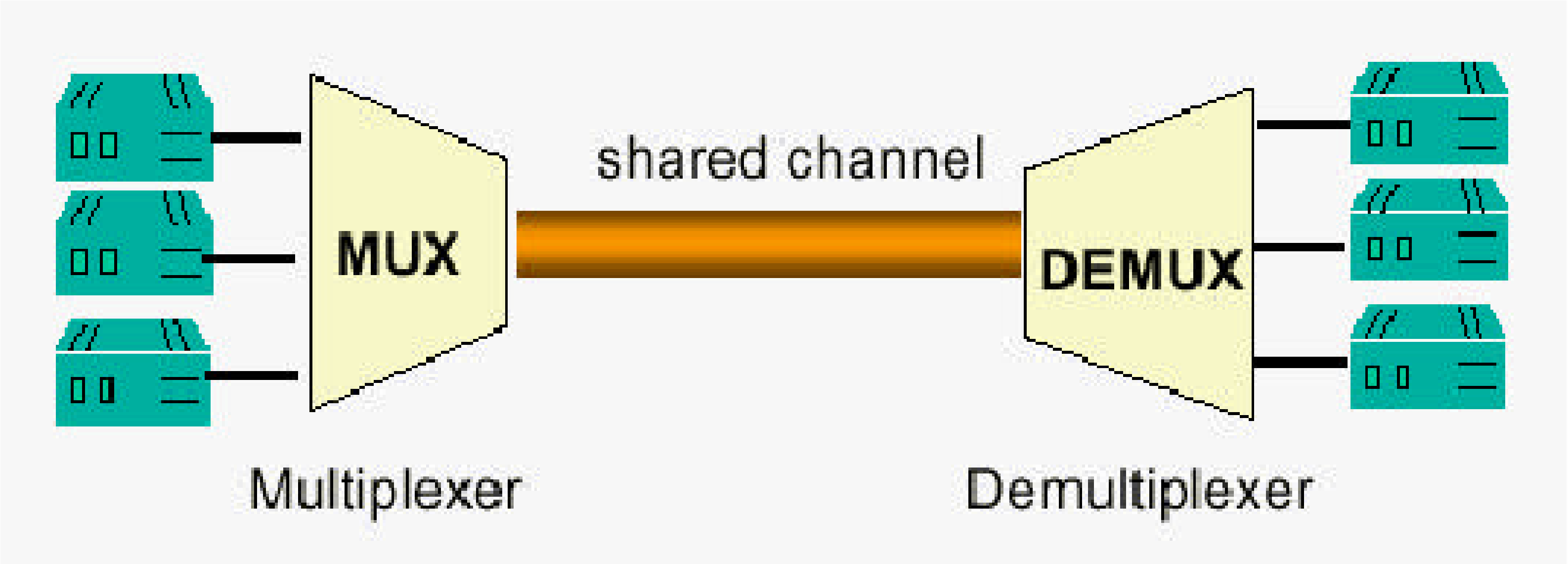
PULSE CODE MODULATION

Multiplexing

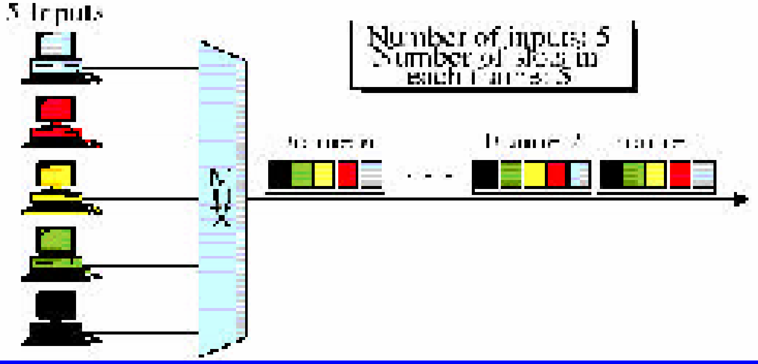


A common application is in long-haul communications

- E.g telephone systems, radio, …
- High-capacity trunks, such as fiber, coaxial, or microwave links are shared
- The link is able to carry *n* separate channels of data, and thereby minimizes infrastructural costs.

Multiplexing

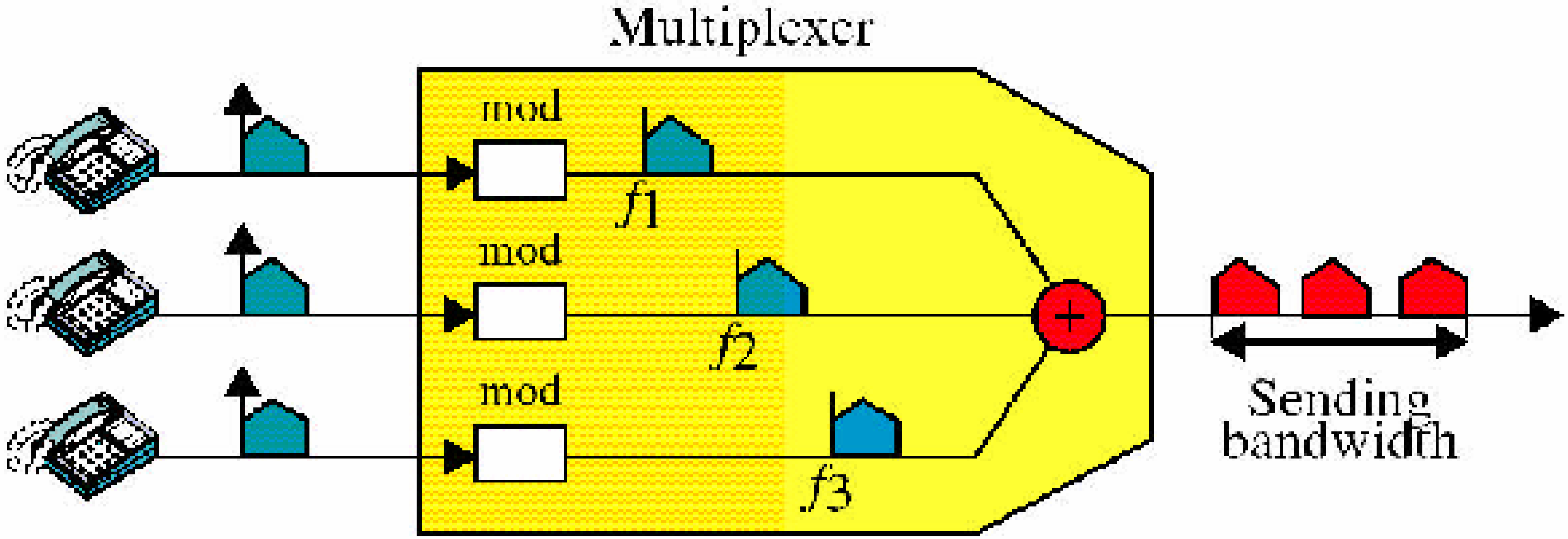
- **Multiplexer** combines data from the *n* input lines and transmits over the high-capacity data link
- **Demultiplexer** accepts multiplexed data stream, separates data according to channel, and delivers them to the appropriate output lines.
- Types of multiplexing
  - Frequency Division Multiplexing (FDM)
  - Time Division Multiplexing (TDM)



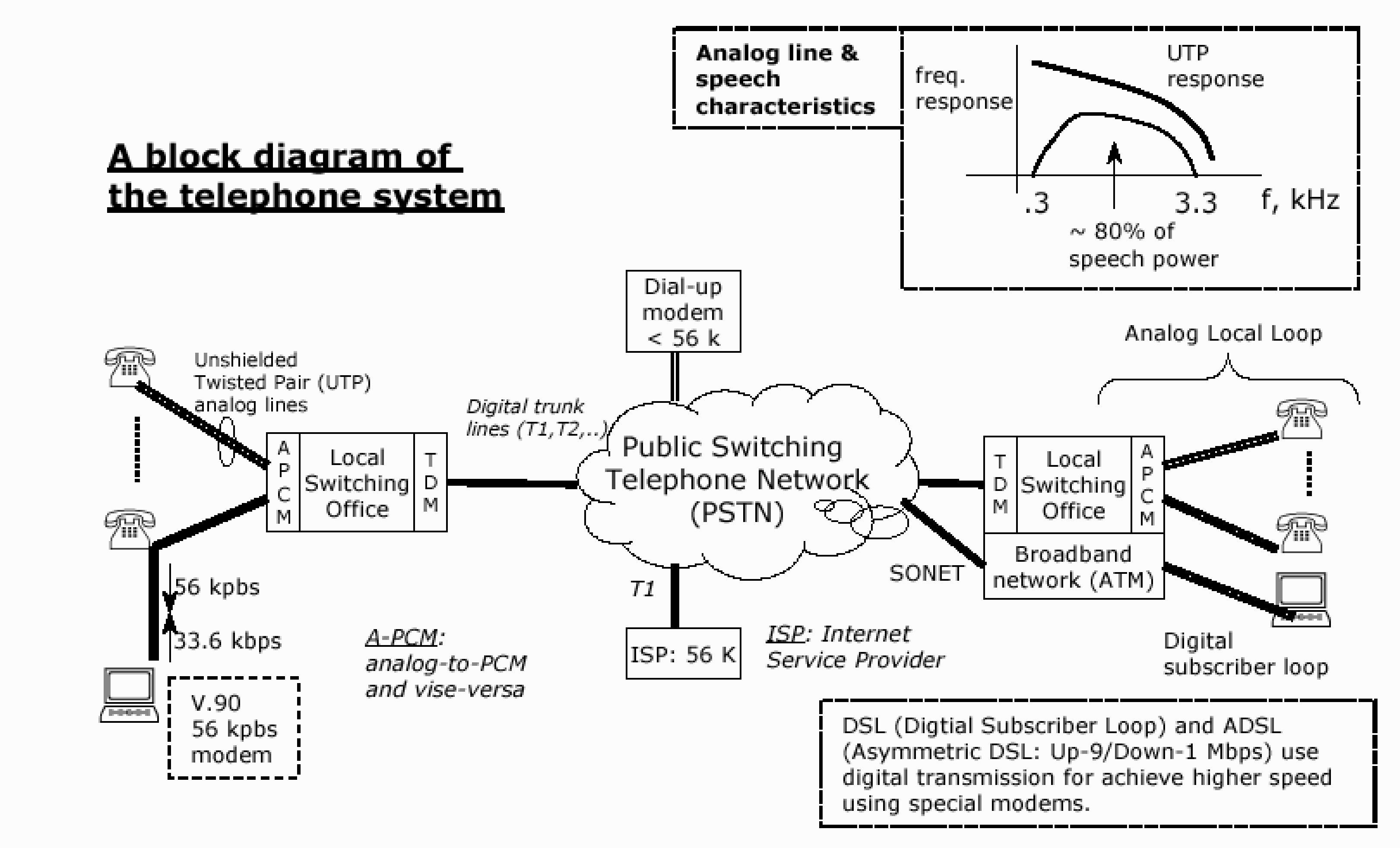
Frequency Division Multiplexing

- FDM is an analog technique
- Useful bandwidth of medium exceeds required bandwidth of individual channels
- Each signal is modulated to a different carrier frequency
- Carrier frequencies are separated so signals do not overlap (the gaps are "guard bands")
- e.g. broadcast radio, TV, telephone
- Channel is allocated even if no data

