→ Received signal is delayed by one bit time then compared with the next signalling element in the balanced modulator. If they are same, logic 1 is generated.
" " " different, logic 0 " .

UNIT - III - DIGITAL TRANSMISSION

Digital transmission is the transmittal of digital pulses between two points in a communication systems.

Advantages:

Noise Immunity, Multiplexing, more resistant to analog system, Simpler to measure & evaluate & Transmission errors can be easily detected & corrected.

Disadvantages:

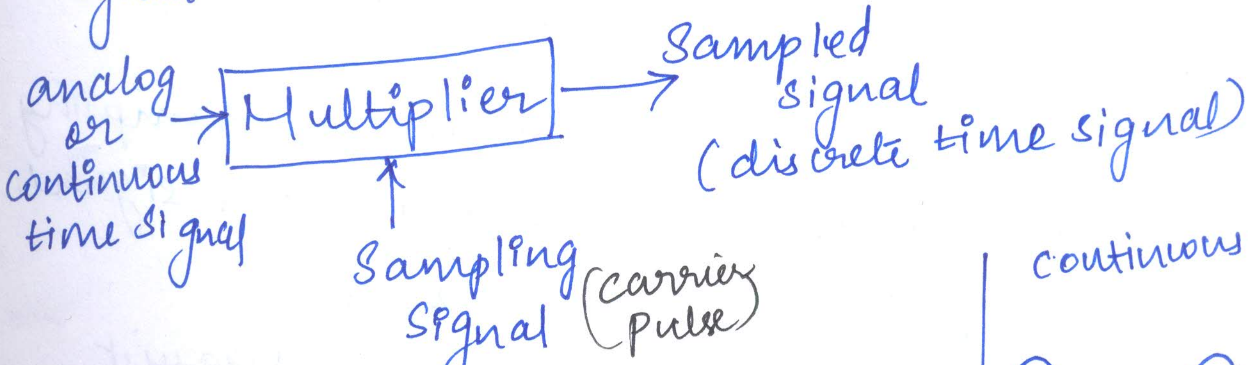
- Require more bandwidth
- Require Precise time Synchronization
- These are incompatible with existing analog facilities.

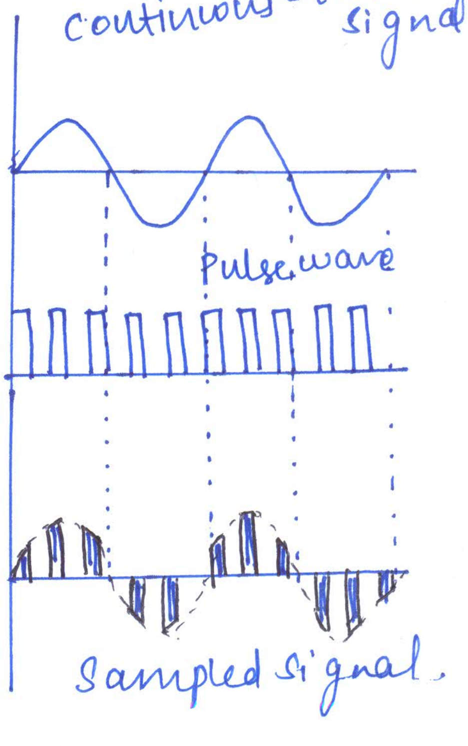
Pulse modulation :-

* Converting information into pulse form.
* In Pulse modulation, the carrier is a pulse train, its parameters are varied inaccordance with the instantaneous value of the modulating signal.

SAMPLING THEOREM:

SAMPLING is the process by which an analog signal is converted into corresponding sequence of samples that are spaced uniformly in time. It is necessary to choose the sampling rate properly, so that the sequence of samples uniquely defines or recovers the original analog signal.



