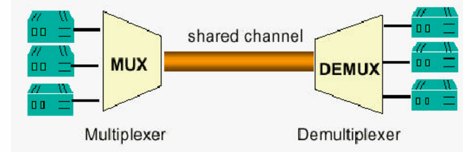
PULSE CODE MODULATION

Multiplexing

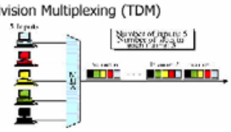


A common application is in long-haul communications

- **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

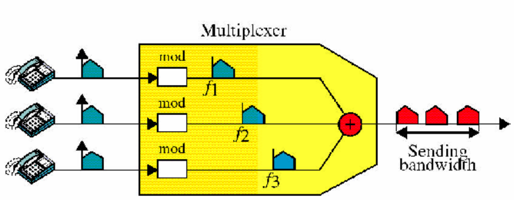
- **Multiplexer** combines data from the *n* input lines and transmits over the high-capacity data link
- **Demultiplexer** accepts multiplexed data stream, separates data according to channel, and delivers them to the appropriate output lines.
- Types of multiplexing
  - Frequency Division Multiplexing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



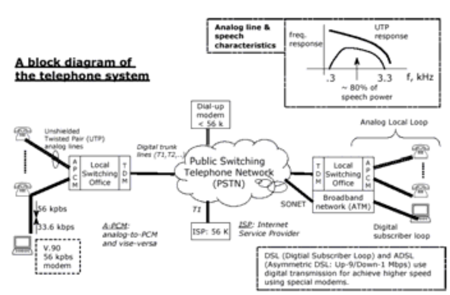
Frequency Division Multiplexing

- 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 coding is also applied to other operations e.g.:

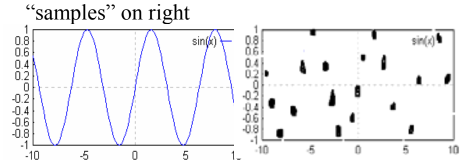
- *Line coding* modifies the analogue or digital source to make it best suited for transmission (e.g HDB3).
- *Channel coding* adds redundancy to improve detection / correction of errors
- *Source coding* typically achieves bit rate reduction by analysing signal structure (e.g. MPEG/JPEG).

PCM - Three Steps

- Sampling
- Quantising
- Encoding

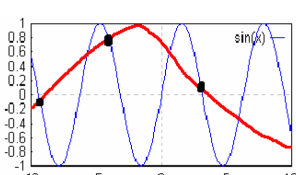
Sampling of a continuous signal

- 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*

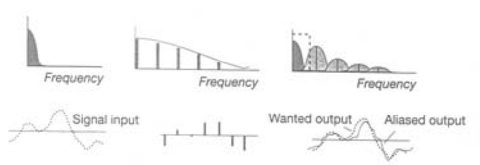
Nyquist Sampling Theorem

- 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



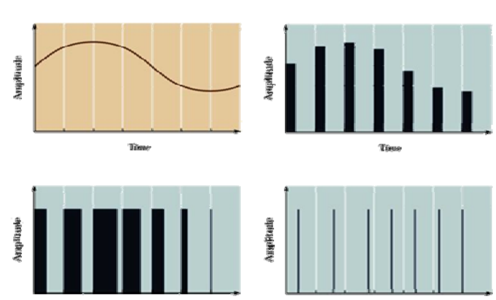
Practical Sampling

- 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

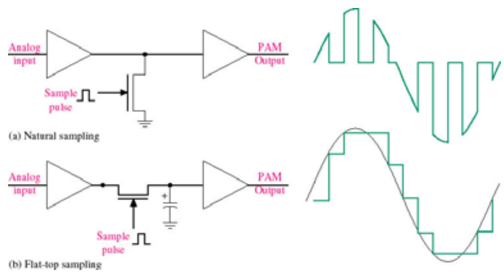
Sampling

- 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

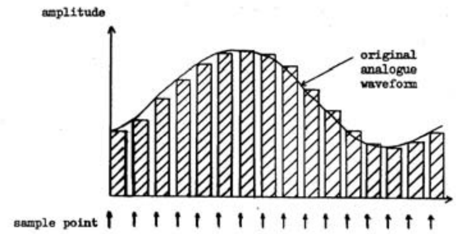
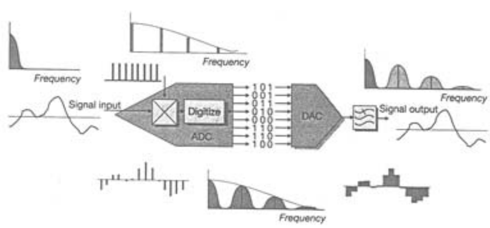