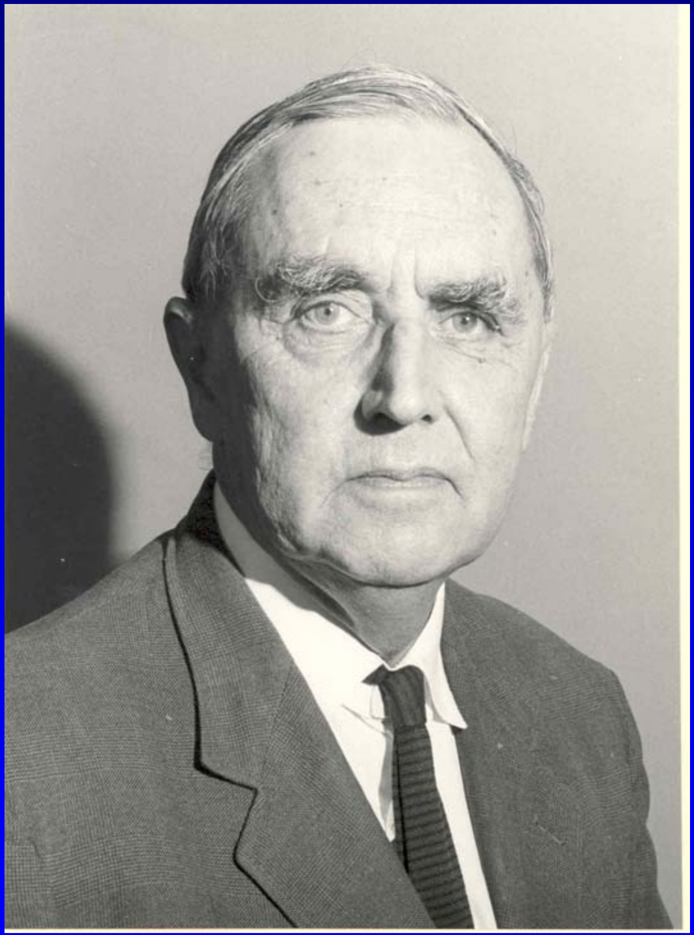
FDM – view 2



Telephone System



Alec Reeves



- 1937 A.H. Reeves conceived idea of PCM and digital transmission
- 1938 French Patent
- 1939/42 British/US Patents
- 1962 AT&T introduces PCM in US

Coding

Converting an analogue speech (or other) signal into digital form is known as Waveform Coding. The term <u>coding</u> is also applied to other operations e.g.:

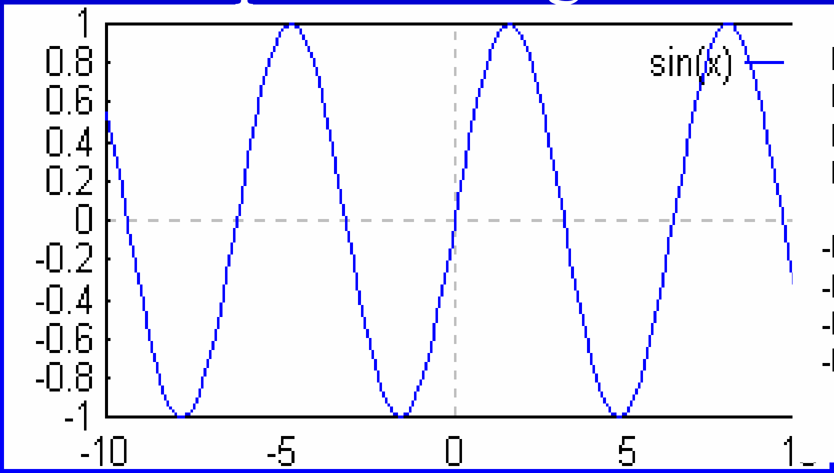
- *Line coding* modifies the analogue or digital source to make it best suited for transmission (e.g HDB3).
- *Channel coding* adds redundancy to improve detection / correction of errors
- *Source coding* typically achieves bit rate reduction by analysing signal structure (e.g. MPEG/JPEG).

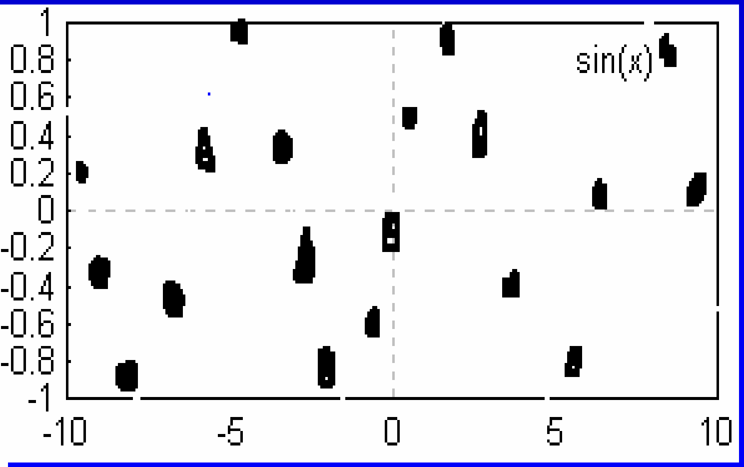
PCM - Three Steps

- Sampling
- Quantising
- Encoding

Sampling of a continuous signal

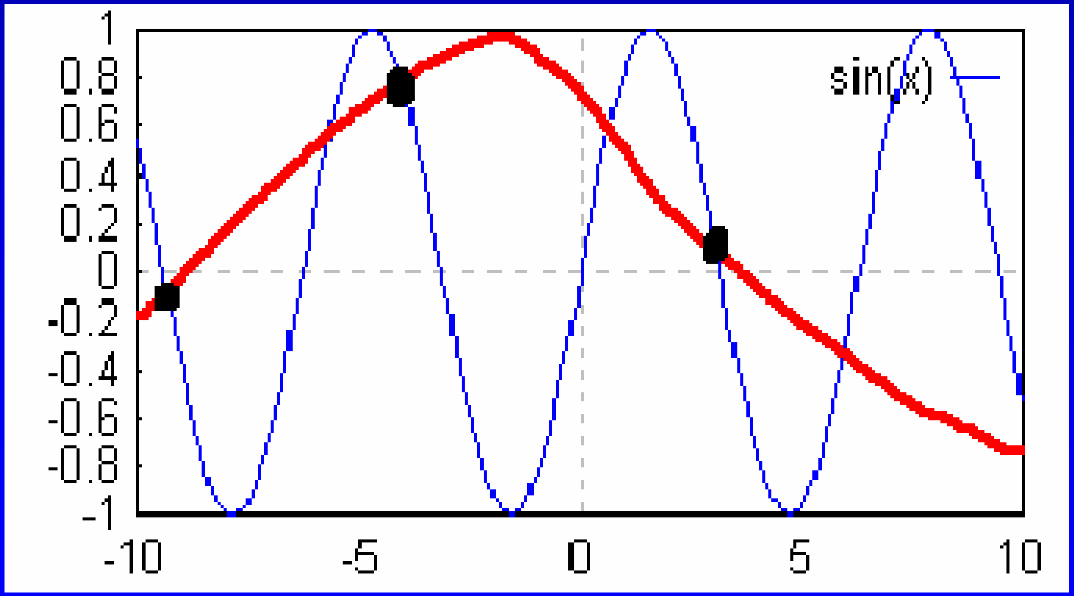
- A continuous signal can be sampled at discrete intervals in time
- The sine wave on left can be reconstructed from "samples" on right





Sampling Theorem

- However, if insufficient samples are taken, a lower frequency signal will result after reconstruction



- *This low frequency o/p is called an Alias of the original signal*

Nyquist Sampling

Key goal - minimise number of samples for accurate representation of the signal?

– *Nyquist* criterion; sample at twice maximum frequency in a baseband signal.
– Sampling at less than this results in *aliasing*

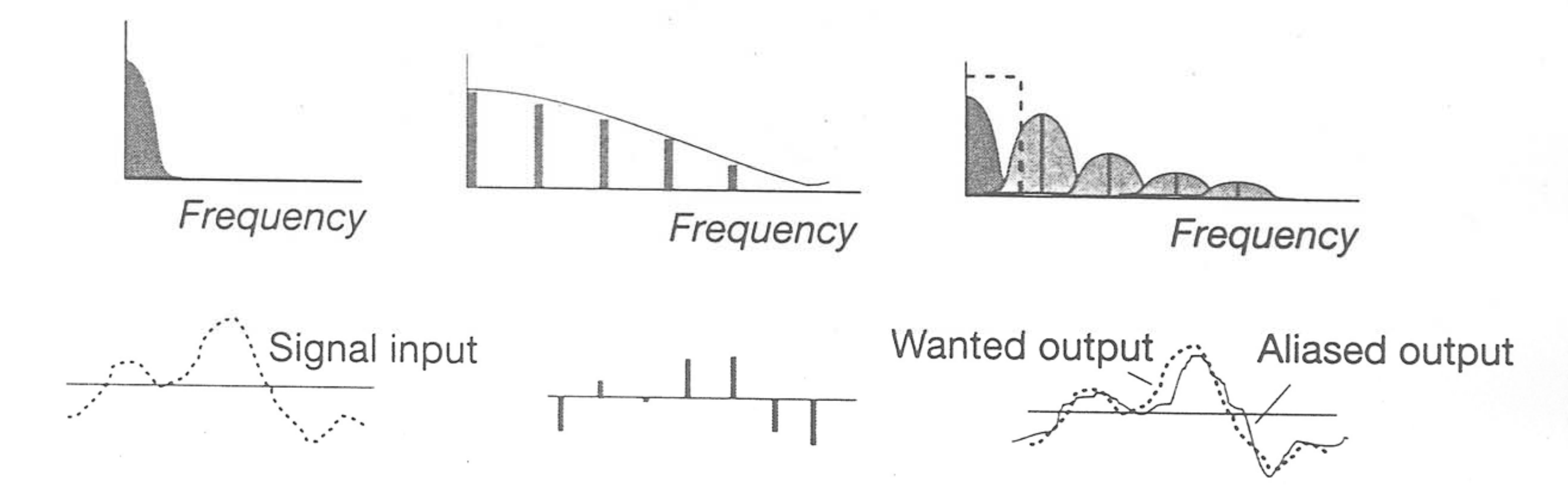
Nyquist Sampling Theorem

- This states
  - "it is sufficient to know the value of a waveform at $2f$ instances per second to recover the original wave, which is assumed to have no frequency component greater than f"
- $f_s = 2f_B$
  - where f_s = sampling frequency and f_B = highest frequency component of signal

Aliasing

– If the sampling criterion is not met and we sample at *less* than the twice the maximum frequency of the input waveform, the sum and difference components associated with each harmonic of the input waveform overlap with those of adjacent harmonics and we can no longer separate out the sampled waveform by filtering.

