 1-bit for signaling in D1 signaling

❑ D2 channel bank needs better voice quality

  ❑ 8 bits PCM and $\mu=255$, signaling only repeats every sixth frame
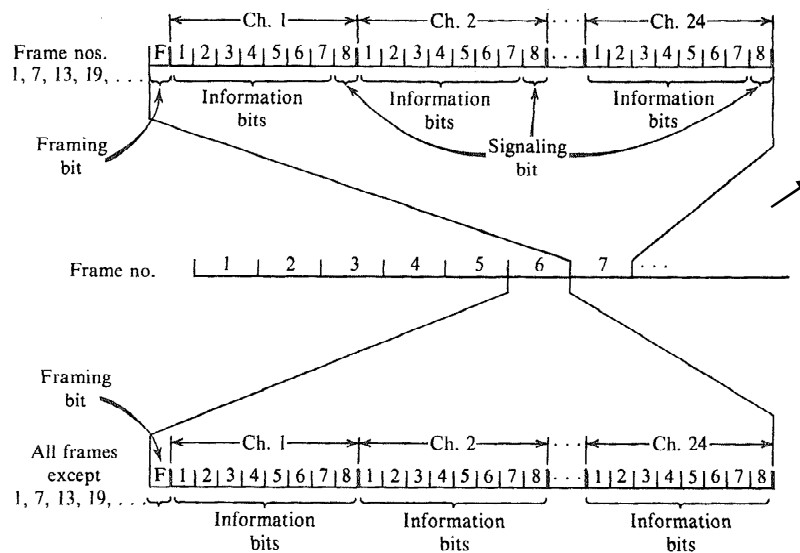
Robbed bit signaling

DS-0   provides 64 Kbps

T-1	DS-1	provides	1.544	Mbps
T-2	DS-2	provides	6.312	Mbps
T-3	DS-3	provides	33.375	Mbps
T-4	DS-4	provides	274.176	Mbps

Voice Compression

❑ How does Vonage system work?

❑ PCM system is inherently capable of encoding an arbitrarily random waveform as long as the maximum-frequency component does not exceed one-half the sampling rate.

❑ Speech signals have considerable redundancy from one sample to the next.

  ❑ Highly correlated from one sample-to-next, about 85%

  ❑ Ex: Encode only derivative of the signal

❑ Significant savings in bandwidth are possible through more efficient coding

What is a Codec!

- **Codec** = **Co**der + **Dec**oder
- **Modem** = **Mo**dulator + **Dem**odulator
- Codec is a voice-connector device that samples the analog POTS transmission into a stream of binary digits using PCM.
- For quality digital transmission over a Codec:
  - 8000 samples/sec
  - Each sample = 1 byte = 8 bits
  - 8000 X 8 = 64,000 bits/sec----(DS-0 circuit)
  - Uses T-1 circuit—1.544 Mbps---(equivalent to 24 DS-0)

Codec and Modem!

❑ Codec is a voice-connector device

❑ Modem is a data-connector device.

❑ All high speed Codecs operate at multiples of   DS-0 speed:

  ❑ DS-0   provides 64   Kbps
  ❑ T-1   DS-1   provides 1.544   Mbps
  ❑ T-2   DS-2   provides 6.312   Mbps
  ❑ T-3   DS-3   provides 33.375   Mbps
  ❑ T-4   DS-4   provides 274.176 Mbps

Analog Signal to be Digitized

8 possible amplitudes in example are really 256 (2⁸) amplitudes in PCM

1/8000 of a second   Sampling rate = 8,000 times/second
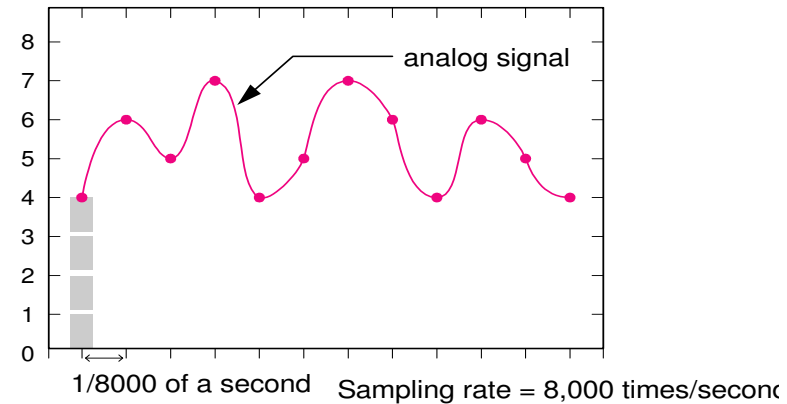
Step 1: Sample Amplitude of Analog Signal