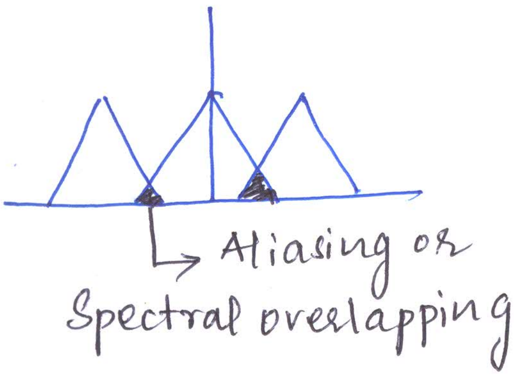


A continuous time signal can be represented in its sampled form and recovered back from the sampled form if the sampling frequency $f_s \geq 2f_m$, f_m is frequency of continuous time signal.

Natural Sampling: The top pulses of the sampled signals are not flat but they follow the natural waveform of the input signal during respective pulse intervals.

Flat top sampling: The top pulses of the sampled signals are flat, thus the pulses have constant amplitude within the pulse interval.

Aliasing:



Aliasing refers to the phenomenon of high frequency components in the spectrum of the signal appears to overlap with the lower frequency spectrum of its sampled version. ($f_s < 2f_m$)

Nyquist Rate: It is the minimum sampling rate required to represent the continuous signal in its sampled form. $f_s = 2f_m$.

Also called as Nyquist sampling rate.

Nyquist Interval $\rightarrow T_s = \frac{1}{f_s}$.

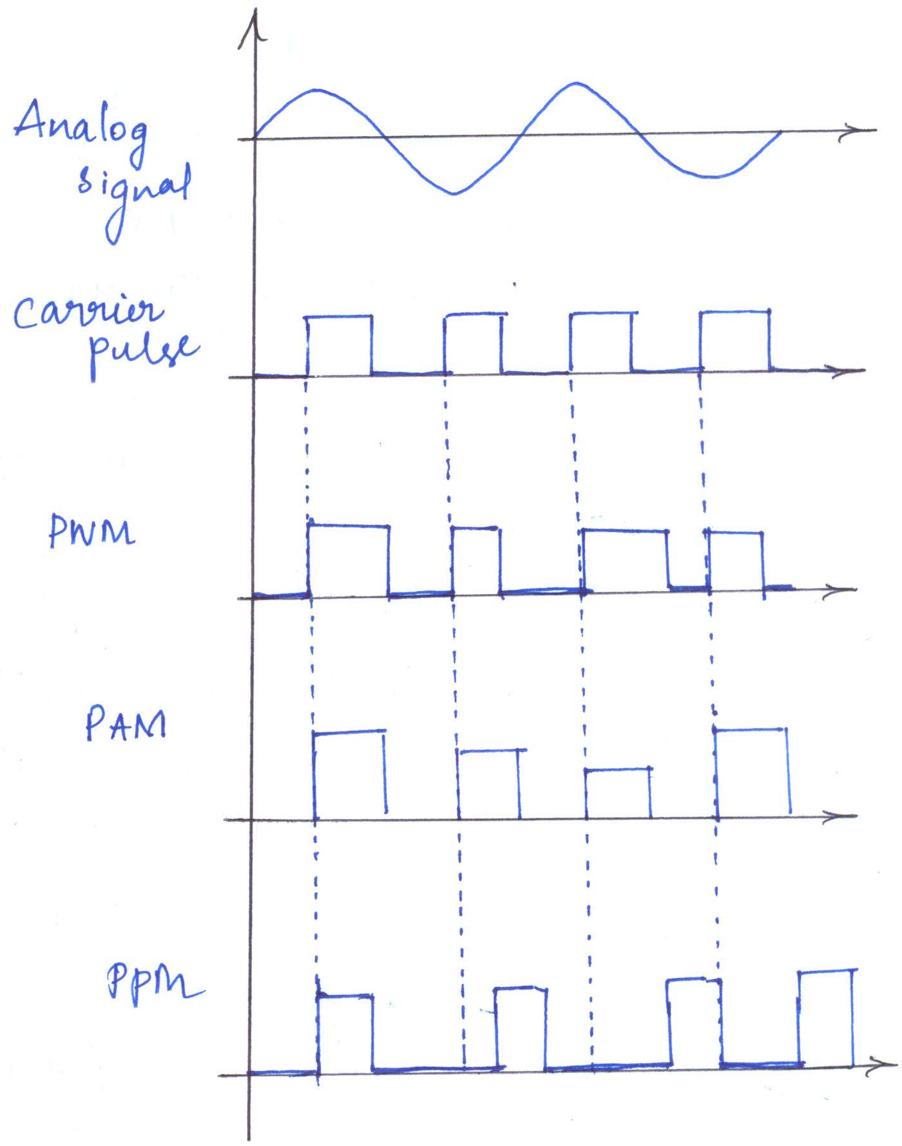
Types of Pulse Modulation: / Methods of Pulse modulati

1) Pulse Amplitude Modulation

PAM: The amplitude of constant width, constant position pulse is varied according to the amplitude of the analog signal.