Alias Effect



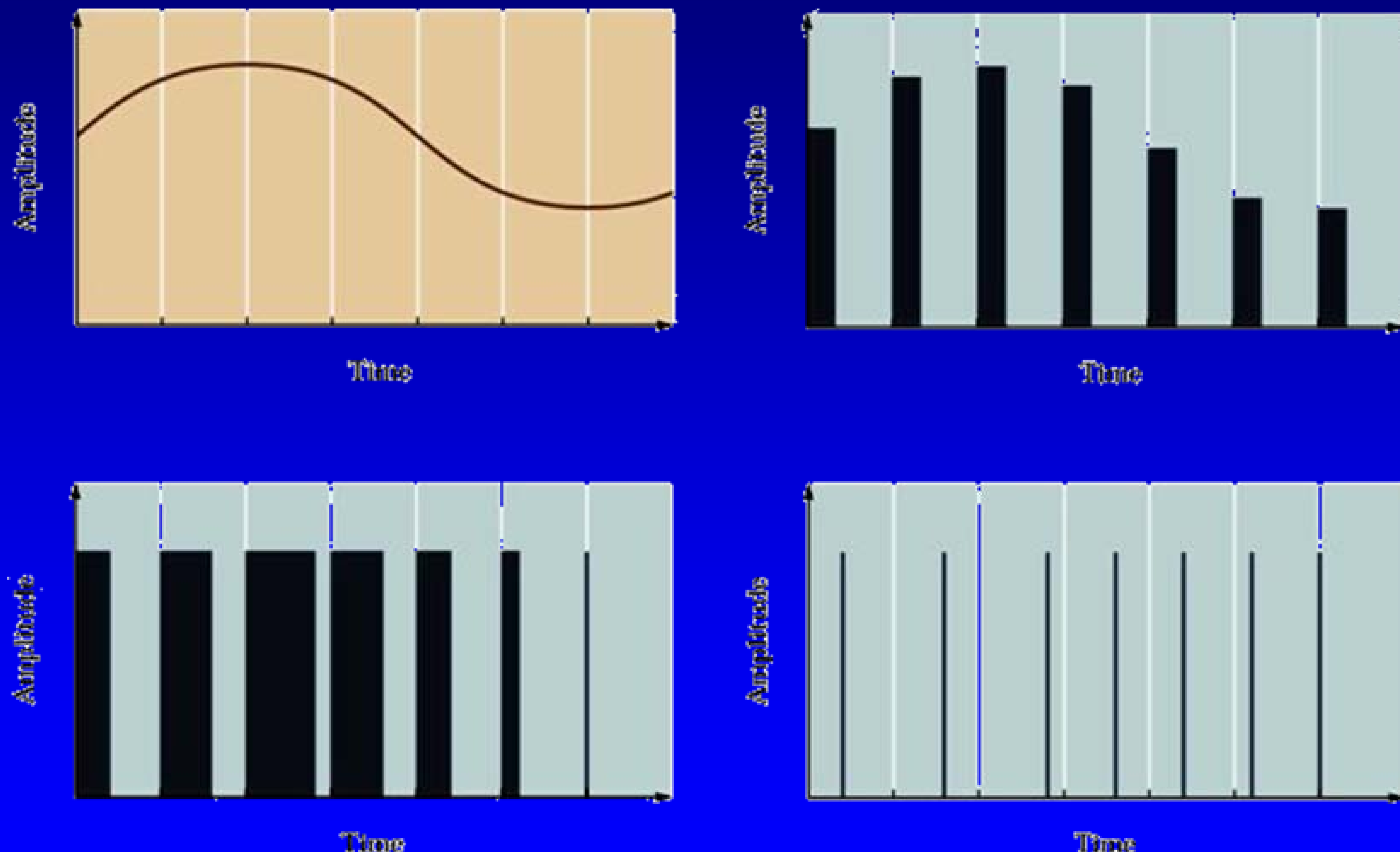
Practical Sampling

- Need to filter out baseband components above the range of interest
- Use a sampling frequency of ~2.2 f_{max} to allow for practical filters
- e.g. speech telephony 300Hz – 3.4kHz is sampled at 8 kHz
- CD sampling at 44.1 kHz for 20 kHz audio band

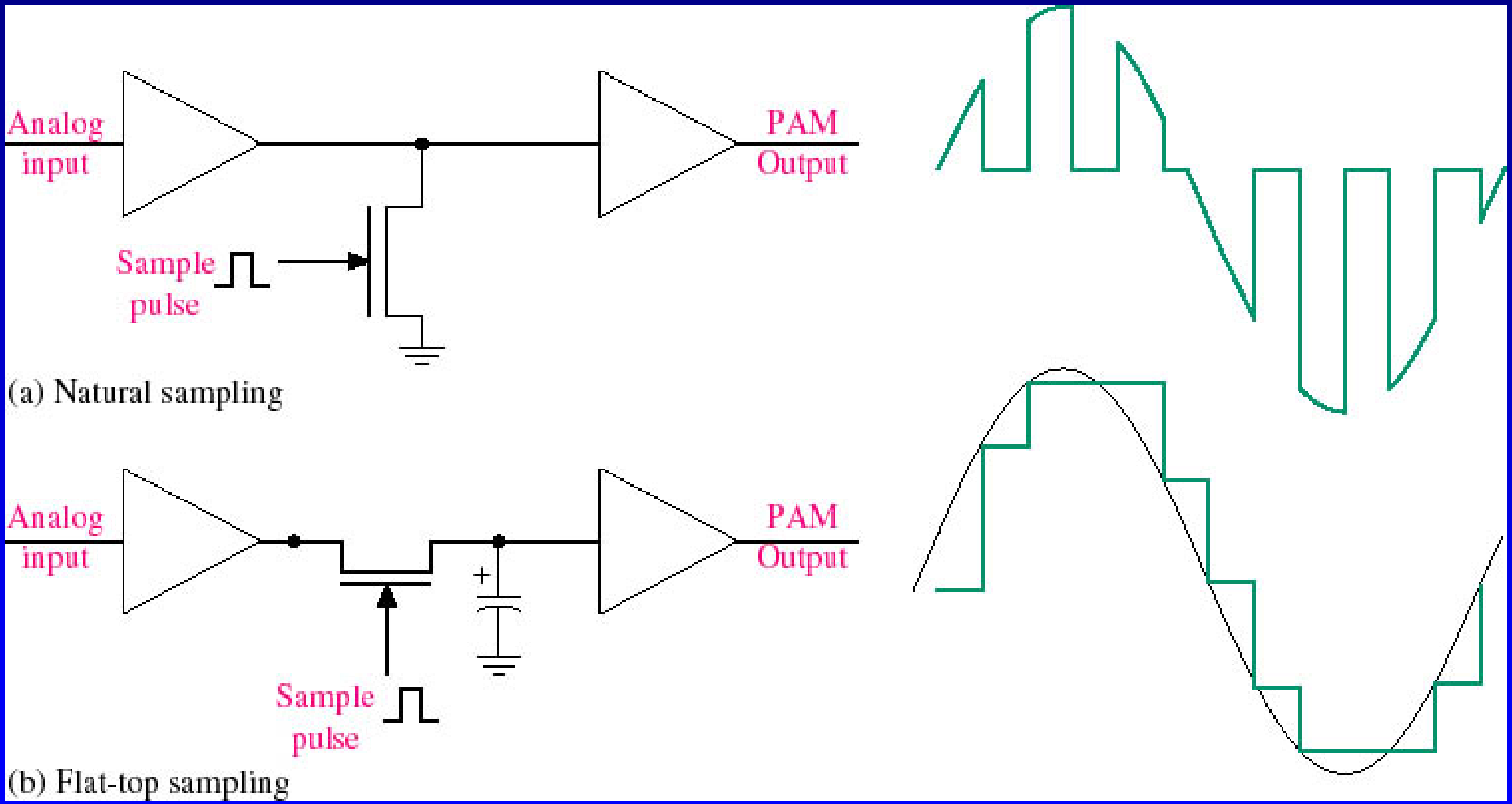
Sampling

- Sampling alone is not a digital technique
- The immediate result of sampling is a **pulse-amplitude modulation (PAM)** signal
- PAM is an analog scheme in which the amplitude of the pulse is proportional to the amplitude of the signal at the instant of sampling.