Figure 2: Sample and Hold Pulses

Nyquist Sampling System



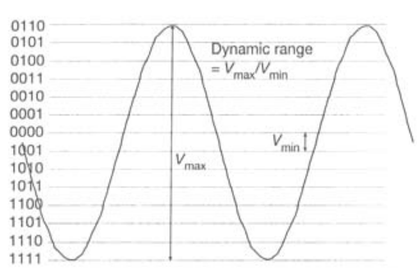
Quantising

- 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

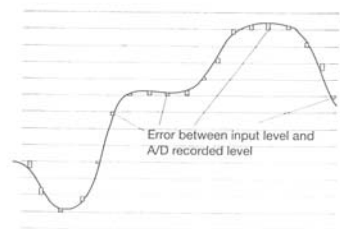
- Important that A/D converter can deal with both large and small signals.
- Ratio of V_{max} to V_{min} over which converter will operate is its Dynamic Range
- Depends on the number of bits the converter uses
- More bits means more *quantisation* levels

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

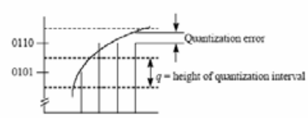
[$20\log V_{max}/V_{min} = 20\log 2^n = n \times 20\log 2$]

Quantisation Noise…1



Quantisation Noise

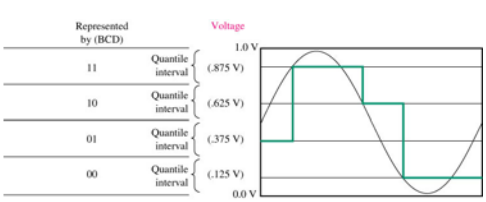
- Assume:
  - Analog signal is equally likely to fall anywhere in the interval, so quantization errors are uniformly distributed (between $-q/2$ and $q/2$)
  - Errors in adjacent samples are independent (generally true if signal is not oversampled)
- Then the quantization errors are random signals perturbations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



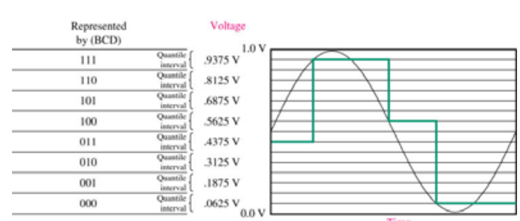
Quantisation Noise…2

- 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

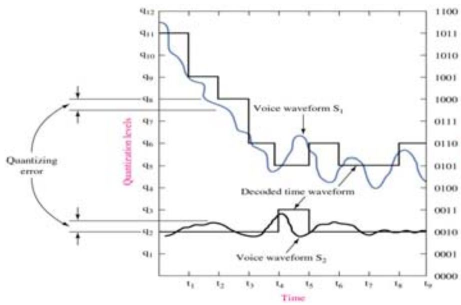
2 Bit A-D Converter



3 Bit A-D Converter

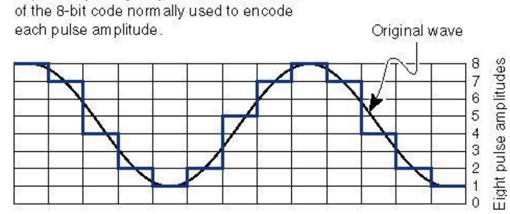


4 bit Analog Digital PCM Quantisation

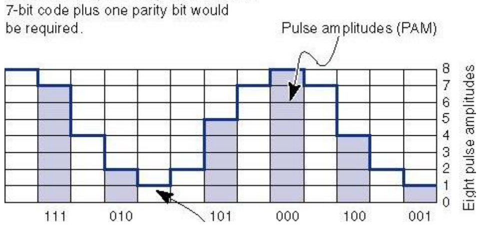


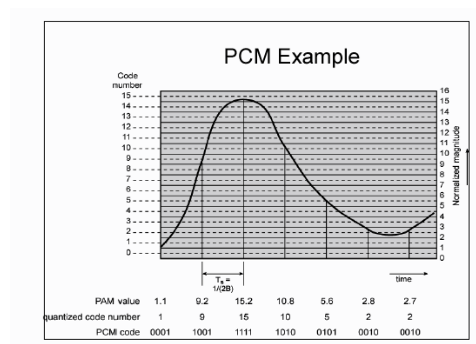
Pulse Code Modulation

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.