Amplitude in example at sample position 1 (the gray shaded box) = 4

Step 2: Represent Measured Amplitude in Binary Notation

Power of 2	2^7	2^6	2^5	2^4	2^3	2^2	2^1	2^0		
Value	128	64	32	16	8	4	2	1		
Binary notation	0	0	0	0	0	1	0	0	=	4

8 bits = 1 byte

Step 3: Transmit Coded Digital Pulses Representing Measured Amplitude
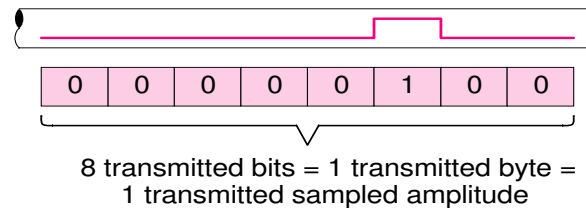
Voice Digitization: Pulse Code Modulation (PCM)

0	0	0	0	0	1	0	0

8 transmitted bits = 1 transmitted byte = 1 transmitted sampled amplitude

Voice Compression!

❑ **ADPCM**--adaptive differential PCM-is also known as a voice compression technique, because of its ability to transmit 24 digitized voice conversations in half the bandwidth required by the PCM.

❑ More advanced **VC**s use—*digital signal processors* that take the digitized PCM and further manipulate and compress it.

❑ Each voice compression technique seeks to reduce the amount of transmitted voice-info in one way or another.

Differential PCM

❑ Differential PCM is designed specifically to exploit the sample-to-sample redundancies in a typical speech waveform.

❑ The range of sample differences is less than the range of individual samples, fewer bits are needed.

❑ DPCM is a first order prediction → How do you predict the future sample?
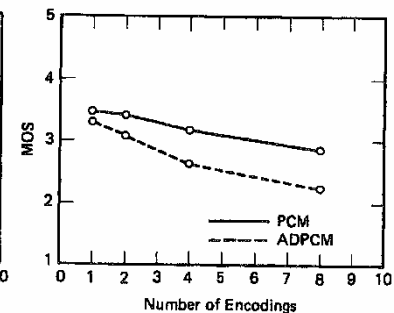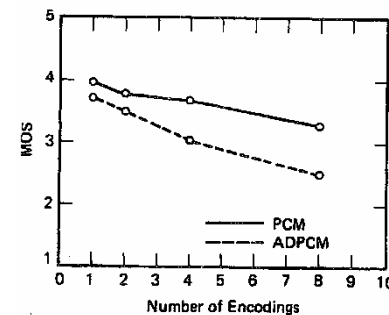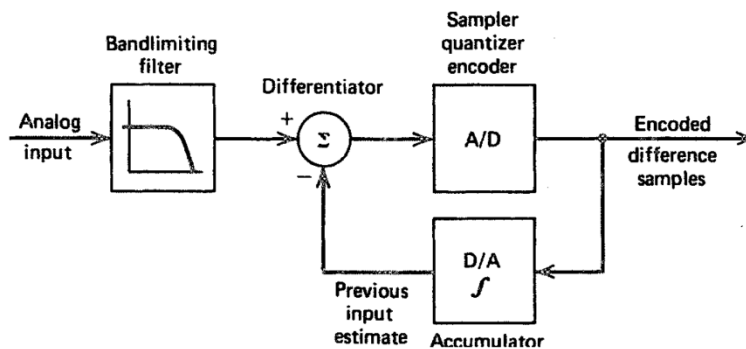
Example

❑ Speech digitization techniques are sometimes measured for quality by use of an 800-Hz sine wave as a representative test signal. Assuming a uniform PCM system is available to encode the sine wave across a given dynamic range, determine how many bits sample can be saved by using a uniform DPCM system.