Analog Pulse-Modulation Techniques



Sample and Hold



Sampled Signal

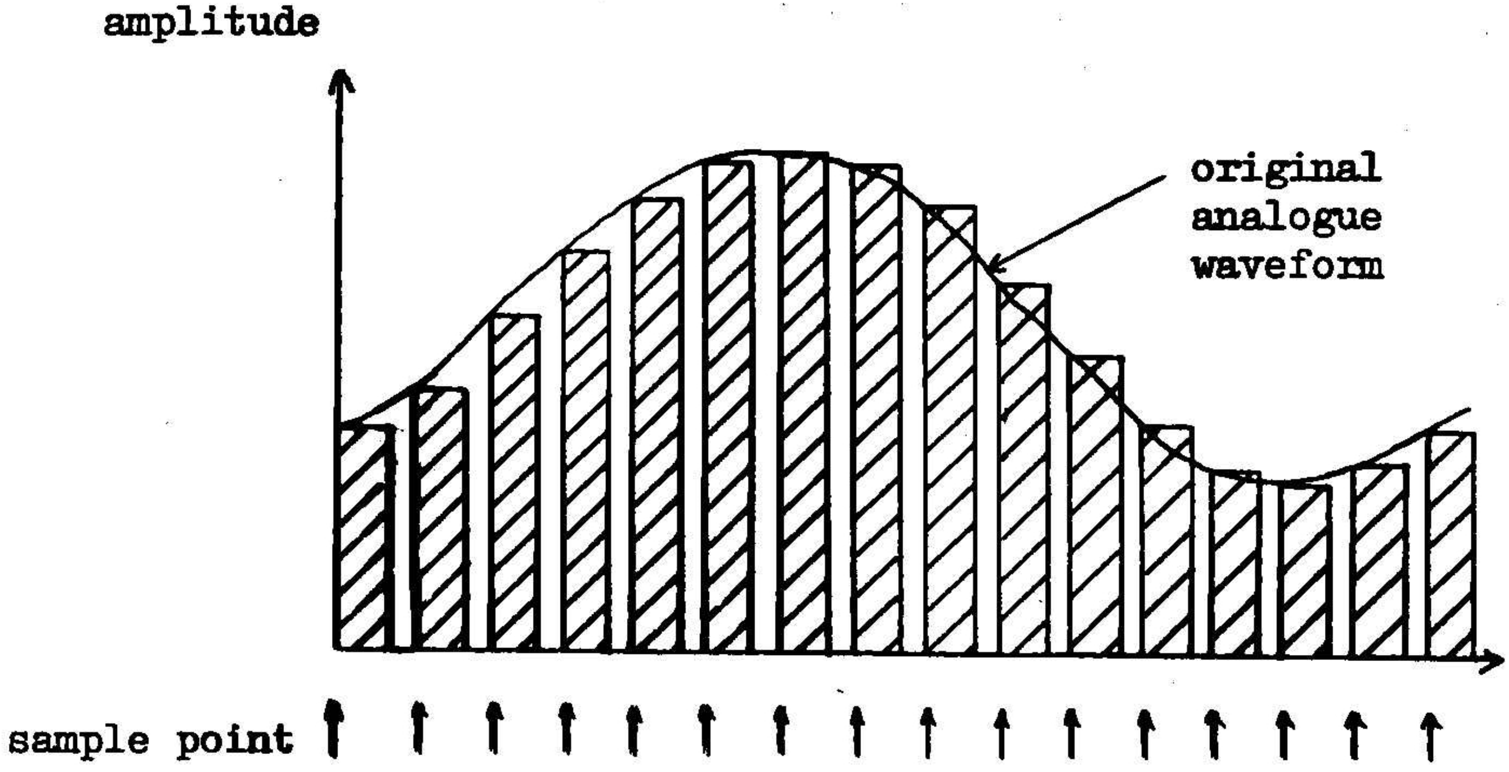
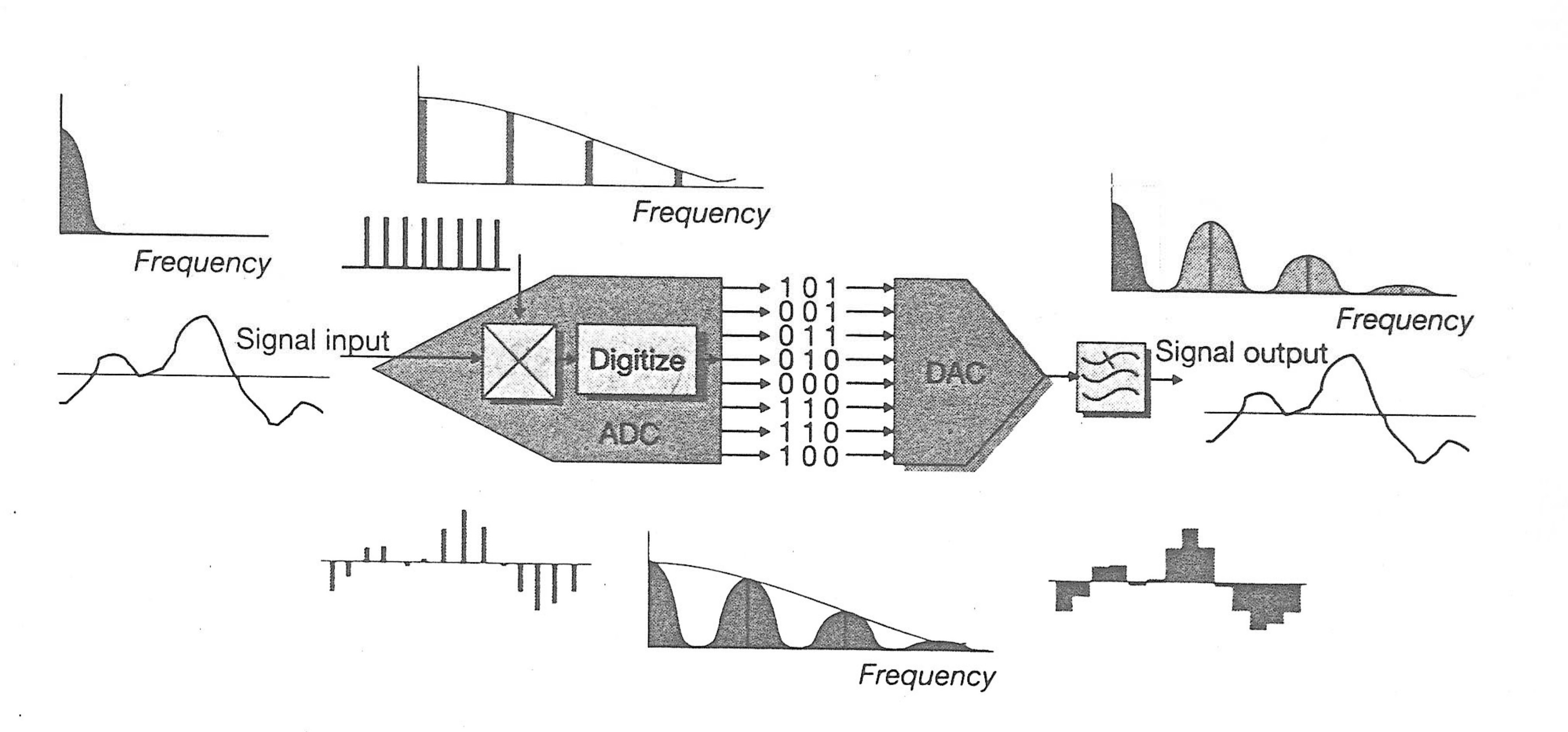


Figure 2: Sample and Hold Pulses

Nyquist Sampling System



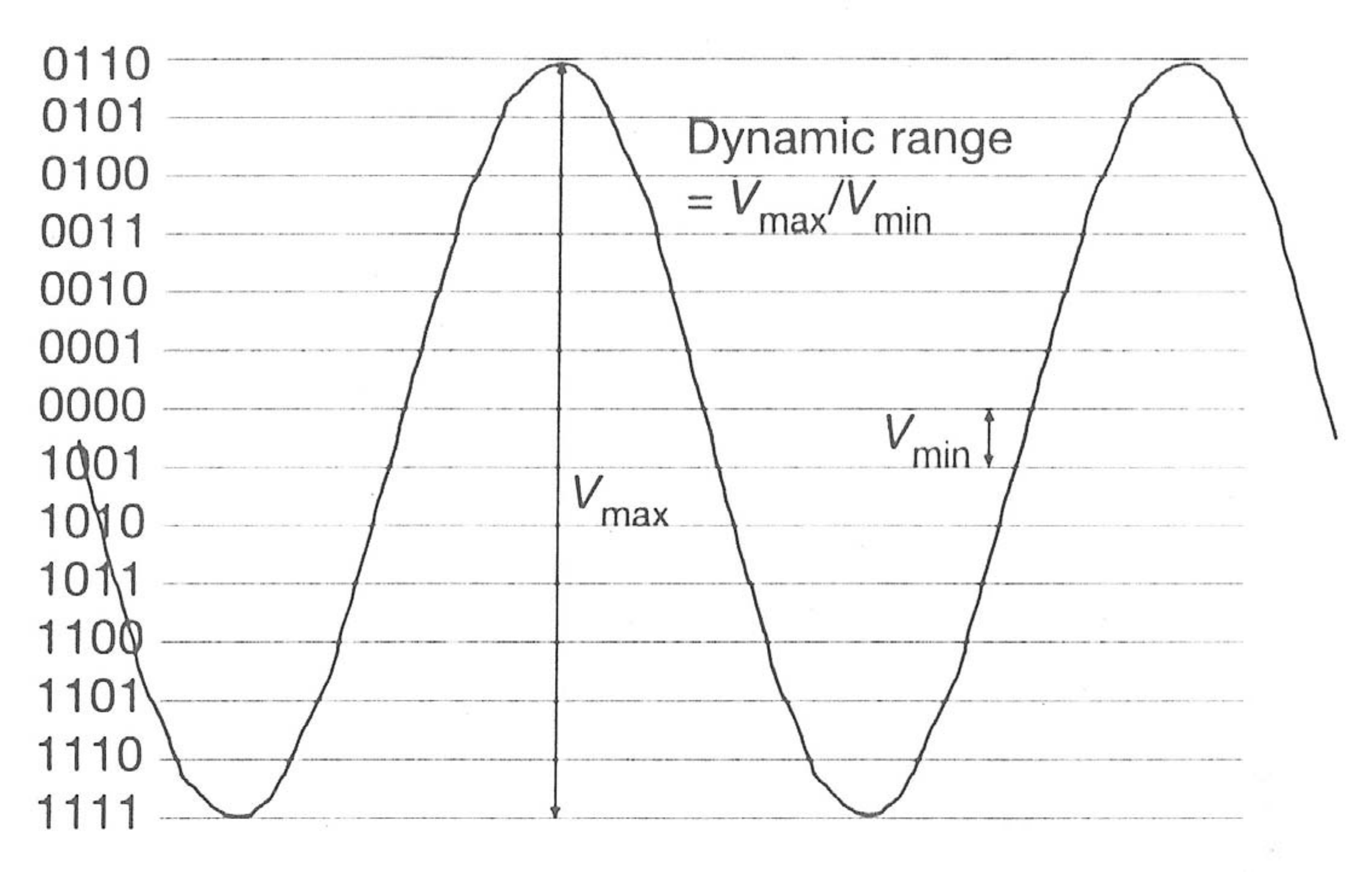
Quantising

- Since the original signal can have an infinite number of signal levels, the quantising process will produce errors called **quantising errors** or **quantisation noise**
- The dynamic range of a system is the ratio of the strongest possible signal that can be transmitted to the weakest discernible signal
- In a linear PCM system, the maximum dynamic range is found by:

$$DR = (1.76 + 6.02n)\ \mathrm{dB}$$

- where n is the number of bits per sample

Dynamic Range…1



Dynamic Range…2

– Important that A/D converter can deal with both large and small signals.
– Ratio of V_{max} to V_{min} over which converter will operate is its Dynamic Range
– Depends on the number of bits the converter uses
– More bits means more *quantisation* levels