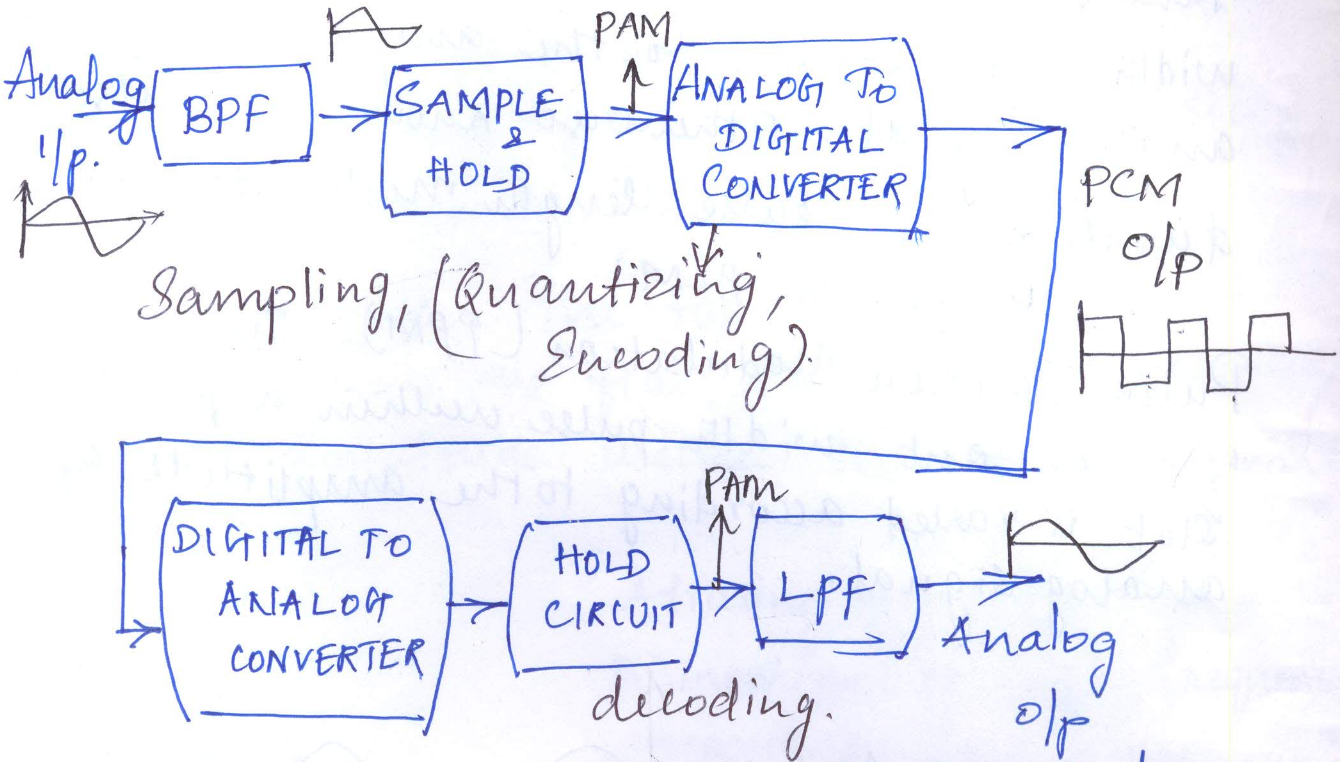
Pulse Width Modulation (PWM): The pulse width is proportional to the amplitude of the analog signal. Otherwise known as Pulse duration or pulse length modulation.
ie PDM or PLM.