❑ A basic solution can be obtained by determining how much smaller the dynamic range of the difference signal is in comparison to the dynamic range of the signal amplitude. Assume the maximum amplitude of the sine wave is A, so that

$$x(t) = A\sin(2\pi 800 t)$$

   ❑ The maximum amplitude of the difference signal can be obtained by differentiating and multiplying by the time interval between samples:

$$\frac{dx}{dt} = A(2\pi)(800)\cos(2\pi 800 t) \qquad |\triangle x(t)|_{\text{max}} = A(2\pi)(800)\left(\frac{1}{8000}\right) = 0.628A$$

   ❑ The savings in bits per sample can be determined as

$$\log_2\left(\frac{1}{0.628}\right) = 0.67\text{bits} \quad \longrightarrow \quad \frac{2}{3}\text{ bits less than a regular PCM}$$

Adaptive Differential PCM

❑ Adaptation logic can be added to the basic DPCM algorithm to create what is referred to as adaptive differential PCM (ADPCM).

❑ ITU has established a 32-kbps ADPCM standard (G.721) for voice messaging and for increasing the number of voice channels on a T1 line.

❑ It uses a 8th order predictor, adaptive quantization, and adaptive prediction.
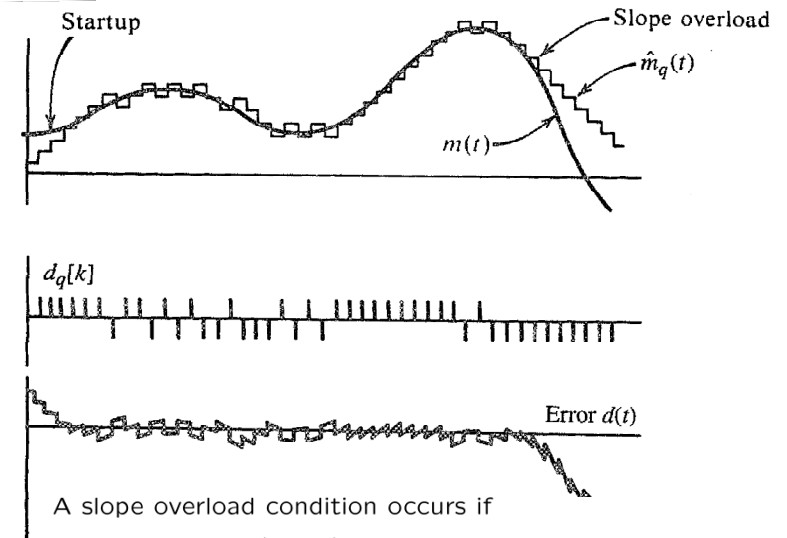
Delta Modulation

- ❑ Special case of DPCM with 1-bit per sample difference signal
- ❑ Requires high-sampling rate than DPCM
- ❑ Problem: Slope overload with sampling rate is low
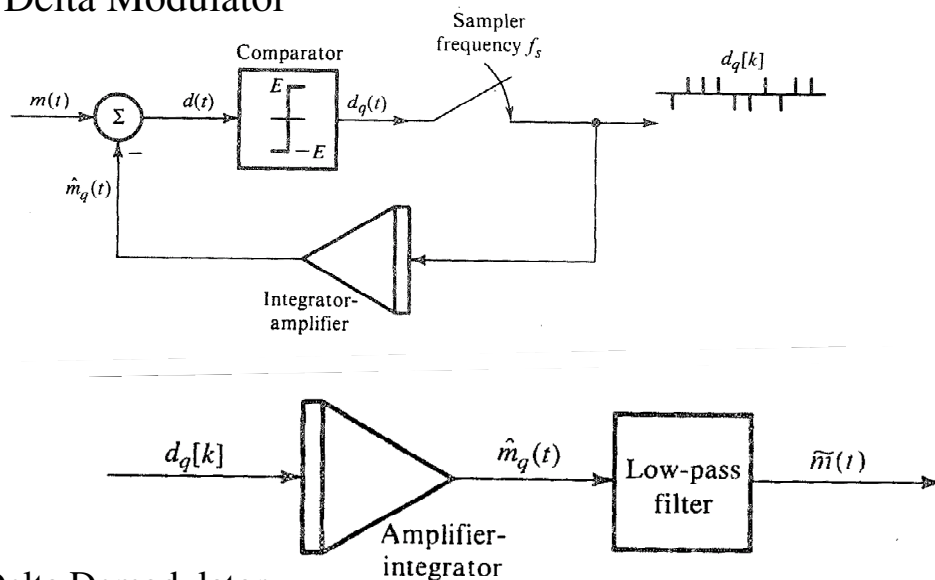
A slope overload condition occurs if

$$\left|\frac{dx(t)}{dt}\right| > q f_s$$

where $x(t)$=input signal, q = step size, and f_s=sampling frequency.

Subband Coding

❑ Human ear perceives speech by measuring the short-term energy level of individual frequency bands