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)

Companding

– A technique for reducing the number of bits while achieving an equivalent dynamic range or signal to quantisation noise level

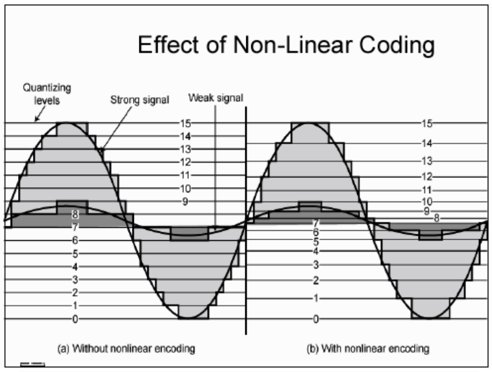
– COMPressing and expANDING

– Decrease quantisation step size for small signals and increase for large signals

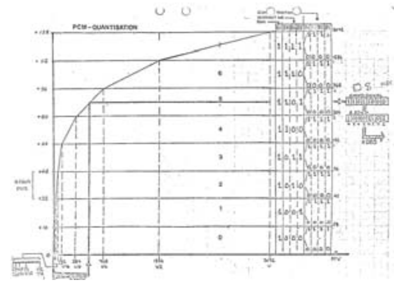
– International standards for telephony

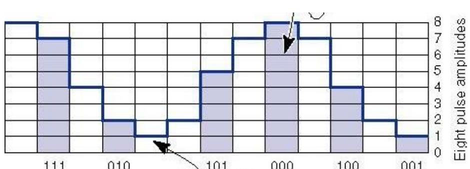
» A Law – European

» µ Law – USA



Non Uniform Quantisation





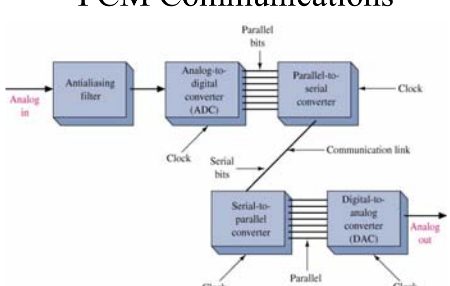
001 = PAM level 1
010 = PAM level 2
011 = PAM level 3
100 = PAM level 4
101 = PAM level 5
110 = PAM level 6
111 = PAM level 7
000 = PAM level 8

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.

Band limiting PAM signals

- where $Tx = T/n$ for PAM TDM system with sampling period T & n signals. To satisfy the sampling Theorem the required bandwidth Bx will be
  - $Bx > 1/(2Tx)$
- The sampling theorem assumes that an ideal low pass filter is available to satisfy $Bx = 1/(2Tx)$
- This will only work if careful consideration is given to sampling instant at demodulator

PCM Communications

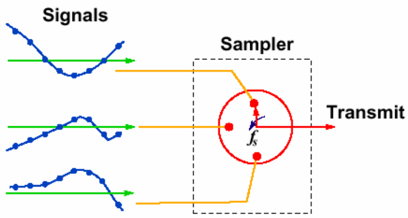


Speech Telephony

- 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

Time Division Multiplexing

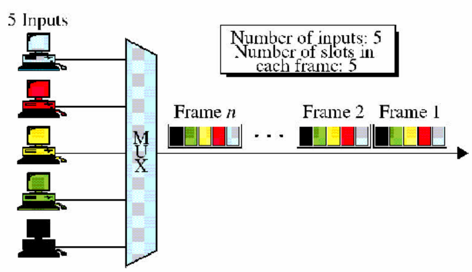
Definition 1 *Time-division multiplexing (TDM) is the time interleaving of samples from several sources so that the information from these sources can be transmitted serially over a singe communications channel.*