Pulse Position Modulation (PPM): The position of a constant-width pulse within a prescribed slot is varied according to the amplitude of analog signal.



Transmit the information in the form of code words.

PULSE CODE MODULATION (PCM)



* It is the only one of the digitally encoded pulse modulation technique.
* Pulses are of fixed length and amplitude.
* BPF limits the i/p analog signal to the standard voice band frequency range 300 to 3000 Hz.
* Sample and Hold: It samples the analog input and converts PAM samples to a serial binary data stream for transmission.

  Purpose of sample & hold ÷ To sample periodically the continually changing analog i/p signal & convert the samples to a series of constant amplitude PAM levels.
* Media : Wire or optical fibre.

DAC - Converts serial binary stream to multilevel PAM signal.

Hold circuit and LPF: Converts PAM to its original form.

QUANTIZATION NOISE & Signal to Noise Ratio (SNR) IN PCM

Quantizer: (After sample & hold, quantizer is placed)

It converts the I/P sampled signal into an approximate quantized signal which consists of only a finite number of predecided voltage levels. The standard predecided levels are known as quantization levels.

$$x(t) \rightarrow \boxed{\text{Quantizer}} \rightarrow x_q(t) \qquad \text{Error } (\mathcal{E}) = x_q(t) - x(t)$$

$x(t)$ – ~~sampled~~ analog signal
$x_q(t)$ – quantized o/p signal.

The input signal is having peak to peak voltage of V_{high} & V_{low}. The entire voltage range has divided into 'Q' equal intervals each of size 's'. S - step size.

$$S = \frac{V_H - V_L}{Q}.$$

At the centre of these ranges, quantization levels are placed, then the number of quantization levels called as decision thresholds

Relation between number of quantization levels 'Q' and number of bits per word 'N' can be given as,

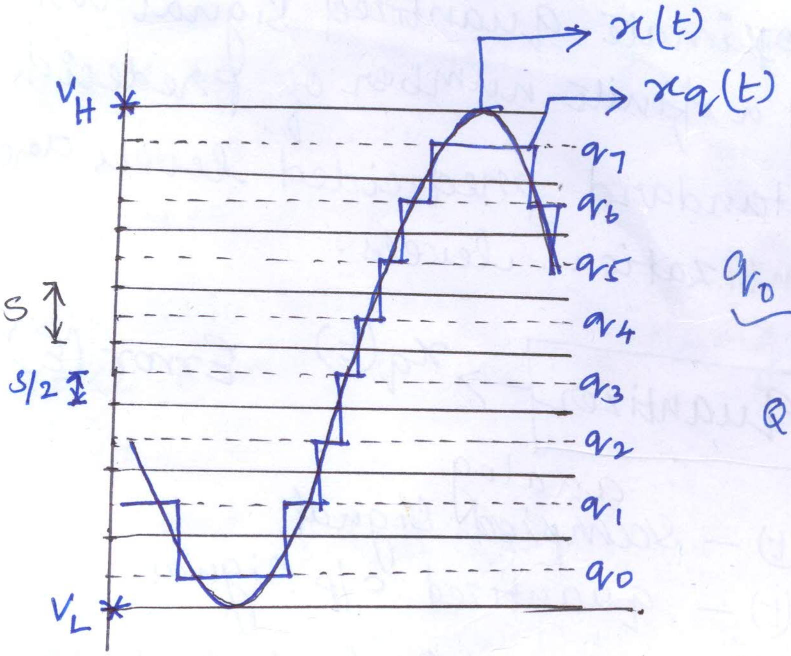
$$\boxed{Q = 2^N}$$

Q → Number of quantization levels

N → Number of combinations of bits/word.

To minimize the quantization error, reduce the step size by increasing number of quantization levels.

Maximum value of quantization error is $\pm S/2$.



$\underbrace{q_0 \text{ to } q_7}$ → Quantization levels.

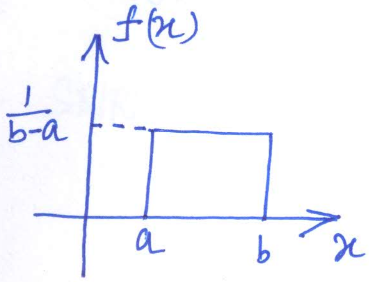
* Mean Square value of the quantization is given by $S^2/12$.