Dynamic Range…3

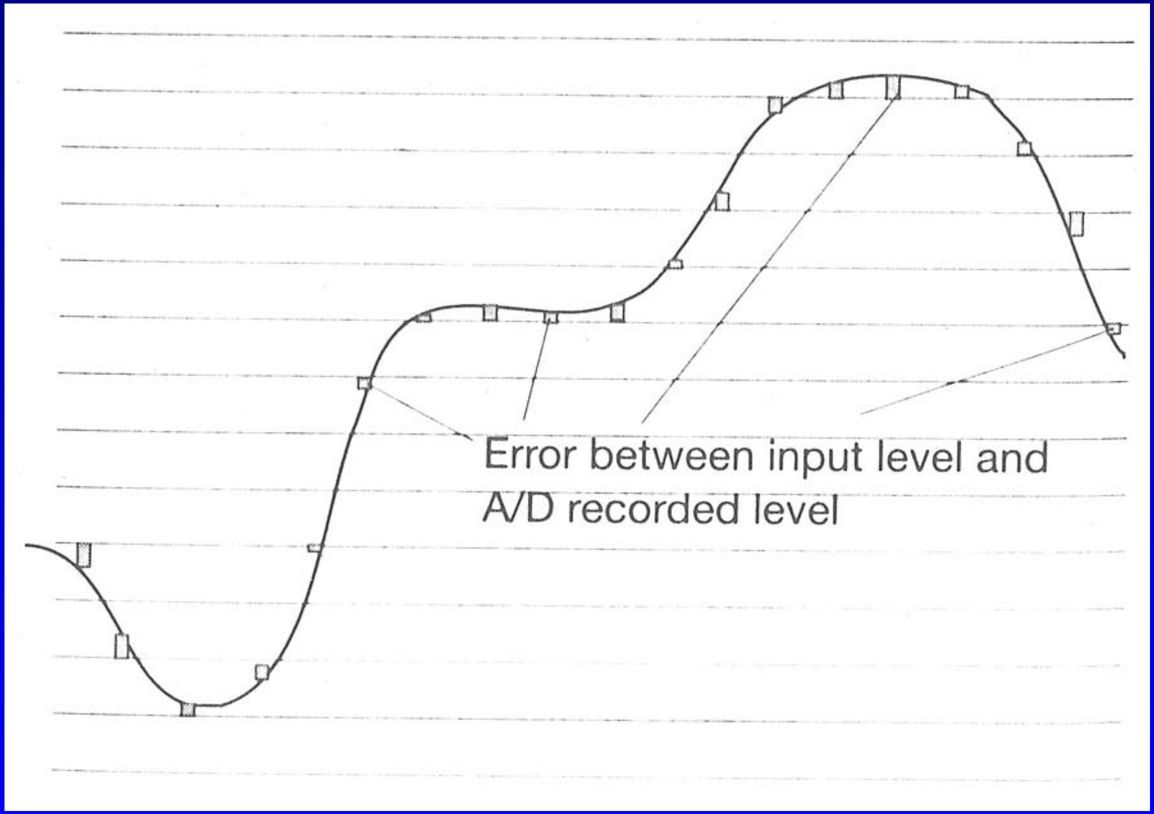
– An n-bit converter can differentiate between $2^n = M$ discrete signal levels

– Minimum signal variation it can detect is V_{max} / M volts, known as the quantisation step size

– Dynamic range approximates to n x 6 dB

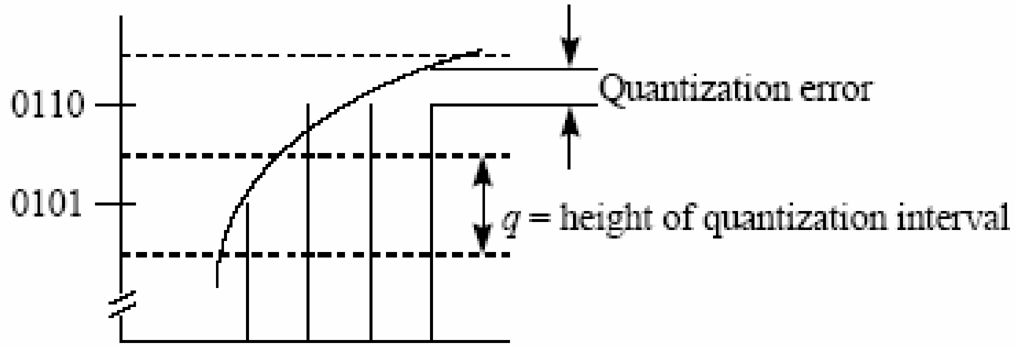
[$20\log V_{max}/V_{min} = 20\log 2^n = n \times 20\log 2$]

Quantisation Noise…1



Quantisation Noise

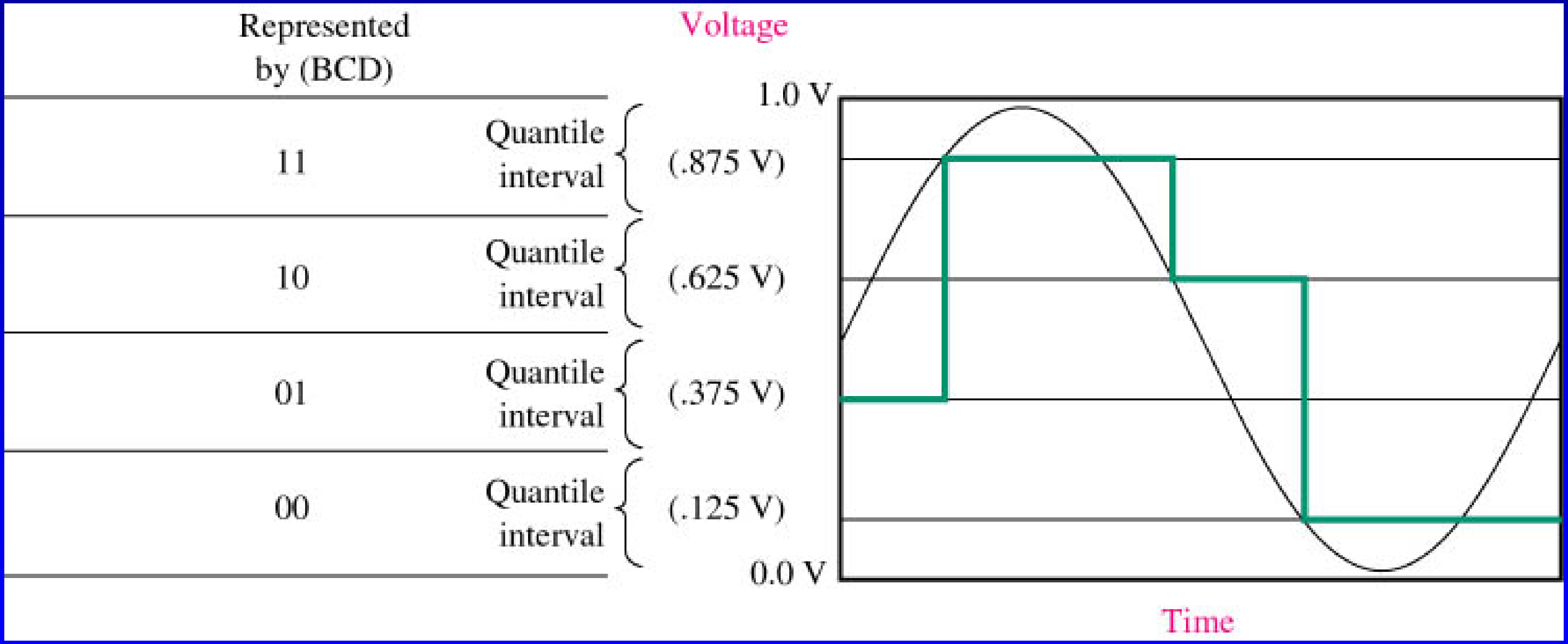
- Assume:
  - Analog signal is equally likely to fall anywhere in the interval, so quantization errors are uniformly distributed (between $-q/2$ and $q/2$)
  - Errors in adjacent samples are independent (generally true if signal is not oversampled)
- Then the quantization errors are random signals perturbations -- or noise



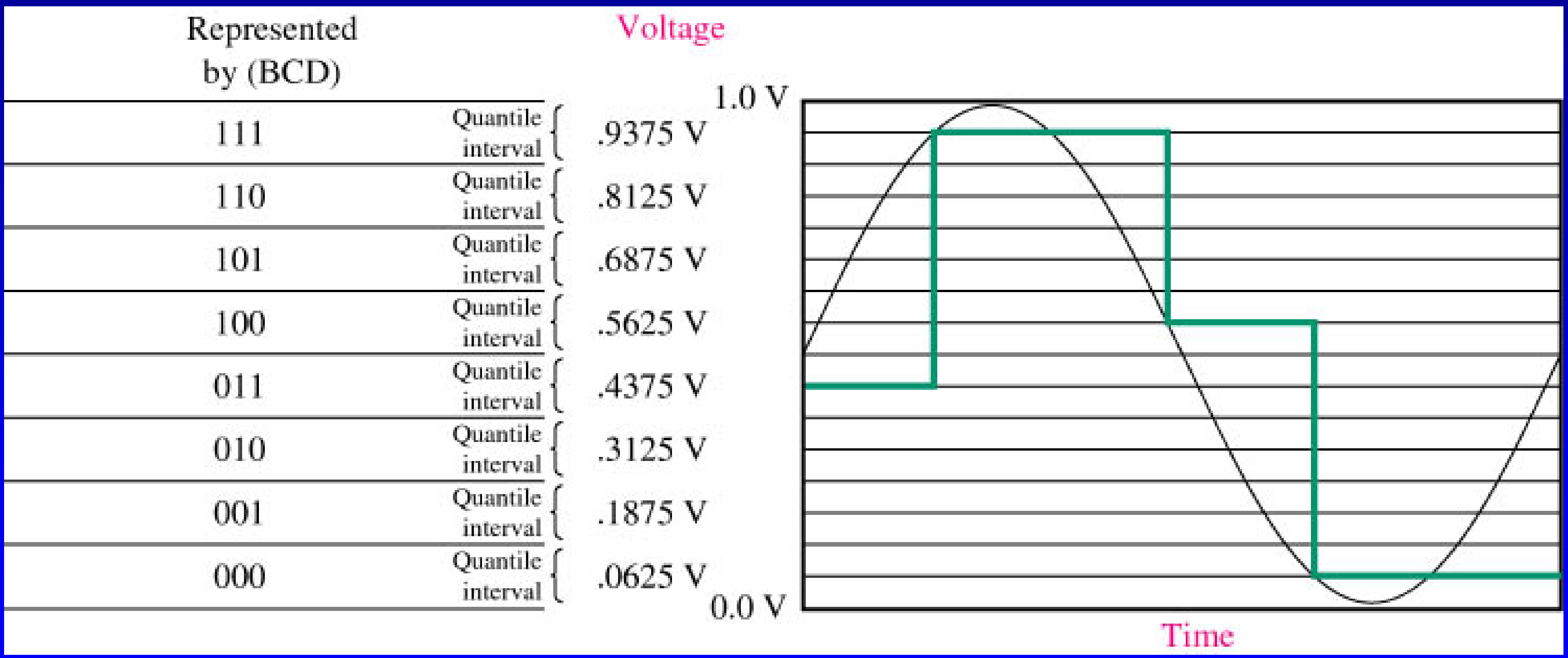
Quantisation Noise…2

- The other important parameter in any source encoding scheme is the amount of noise or distortion introduced.
- For waveform encoding this is amplitude errors that depend how close the input waveform is to any quantisation level

