CSE / T / 315A
Data Communications
Topic 4- Digital Transmission

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Main Topics

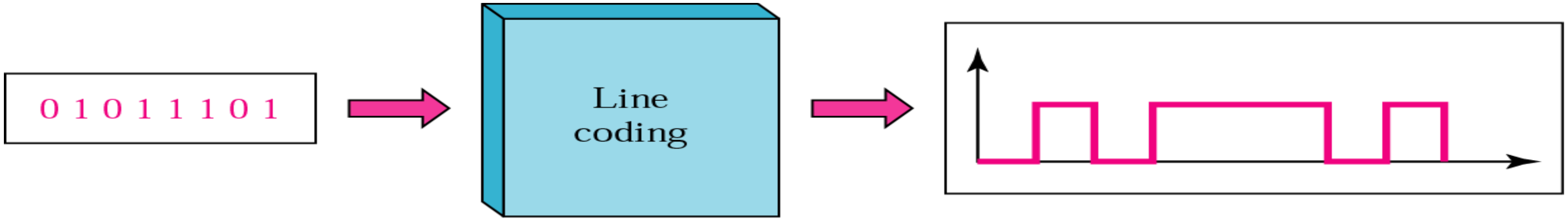
- Issues
- Line coding
- Block Coding
- Scrambling Techniques
- Analog to Digital

Overview

- How to represent digital data by using digital signals.
- The conversion involves three techniques: line coding, block coding, and scrambling.
  - Line coding is always needed;
  - block coding and scrambling may or may not be needed.

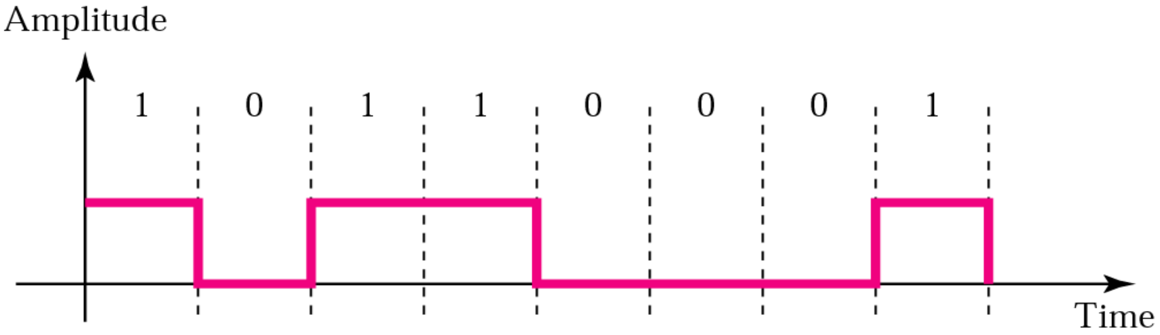
Line Coding

- Converting a string of 1's and 0's (digital data) into a sequence of signals that denote the 1's and 0's.

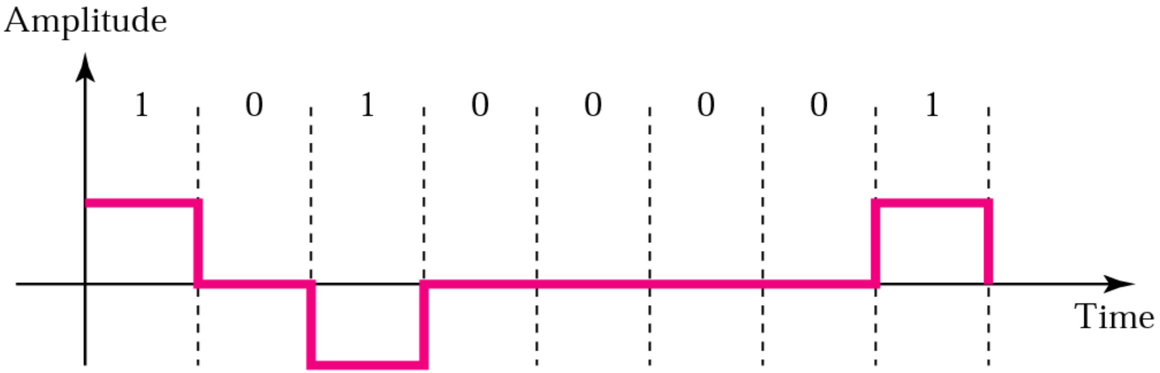


Signal level versus data level

- For example a high voltage level (+V) could represent a “1” and a low voltage level (0 or -V) could represent a “0”.