* $S = \dfrac{V_H - V_L}{Q}$ consider $V_H = V_{max}$, $V_L = -V_{max}$

$$S = \frac{2V_{max}}{Q}.$$

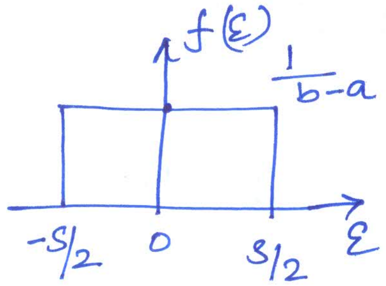
Error = $x_q(t) - x(t) = S/2$.

Uniform distribution :



$$f(x) = 0, \quad x < a$$
$$= \frac{1}{b-a}, \quad a \le x \le b$$
$$= 0, \quad x > b$$

Probability density function for quantization error be,



$$\frac{1}{\frac{S}{2} - \left(-\frac{S}{2}\right)}$$
$$= \frac{1}{S}$$

$$f(\varepsilon) = 0, \quad \varepsilon < -S/2$$
$$= \frac{1}{S}, \quad -\frac{S}{2} \le \varepsilon \le S/2$$
$$= 0, \quad \varepsilon > S/2.$$

Signal to quantization Noise Ratio (SNR):

$$SNR = \left[\frac{\text{Signal power}}{\text{Noise Power}}\right]_{\text{Normalised}}.$$

$$\text{signal power} = \frac{V^2}{R} = \frac{(V/\sqrt{2})^2}{R}$$
$$= \frac{V^2}{2R}$$
$$R = 1 \Rightarrow \frac{V^2}{2}$$

Noise power :

Noise is defined by the random variable(ε) and its pdc $f(\varepsilon)$.

Mean square value of Error is

$$E[\varepsilon^2]$$

↑
Expectation

$$E[x^2] = \int_{-\infty}^{\infty} x^2 f(x)\, dx$$

$$E[\varepsilon^2] = \int_{-s/2}^{s/2} \varepsilon^2 f(\varepsilon)\, d\varepsilon$$

$$= \int_{-s/2}^{s/2} \varepsilon^2 \frac{1}{s}\, d\varepsilon$$

$$= \frac{1}{s}\left[\frac{\varepsilon^3}{3}\right]_{-s/2}^{s/2}$$

$$= \frac{1}{3s}\left[\frac{s^3}{8} + \frac{s^3}{8}\right]$$

$$= \frac{1}{3s}\left[\frac{s^3}{4}\right]$$

$$= \frac{s^3}{12s}$$

$$= \frac{s^2}{12}$$ //

Noise power = $\dfrac{v^2_{Noise}}{R}$ → Mean square value of noise.

$$\therefore SNR = \frac{v^2/12}{s^2/12}$$

$$= \frac{6v^2}{s^2}$$

$\therefore S = \dfrac{V_H - V_L}{Q}$

$V_H = +V$

$V_L = -V$

$S = \dfrac{2V}{Q}$.

$$= \frac{6v^2}{(2V/Q)^2}$$

$$= \frac{6v^2}{4v^2} \times Q^2$$

$$SNR = \frac{3}{2} Q^2$$

Subs, $Q = 2^N$,

$$\therefore SNR = 3/2 (2^N)^2 = \frac{3}{2} 2^{2N}.$$

$$(SNR)_{dB} = 10 \log_{10} \left(3/2 \; 2^{2N}\right)$$

$$= 10 \log 3/2 + 10 \log 2^{2N}$$

$$= 1.76 + N \cdot 20 \log 2$$

$$[SNR]_{dB} = [1.76 + 6.02\,N]_{dB}$$

COMPANDING :

1) It is the process of Compressing, then expanding.

2) With Companded systems, the higher amplitude analog signals are compressed prior to transmission, then expanded at receiver.