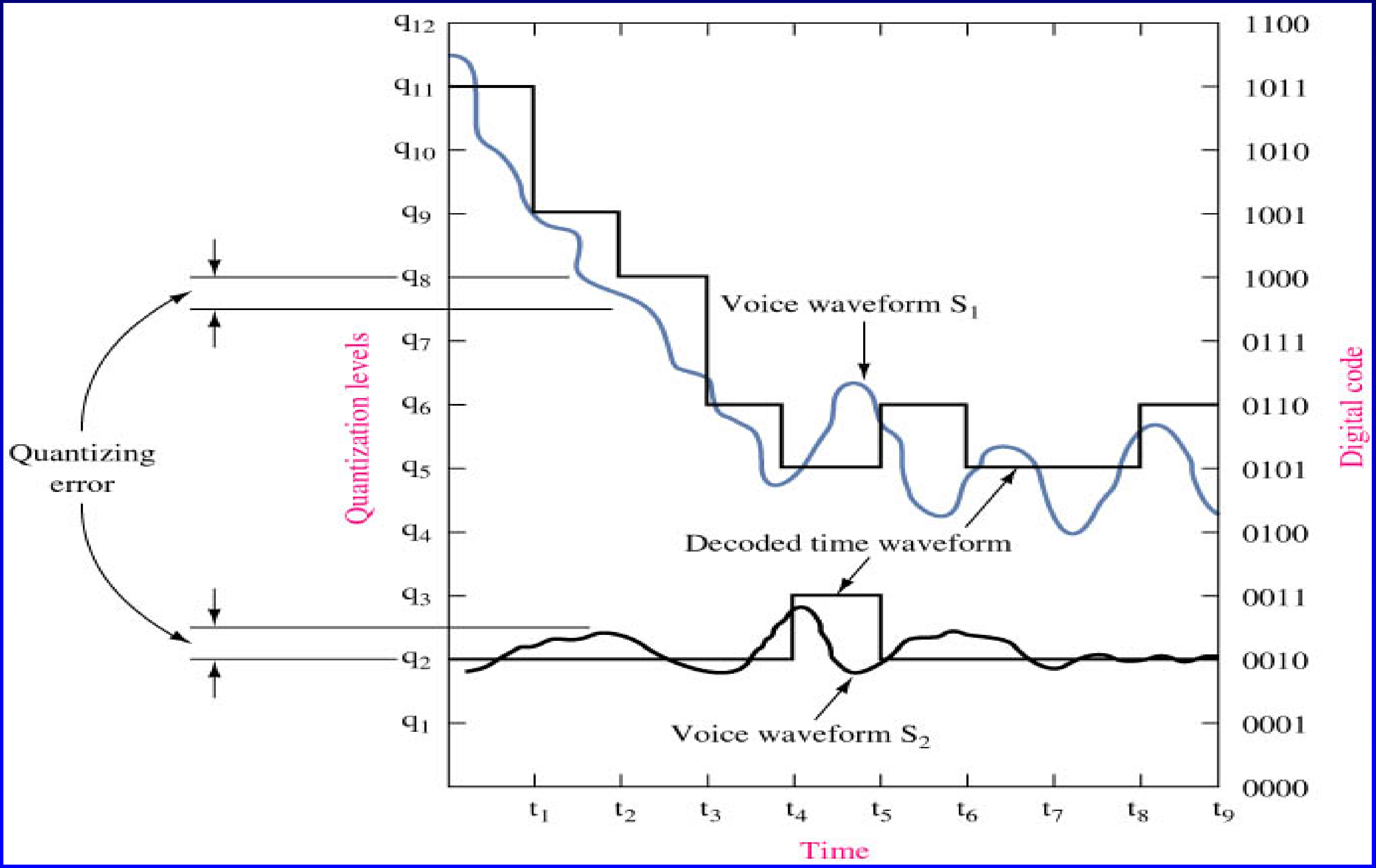
2 Bit A-D Converter



3 Bit A-D Converter



4 bit Analog Digital PCM Quantisation



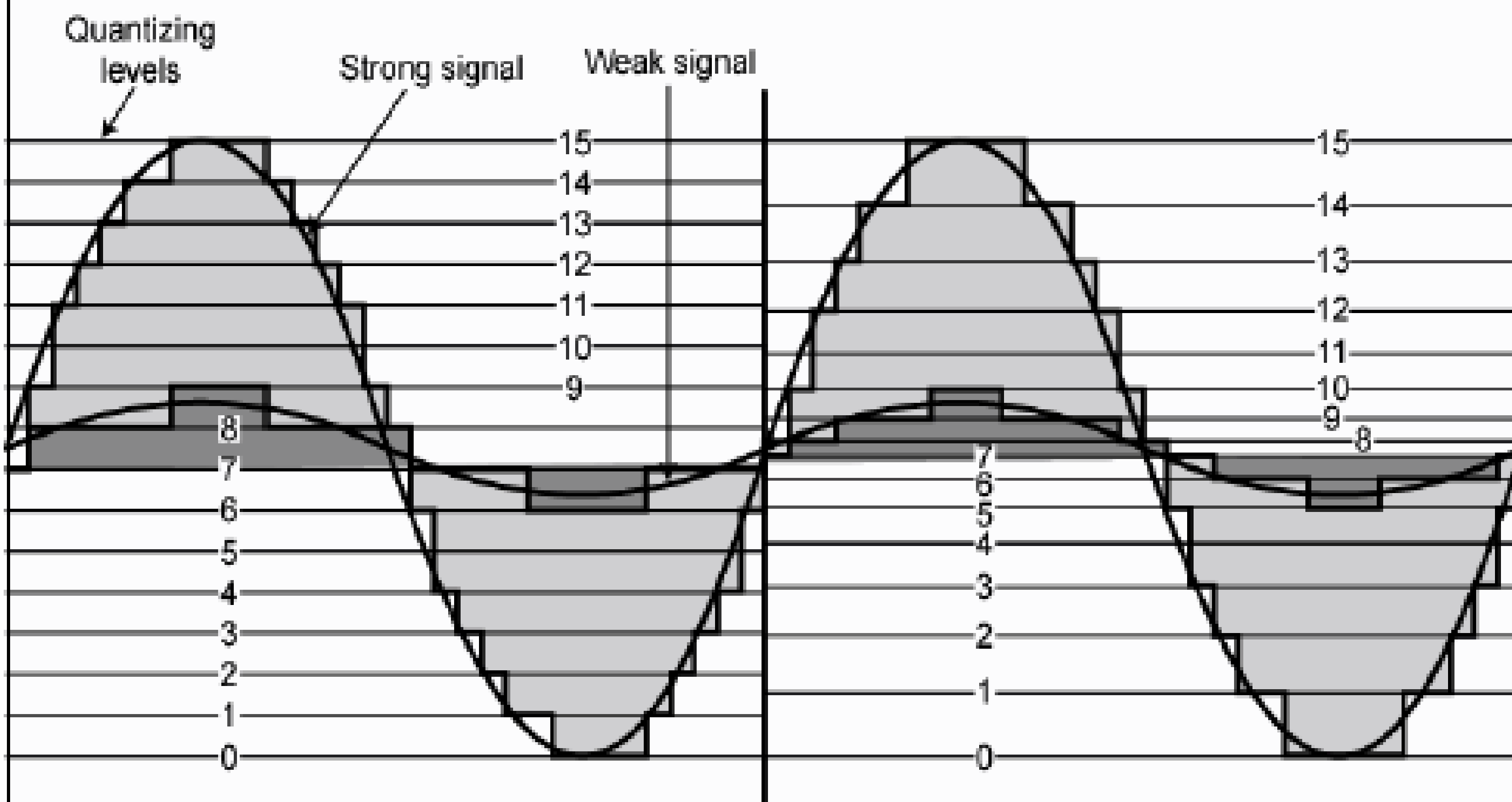
Non-Linear Quantisation `Companding'

- **Companding** is used to improve dynamic range
- Compression is used on the transmitting end and expanding is used on the receiving end, hence *companding*
- ***μ- law*** Companding is used by U.S. phone companies
- Vout= (Vmax*ln(1+μVin/Vmax))/ ln(1+ μ)

Non-Linear Quantisation Companding

- A technique for reducing the number of bits while achieving an equivalent dynamic range or signal to quantisation noise level
- COMPressing and expANDING
- *Decrease quantisation step size for small signals and increase for large signals*
- International standards for telephony
  - A Law – European
  - μ Law – USA