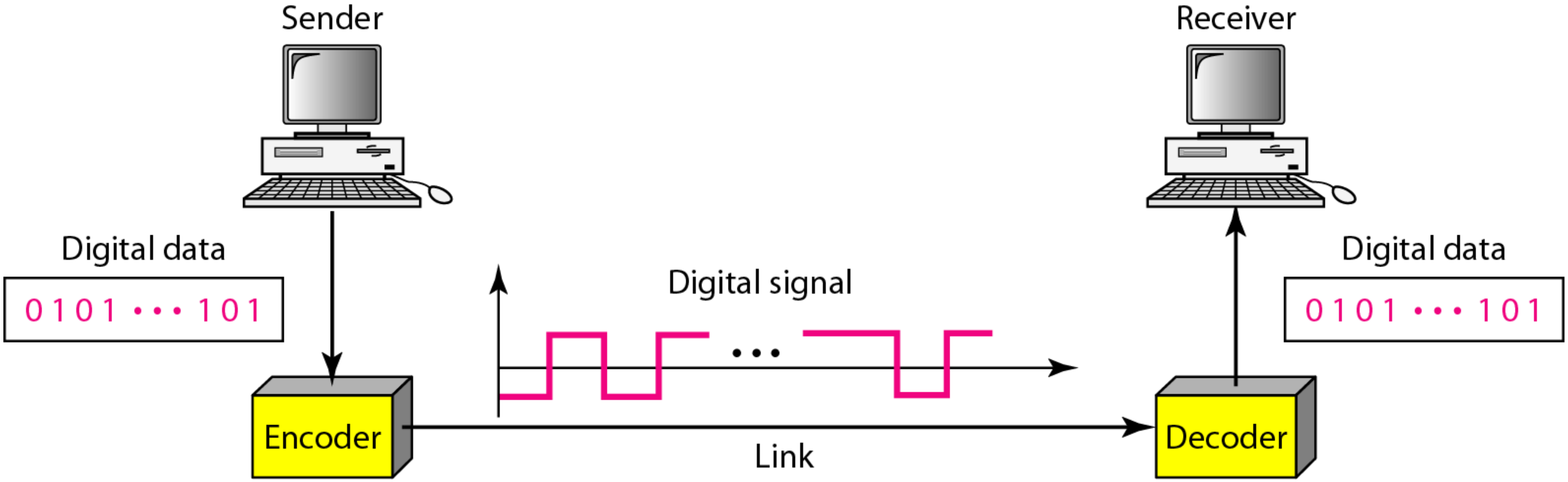


a. Two signal levels, two data levels



b. Three signal levels, three data levels

End-to-End System:
Line coding and Decoding



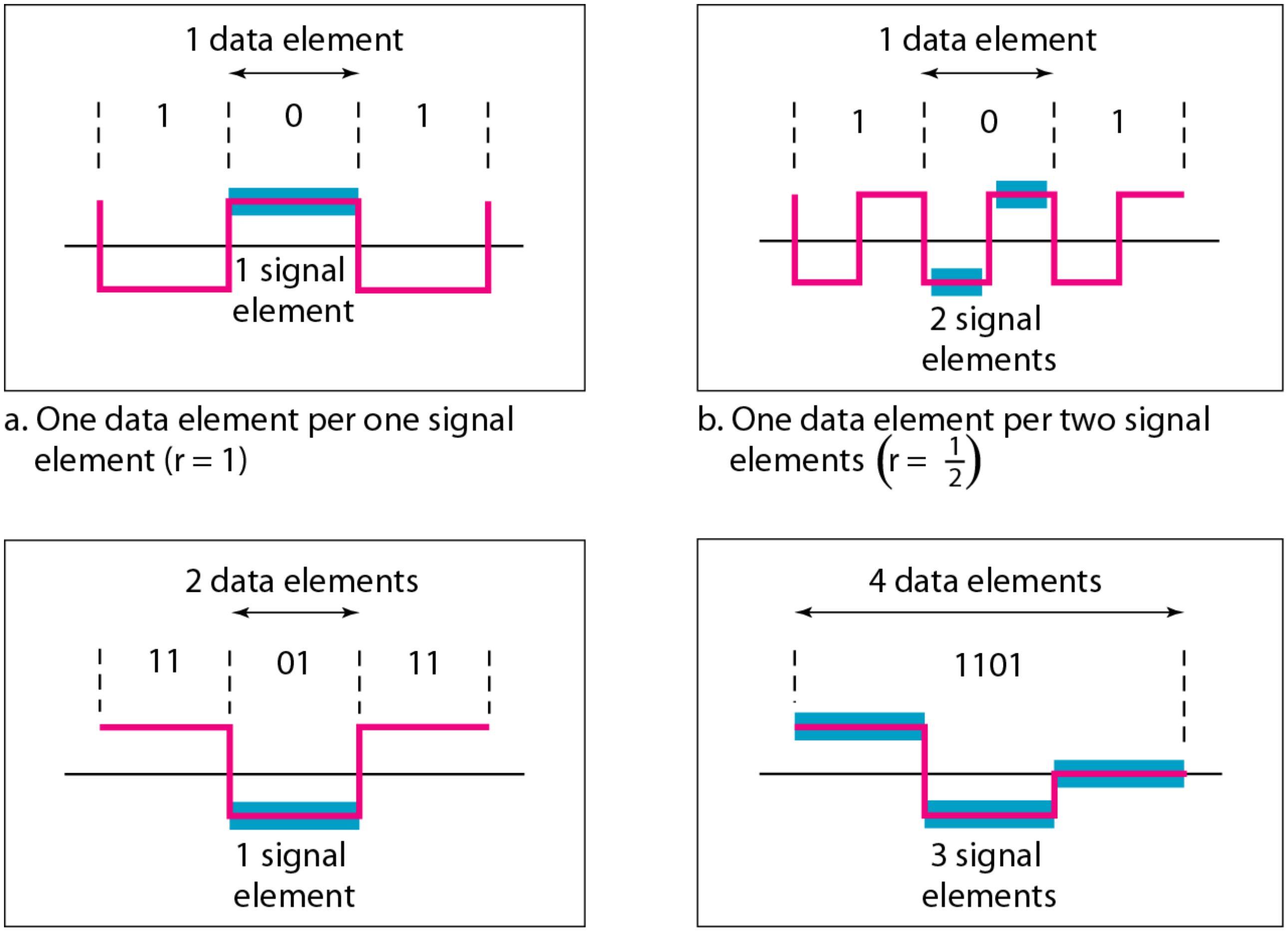
Mapping Data symbols onto Signal levels

- A data symbol (or element) can consist of a number of data bits:
  - 1 , 0 or
  - 11, 10, 01, ……
- A data symbol can be coded into a single signal element or multiple signal elements
  - 1 -> +V, 0 -> -V
  - 1 -> +V and -V, 0 -> -V and +V
- The ratio ‘r’ is the number of data elements carried by a signal element.

Relationship between data rate and signal rate

- The data rate defines the number of bits sent per sec - bps. It is often referred to the bit rate.
- The signal rate is the number of signal elements sent in a second and is measured in bauds. It is also referred to as the modulation rate.
- Goal is to increase the data rate whilst reducing the baud rate.

Signal element versus data element



a. One data element per one signal element (r = 1)

b. One data element per two signal elements $\left(r = \frac{1}{2}\right)$

c. Two data elements per one signal element (r = 2)

d. Four data elements per three signal elements $\left(r = \frac{4}{3}\right)$

Data rate and Baud rate

- The baud or signal rate can be expressed as:

S = c x N x 1/r bauds

where N is data rate

c is the case factor (worst, best & avg.)

r is the ratio between data element & signal element

Problem

- A signal is carrying data in which one data element is encoded as one signal element (r = 1). If the bit rate is 100 kbps, what is the average value of the baud rate if c is between 0 and 1?

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  - We assume that the average value of c is 1/2 . The baud rate is then

$$S = c \times N \times \frac{1}{r} = \frac{1}{2} \times 100{,}000 \times \frac{1}{1} = 50{,}000 = 50 \text{ kbaud}$$

Is there any relation with Nyquist Formula?

- The maximum data rate of a channel is

  $N_{max} = 2 \times B \times \log_2 L$ (defined by the Nyquist formula). Does this agree with the previous formula for N_{max}?

Relation with Nyquist Formula

- The maximum data rate of a channel is

  N_{max} = 2 × B × $\log_2$ L (defined by the Nyquist formula). Does this agree with the previous formula for N_{max}?