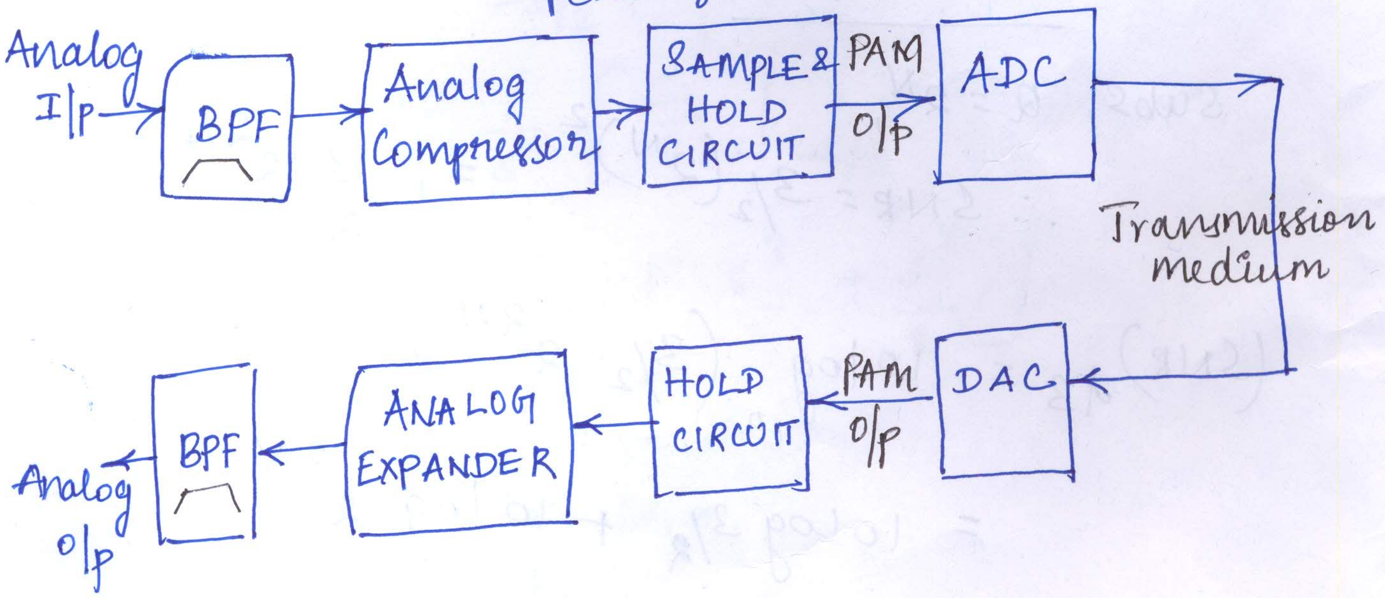
3) It may be accomplished through analog or digital techniques.

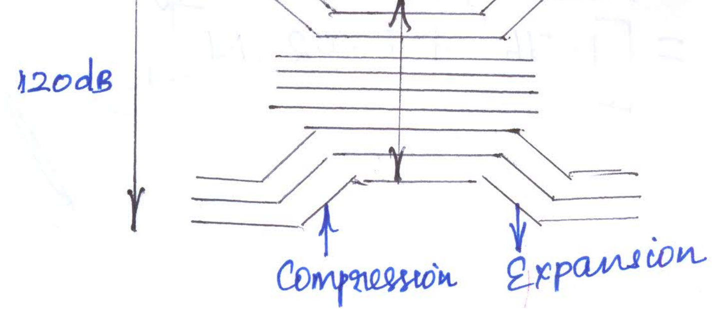
Analog Companding :

Analog compression is implemented using designed diodes inserted in the analog path in PCM transmitter prior to sample & hold.

Analog expansion with diodes, placed after ~~re~~ the LPF in PCM receiver.

PCM system:





Voice signals require constant SQR performance ove
a wide dynamic range, which means that the distortion must be proportional to signal amplitu
for any i/p signal level. This requiries logarith
compression ratio.

Two analog Companding method used which called as log pcm codes. They are

1) μ-law companding
2) A-law companding.

μ-law Companding: Used in Japan, US.