Time-Division Multiplexing

- In TDM, each information signal is allowed to use all available bandwidth
- In theory, it is possible to divide the bandwidth or the time among the users of a channel
- Continuously variable signals, such as analog, are not well adapted to TDM because the signal is present all the time

TDM – view 1

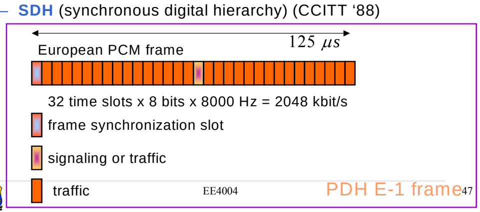


TDM in Telephony

- TDM is used extensively in telephony
- The most common US standard is the DS-1 signal, which consists of 24 PCM voice channels, multiplexed using TDM
- Each channel is sampled at 8 kHz with 8 bits per sample, which gives a bit rate of 64 kb/s for each voice channel
- The samples must be transmitted at the rate they were obtained to be reconstructed
- The overall bit rate is 1.544 Mb/s
- The whole system is known as a *T1 Carrier*

PCM systems and digital time division multiplexing (TDM)

- In digital multiplexing several messages are transmitted via same physical channel. For multiplexing 64 kbit/s channels in digital exchanges following three methods are available:
  - **PDH** (plesiochronous digital hierarchy) (the dominant method today, E1 & T1) ('50-'60, G.702)
  - **SONET** (synchronous optical network) ('85)
  - **SDH** (synchronous digital hierarchy) (CCITT '88)

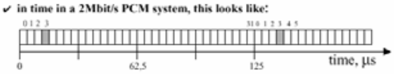


PCM-hierarchy

- ✔ PCM-hierarchy is created by overlapping time division multiplexed signal connections byte by byte (sample by sample). Bits become shorter.
- ✔ The basic speed in the hierarchy is the bitrate of a single voice channel

S=8000Hz* 8bit = 64kbit/s,

- ✔ in time in a 2Mbit/s PCM system, this looks like:



- ✔ The following voice channel groups are defined
  - 30 voice channels
  - 120 voice channels
  - 480 voice channels
  - 1920 voice channels

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PCM 30 (E1)

- ✔ The most common information switching and transmission format in the telecommunication network is PCM 30.
- ✔ PCM 30 contains:
  - 1 synchronization and management channel
  - 1 signaling channel
  - 30 voice channel
- ✔ A channel is a time slot in the PCM-frame (125µs), created by TD multiplexing.
- ✔ PCM 30 system carries 32 time slots, each 64kbit/s. This gives a total bit rate of 2048kbit/s.

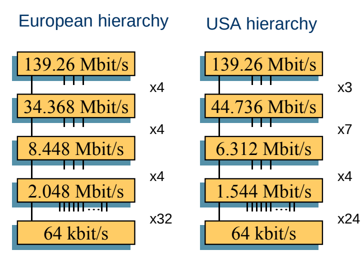
PCM 30 frame

- ✔ PCM 30 -frame contains 32 time slots
  - time slot 0 is dedicated for synchronization and management information
  - Time slot 16 is assigned for signaling information (CAS)
  - Time slots 1-15 and 17-32 are voice or user information channels
- ✔ Even and odd frame structures differ
  - In even numbered frames time slot 0 carries the frame alignment signal (C0 01 10 11). C is the CRC-bit (cyclic redundancy check) for ensuring the frame alignment recovery in case someone is sending X0 01 10 11 on a user information channel – this addition was forced by ISDN which supports transparent 64kbit/s service for data transfer.
  - Time slot 0 in odd frames carries alarm information. To avoid wrong frame alignment, the second bit in tsl 0 is set to the constant value of 1.

US Digital Signal Hierarchy

Carrier	Signal	Voice Channels	Bit Rate (Mb/s)	Typical Medium
T1	DS-1	24	1.544	Twisted-pair
T1C	DS-1C	48	3.152	Twisted-pair
T2	DS-2	96	6.312	Low-capacitance twisted-pair microwave
T3	DS-3	672	44.736	Coax, microwave
T4	DS-4	4032	274.176	Coax, fiber-optic
T5	DS-5	8064	560.16	Fiber optics

PCM hierarchy in PDH



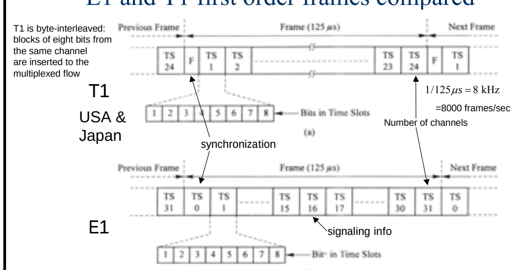
If one wishes to disassemble a tributary from the main flow the main flow must be demultiplexed step by step to the desired main flow level in PDH.