  – A signal with L levels actually can carry $\log_2$ L bits per level. If each level corresponds to one signal element and we assume the average case (c = 1/2), then we have

$$\boxed{N_{\max} = \frac{1}{c} \times B \times r = 2 \times B \times \log_2 L}$$

Note

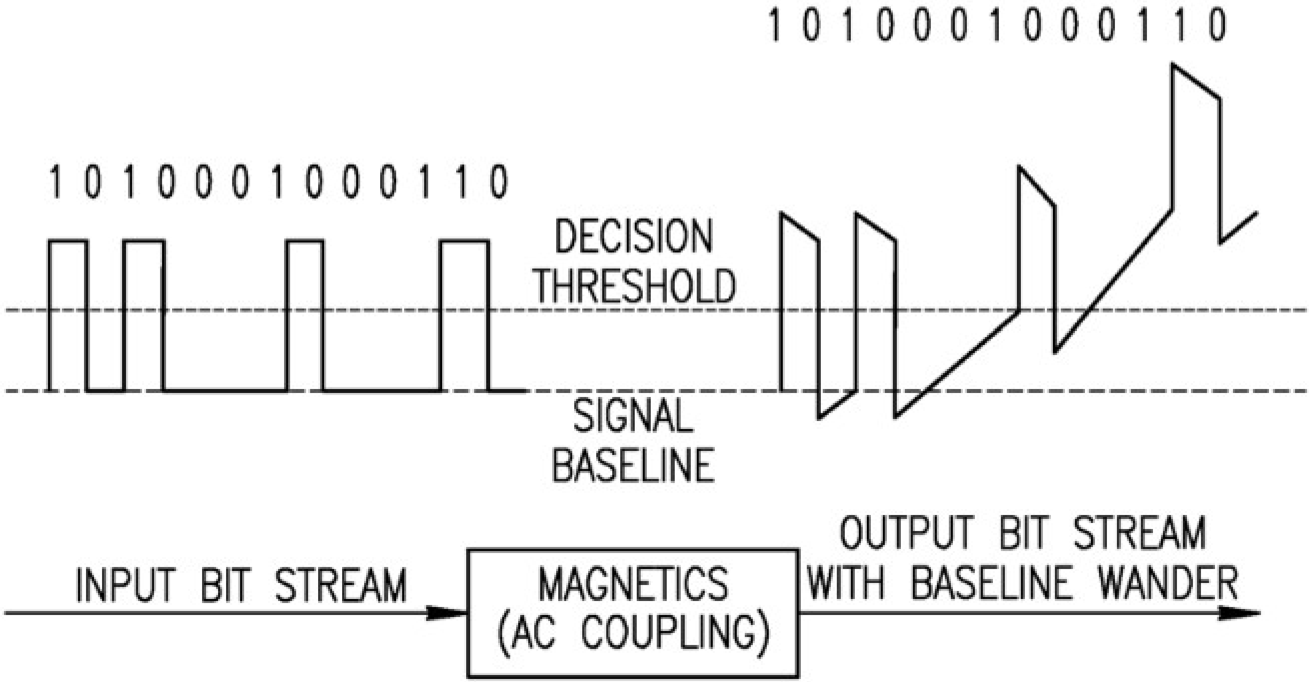
Although the actual bandwidth of a digital signal is infinite, the effective bandwidth is finite.

- Considerations for choosing a good signal element referred to as line encoding

Issues

- Baseline wandering
  - a receiver will evaluate the average power of the received signal (called the baseline) and use that to determine the value of the incoming data elements.
  - If the incoming signal does not vary over a long period of time, the baseline will drift and thus cause errors in detection of incoming data elements.
- A good line encoding scheme will prevent long runs of fixed amplitude.

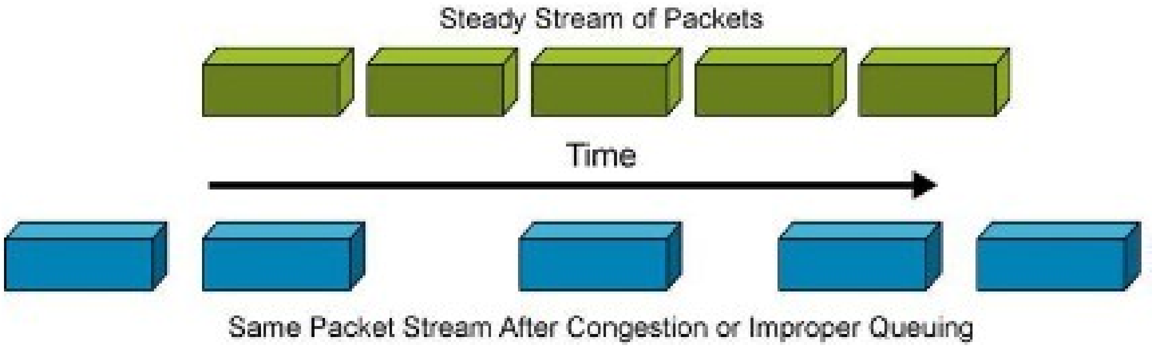
Effects of Baseline Wander



- Baseline wander is a slow variation in the average of a signal waveform. It is caused by attenuation of the low-frequency content of the signal and can result in increased jitter and BER degradation.