The compression characteristics for μ-law is

$$V_{out} = \frac{V_{max} \cdot \ln\left[1 + \mu V_{in}/V_{max}\right]}{\ln(1+\mu)}.$$

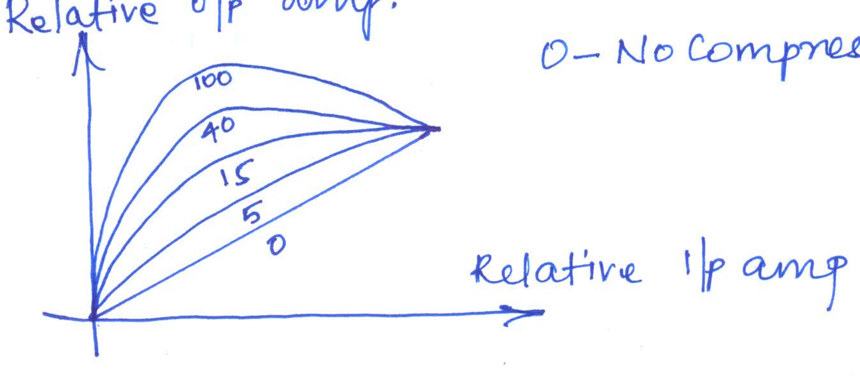
V_{max} - Maximum uncompressed analog i/p amplitude.

V_{in} - amplitude of i/p signal.

μ - parameter used to define the amount of compression.

V_{out} - compressed o/p amplitude.

Higher μ gives more compression. μ determin the range of signal power in which SQR is Constant.



0 - No Compressio

A-law Companding :

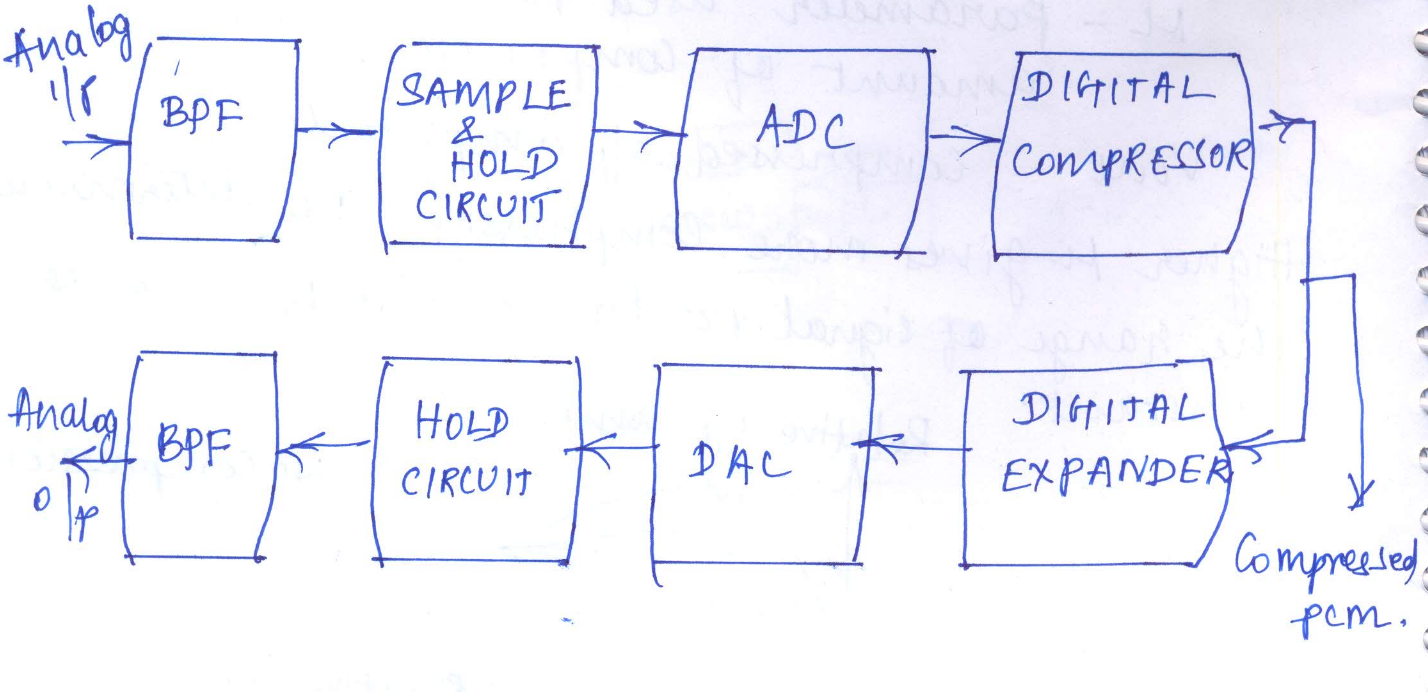
used to approximate true logarithmic Companding. A-law has a slightly flatter SQR than μ-law. the compression characteristics is