T1 and E1 summarized

- In PSTN two PCM systems dominate:
  - T1, developed by Bell Laboratories, used in USA & Japan
  - E1, developed by CEPT* used in most of the other countries
- In both data streams divided in frames of 8000 frames/sec
- In T1
  - 24 time-slots and a framing (F) bit serves 24 channels
  - Frame length: 1+ 8x24=193 bits
  - Rate 193x8000 bits/second=**1544 kb/s**
- In E1
  - frame has 32 time-slots, TS 0 holds a synchronization pattern and TS 16 holds signaling information
  - An E1 frame has 32x8=256 bits and its rate us 8000x256=**2048 kb/s**

European Conference of Postal and Telecommunications Administration

E1 and T1 first order frames compared*



NOTE: In T1 one bit in each time slot in every sixth frame is replaced by signaling information yielding 56 kb/s only

John G. van Bosse: Signaling in Telecommunication Networks

Some codecs and their characteristics

Coding Standard	Algorithm	Sample Size (msec)	Rate Kbit/s	Mean Opinion Score	Year
G.711	PCM	0.125	64	4.10	1972
GSM 06.10	RPE-LTP	20.000	13	3.50	1987
G.726,G727	ADPCM	0.125	16, 24, 32, 40	3.85	1990
G.728	LDCELP	0.625	16	3.61	1992, 1994
IS-96	VSELP	20.000	8.5, 4, 2, 0.8		1993
G.729, G.729a	CS-ACELP	10.000	8	3.92, 3.70	1995
G.723.1	MPC-MLQ	10.200	6.3, 5.3	3.90	1995
PDC	PSI-CELP	40.000	3.45		1996
FS-1015	LPC	25.700		2.40	-
AMR-NB					
AMR-WB				>PCM	

Line Codes

- Line codes are methods of converting binary numbers back into analog voltages or currents
- The simplest line code is to use the presence or absence of a voltage/current to indicate the logic state
- Unipolar NRZ (non-return-to-zero) means that there is no requirement for a signal to return to zero at the end of each element
- RZ (return-to-zero) methods are used to eliminate low-frequency ac components and dc components

Line coding

- Line codes are used to **enable baseband transmission** in
  - Fiber optic systems
  - Cable transmission
  - Data access and storage
- In PCM-links line coding is used to alleviate clock **synchronization** at the receiver (F-transform of the pulse train should contain spikes that the receiver clock can be synchronized)
- Line codes should
  - be immune to **long strings of zeros** that can lead to missing receiver clock synchronization
  - contain zero long term averaged **DC-component**
  - have minimum **bandwidth**
- Line codes can also be used for **error detection**

Line coding (cont.)

Unipolar [0,A] RZ and NRZ

Polar [-A/2,A/2] RZ and NRZ

Bipolar [-A/2,0,A/2] AMI

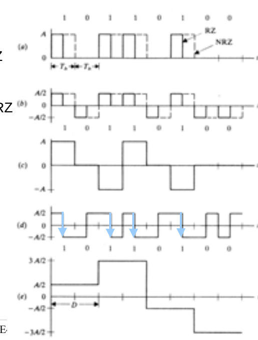
Split-Phase Manchester

Code rate reduced by *n*

Split-Polar quaternary NRZ

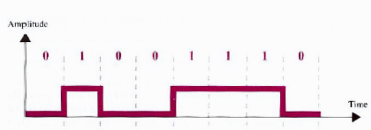
$M = 2^n$

$r_{be} = r/n = r/\log_2 M$



3.2.1.1. Unipolar Encoding

- Unipolar encoding:
  - Uses only one level of amplitude.
  - Is very simple and very primitive.
  - Almost obsolete today.