Effect of Non-Linear Coding



(a) Without nonlinear encoding (b) With nonlinear encoding

Encoding Laws

μ Law – Original Bell T1 system

$$y = \frac{\log_e(1+\mu x)}{\log_e(1+\mu)}$$

Where μ =100

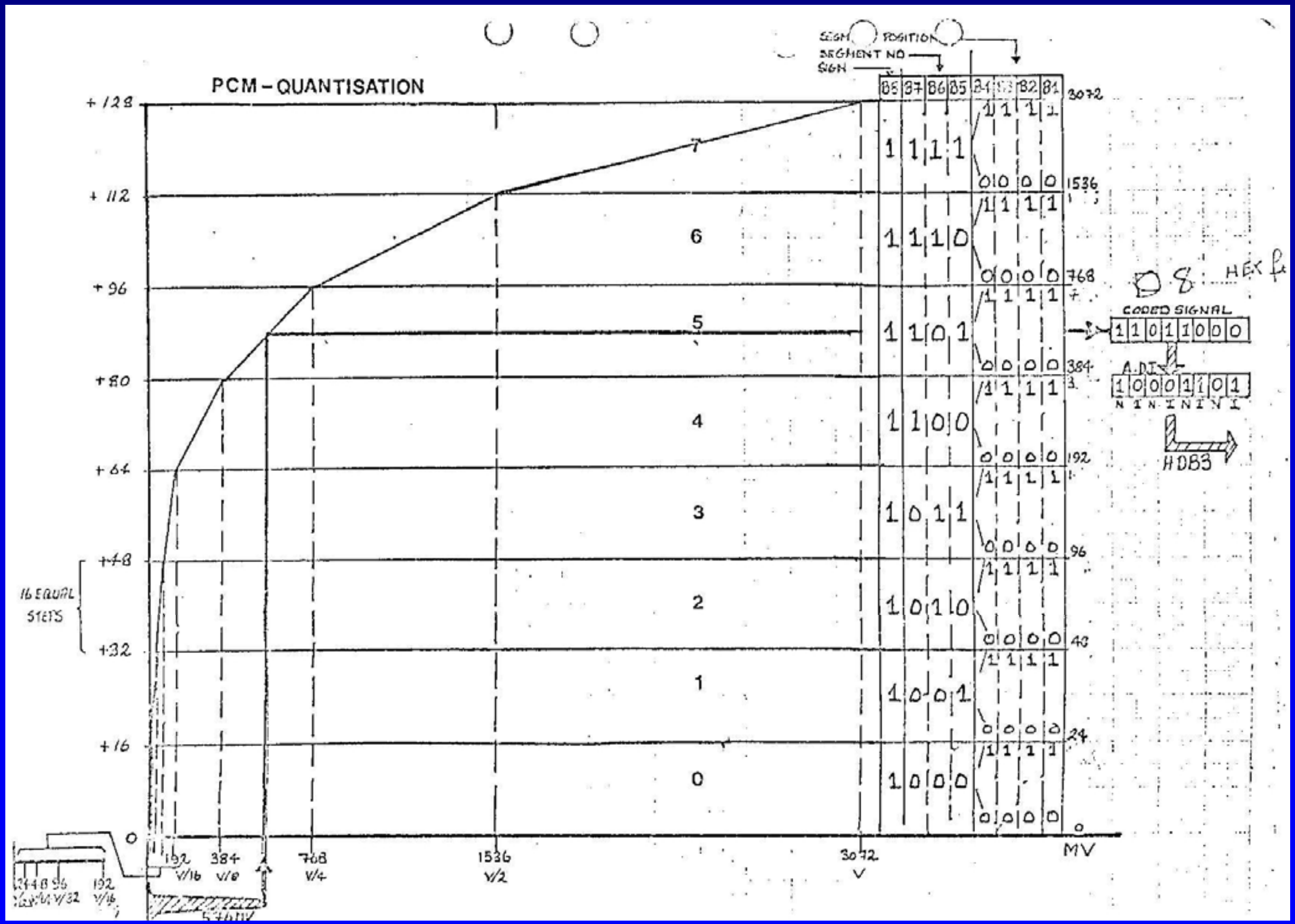
A- Law CCITT 30-channel system

$$y = \frac{1+\log_e(Ax)}{1+\log_e A} \quad for \quad \frac{1}{A} \le x \le 1$$

$$y = \frac{Ax}{1+\log_e A} \quad for \quad 0 \le x \le \frac{1}{A}$$

Where A=87.6

Non Uniform Quantisation

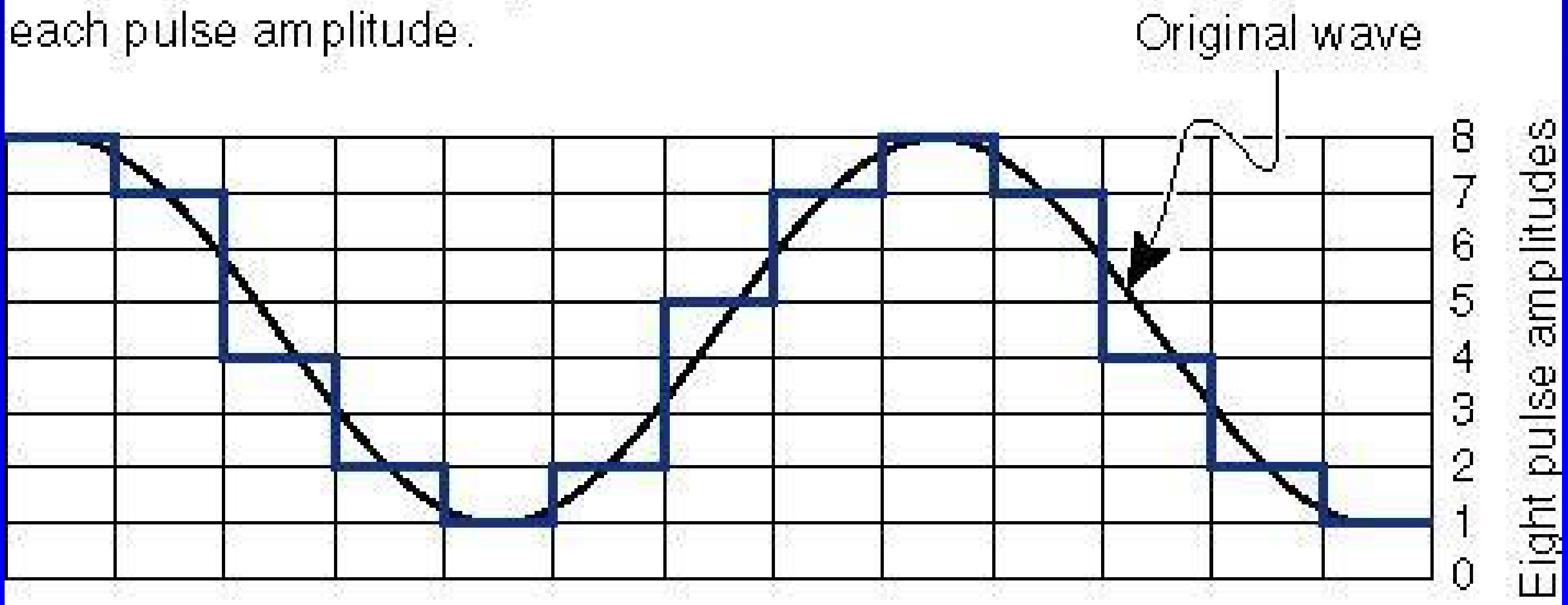


Encoding

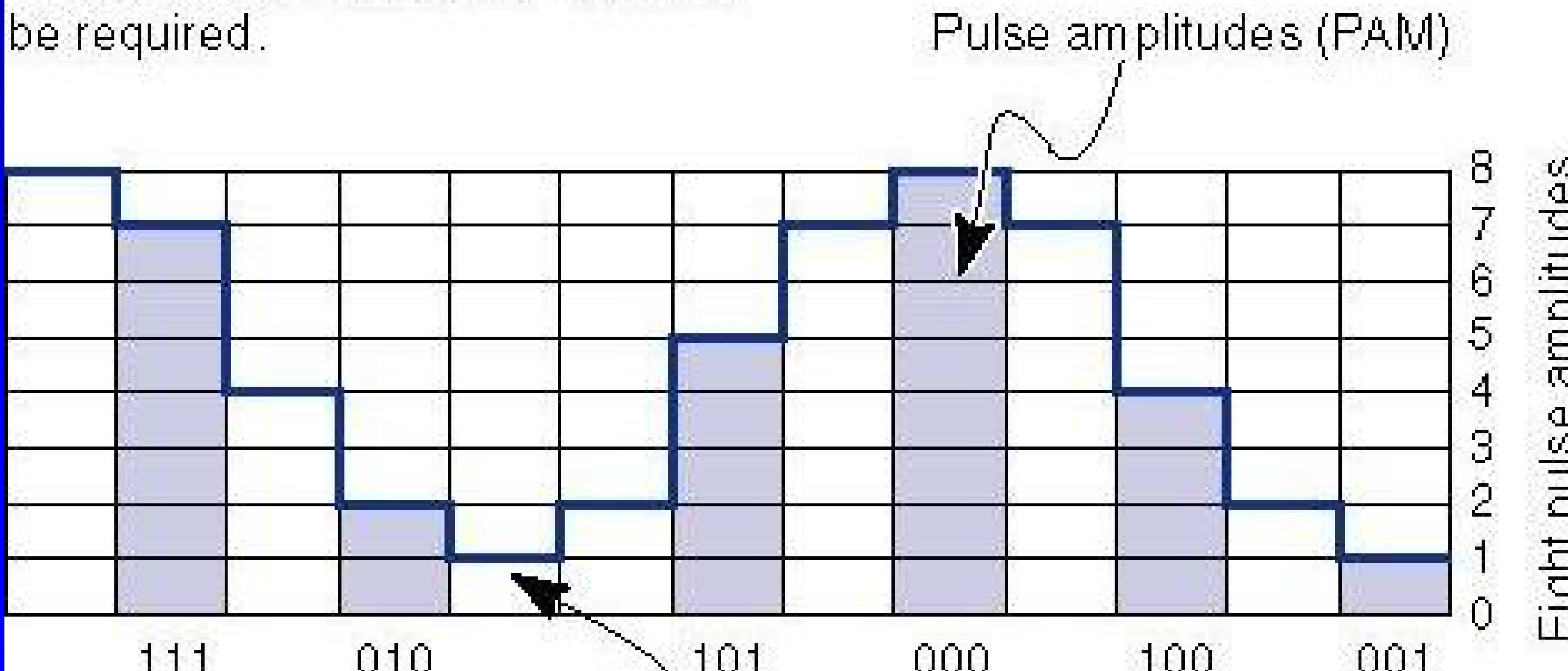
- Generate binary numbers (words) representing quantised values of the samples
- Word length depends on number of quantisation levels e.g. 8 bits => $2^8 = 256$ levels