$$V_{out} = V_{max}\left(\frac{A\, V_{in}/V_{max}}{1+\ln A}\right), \quad 0 \le \frac{V_{in}}{V_{max}} \le \frac{1}{A}$$

$$= V_{max}\left(\frac{1+\ln\left(A\frac{V_{in}}{V_{max}}\right)}{1+\ln A}\right), \quad \frac{1}{A} \le \frac{V_{in}}{V_{max}} \le 1.$$

Digital Companding :

It Involves compression at the transmit end after the input samples are converted to a linear pcm code and expansion at the receive end prior to PCM decoding.



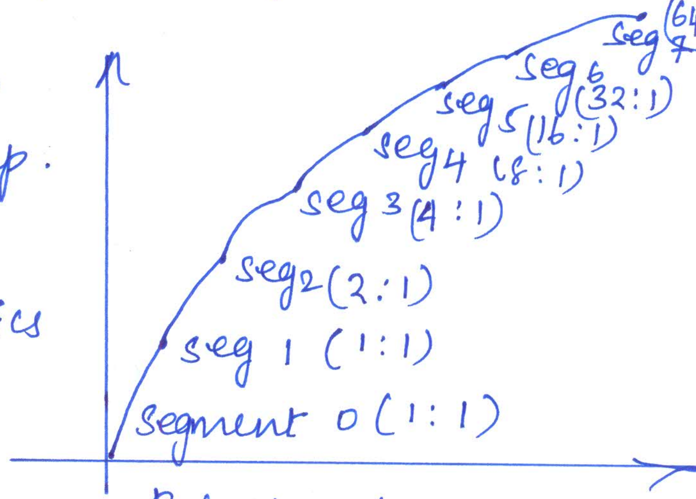
The most recent digitally Compressed PCM system use 12 bit linear code to 8-bit compressed code.

This process closely resembles $\mu = 255$ analog compression curve by approximating the curve with a set of 8 straight line segments slope of each segment is one half that of the previous segment.

12 bit to 8 bit digital Compression Curve.

O/p amp.

M-255 Compression characteristics



Relative i/p amp

It involves compression in the transmitter after the input sample has been converted to a linear PCM code and then expansion

Eigh bit Compressed code consists of sign bit, three bit segment identifier and a 4 bit magnitude code that specifies the quantization interval within the specified segment.

Sign bit	3 bit Identifier	4 bit quantization

* Sign bit 1 = +
  0 = -
* 3 bits 000 to 111
* 4 bit 0000 to 1111

<u>Example:</u>

	Transmit		Receive		
Segment	12 bit linear code	8 bit compressed	8 bit code	Recovered 12 bit code	Segment
0 —	S0000000ABCD	S000 ABCD			
1 —	S0000001ABCD	S001 "			
2 —	S000001ABCDX	S010 "			
3 —	S00001ABCDXX	S011 "			
4 —	S0001ABCDXXX	S100 "			
5 —	S001ABCDXXXX	S101 "			
6 —	S01ABCDXXXXX	S110 "			
7 —	S1ABCDXXXXXX	S111 "			

Dynamic Range : Ratio of largest possible magnitude to the smallest possible magnitude that can be decoded by digital to analog converter in Receiver.

$$DR = \frac{V_{max}}{V_{min}} \qquad [DR]_{dB} = 20\log\left(\frac{V_{max}}{V_{min}}\right)$$

Coding Efficiency : Ratio of minimum number of bits required to acheive a certain dynamic range to the actual number of pcm bits used.