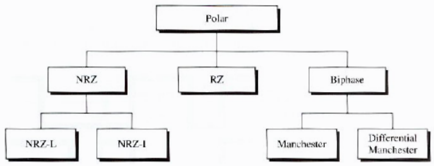


3.2.1.1. Unipolar Encoding

- Two problems:
  - DC component
    - Average amplitude of encoded signal is nonzero.
    - Can not travel through media that can not handle DC components.
  - Synchronization
    - Lack of synchronization between sender and receiver clock can distort the timing of the signal.
    - Results in misinterpretation of the number of 1s or 0s whenever the data stream include a long uninterrupted series of 1s or 0s.

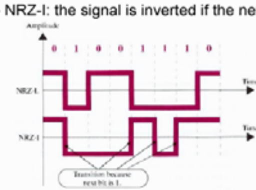
3.2.1.2. Polar Encoding

- Polar encoding:
  - Uses two levels (positive and negative) of amplitude.
  - Reduces the DC component problem.
  - Three popular variations:



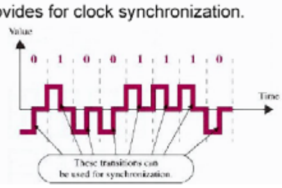
3.2.1.2. Polar Encoding

- Nonreturn to zero (NRZ) encoding:
  - The signal level is always either positive or negative.
  - Two methods:
    - NRZ-L: positive for 0 and negative for 1.
    - NRZ-I: the signal is inverted if the next bit is 1.



3.2.1.2. Polar Encoding

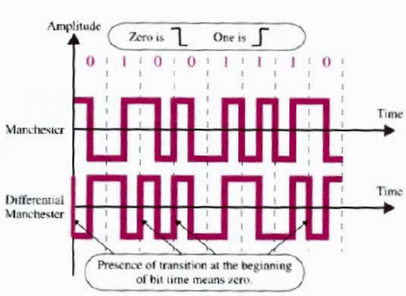
- Return to zero (RZ) encoding:
  - Uses three values: positive, negative, and zero.
  - Signal changes not between bits, but during each bit.
  - Provides for clock synchronization.



3.2.1.2. Polar Encoding

- Biphase encoding:
  - Signal changes at the middle of bit interval but does not return to zero.
  - Provides for clock synchronization.
  - Two possible implementations:
    - Machester:
      - the transition at the middle of the bit is used for both synchronization and bit representation.
    - Differential Manchester:
      - the transition at the middle of the bit is used only for synchronization.
      - the bit representation is shown by the inversion or noninversion at the beginning of the bit.

3.2.1.2. Polar Encoding

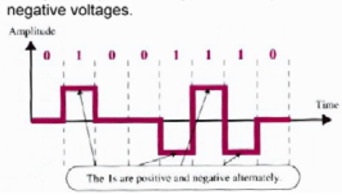


3.2.1.3. Bipolar Encoding

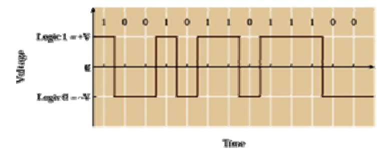
- Bipolar encoding:
  - Uses three voltage levels: positive, negative, and zero.
  - Unlike RZ, the zero level is used to represent binary 0.
  - Provides for synchronization.

3.2.1.3. Bipolar Encoding

- The simplest type of bipolar encoding:
  - Alternate Mark Inversion (AMI):
    - Binary 0 is represented by a neutral, zero voltage.
    - Binary 1 is represented by alternative positive and negative voltages.



Bipolar NRZ Code



Data Compression

- Data compression is a technique used to reduce the bandwidth to transmit an analog signal in a digital form
- The exact bandwidth necessary is dependent upon the modulation scheme

Lossy and Lossless Compression

- There are two main categories of data compression:
  - Lossless compression involves transmitting all of the data in the original signal but using fewer bits. Lossless compression generally looks for redundancies in the data
  - Lossy compression allows for some reduction in the quality of the transmitted signal. Lossy compression involves reducing the number of bits per sample or reducing the sampling rate

Vocoders

- A vocoder (*voice coder)* is an example of lossy compression applied to human speech
- A typical vocoder reduces the amount of data that needs to be transmitted by constructing a model of the human vocal system

Vocoder Types

- There are two main ways of generating the excitation signal in a linear predictive vocoder:
  - Pulse Excited Linear Predictive (PELP)
  - Residual Excited Linear Predictive (RELP)