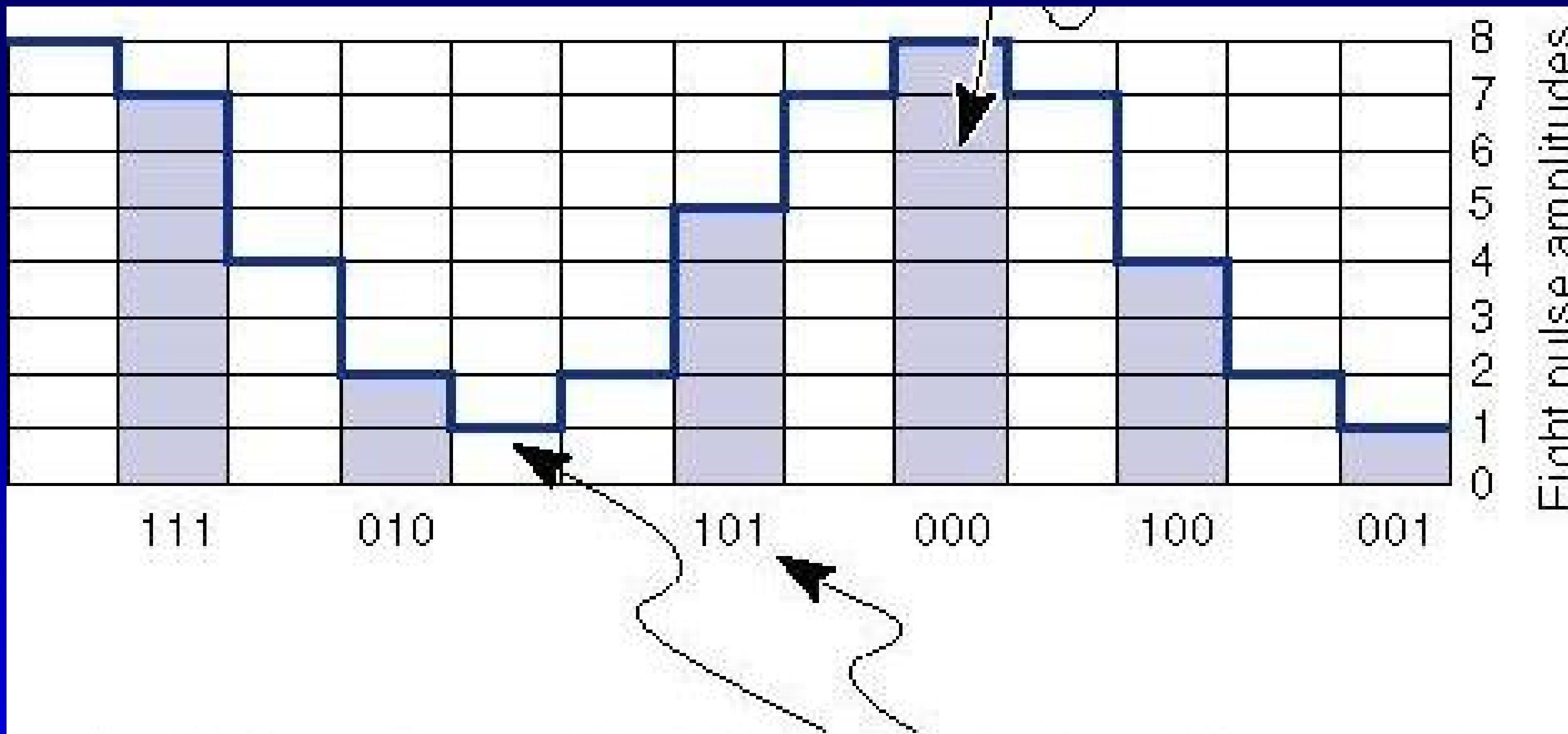
Summary

The signal (original wave) is quantized into 128 pulse amplitudes (PAM). In this example we have used only eight pulse amplitudes for simplicity. These eight amplitudes can be depicted by using only a 3-bit code instead of the 8-bit code normally used to encode each pulse amplitude.



After quantizing, samples are taken at specific points to produce amplitude modulated pulses. These pulses are then coded. Because we used eight pulse levels, we only need three binary positions to code each pulse.[1] If we had used 128 pulse amplitudes, then a 7-bit code plus one parity bit would be required.

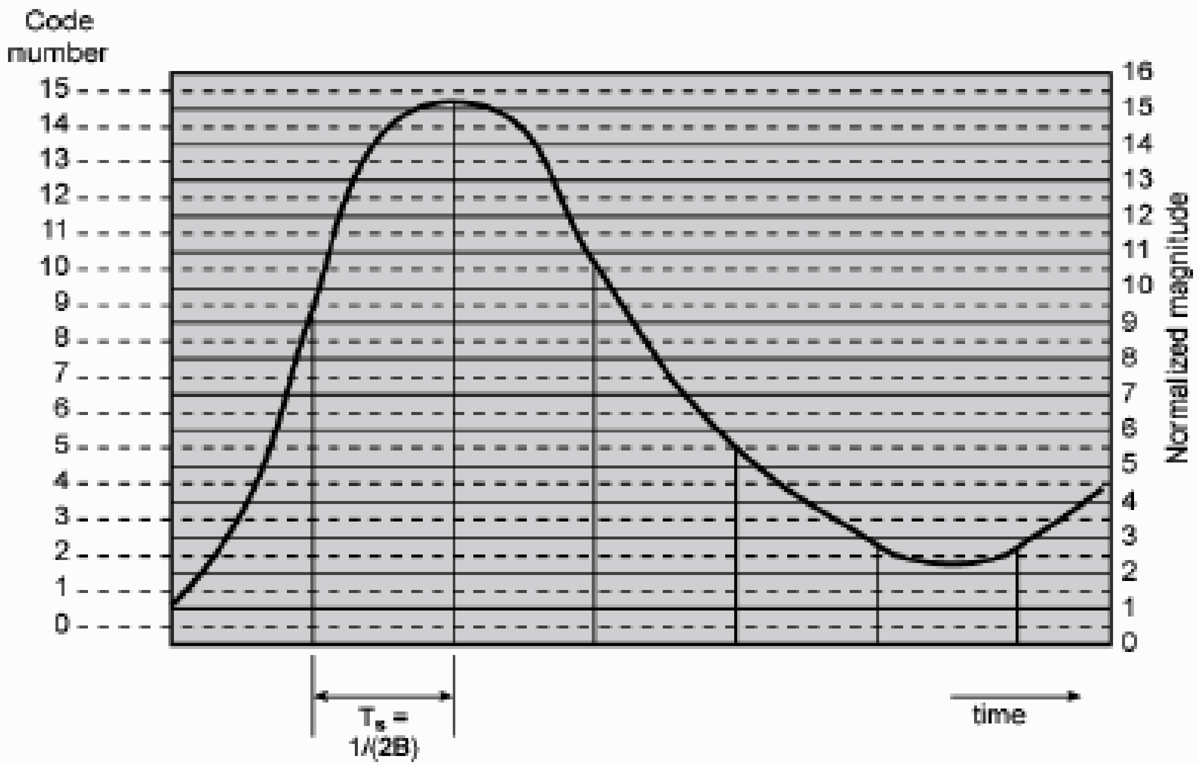




001 = PAM level 1
010 = PAM level 2
011 = PAM level 3
100 = PAM level 4
101 = PAM level 5
110 = PAM level 6
111 = PAM level 7
000 = PAM level 8

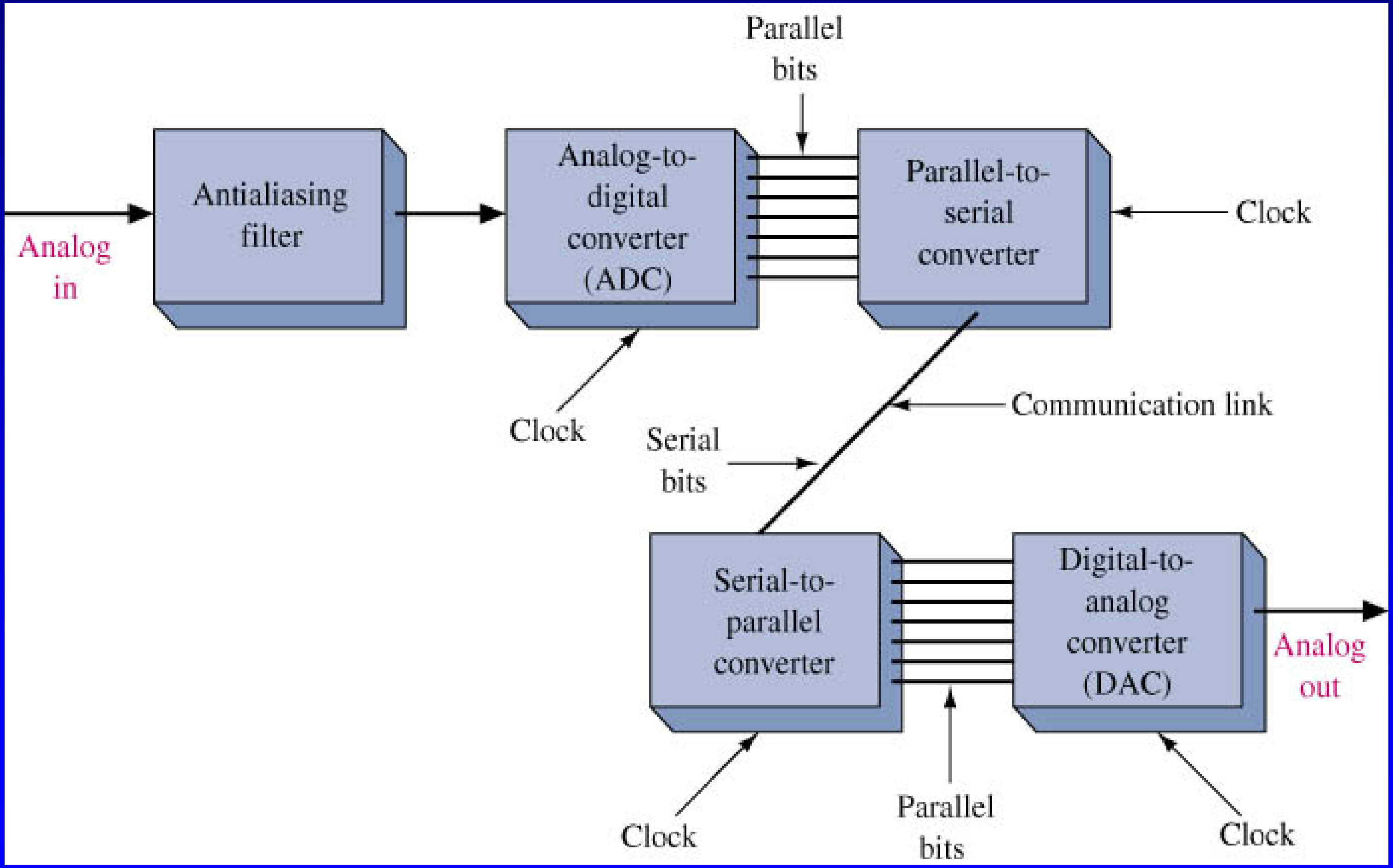
For digitizing a voice signal, 8,000 samples per second are taken. These 8,000 samples are then transmitted as a serial stream of 0s and 1s. In our case 8,000 samples times 3 bits per sample would require a 24,000 bps transmission rate. In reality, 8 bits per sample times 8,000 samples requires a 64,000 bps transmission rate.

PCM Example



PAM value	1.1	9.2	15.2	10.8	5.6	2.8	2.7
quantized code number	1	9	15	10	5	2	2
PCM code	0001	1001	1111	1010	0101	0010	0010

PCM Communications



Speech Telephony

- Bandwidth 300 – 3400 Hz
- Filtered to 4 kHz
- Sampled at 8 kHz
- 8 bit A-Law or μ-Law to give
- 64 kbit/s data rate