Jitter

- **Jitter** is defined as a variation in the delay of received packets. The sending side transmits packets in a continuous stream and spaces them evenly apart. Because of n**etwork** congestion, improper queuing, or configuration errors, the delay between packets can vary instead of remaining constant, as shown in the figure.



BER

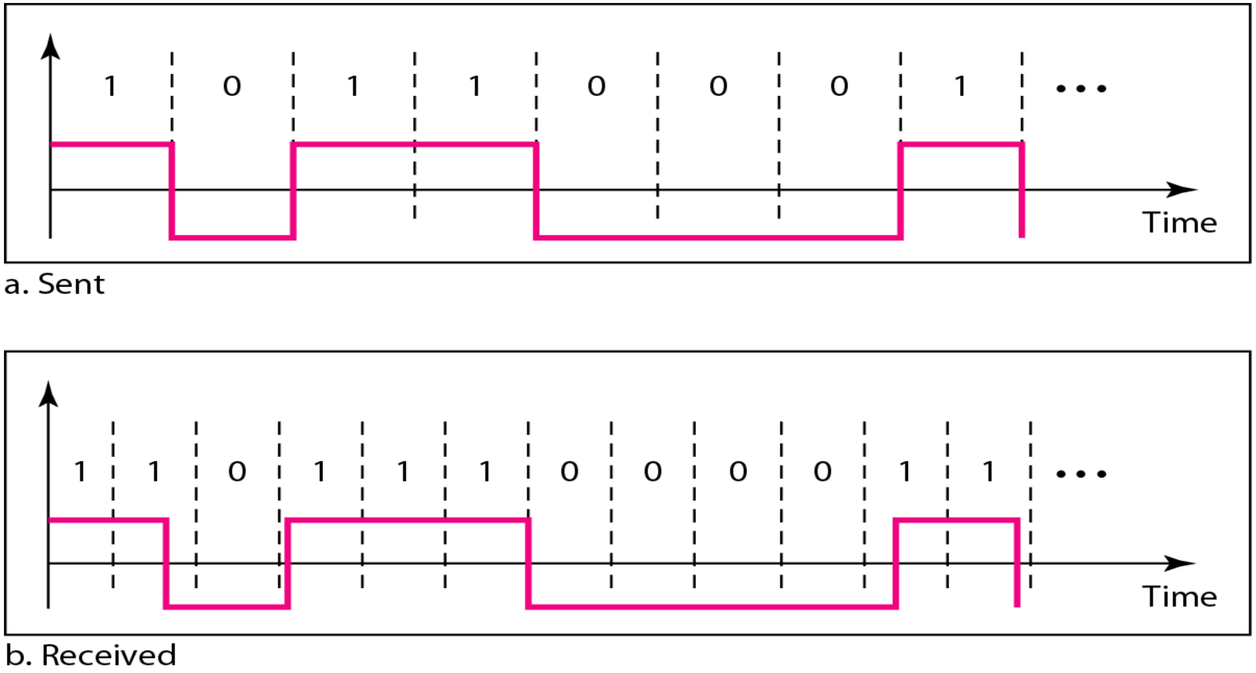
- In a digital transmission, **BER** is the percentage of bits with errors divided by the total number of bits that have been transmitted, received or processed over a given time period. The rate is typically expressed as 10 to the negative power ..i.e., normally expressed in terms of SNR.

Line Encoding- DC Component

- DC components
  - when the voltage level remains constant for long periods of time, there is an increase in the low frequencies of the signal.
    - When the voltage level in a digital signal is constant for a while, the spectrum creates very low frequencies, called DC components, which present problems for a system that cannot pass low frequencies.
  - Most channels are bandpass and may not support the low frequencies.
- This will require the removal of the dc component of a transmitted signal.

Effect of lack of synchronization

- Self synchronization
  - the clocks at the sender and the receiver must have the same bit interval.
- If the receiver clock is faster or slower it will misinterpret the incoming bit stream.



a. Sent

b. Received

- Efficient coding techniques are used to address these issues.

Line Code

- In telecommunication, a **line code** is a code chosen for use within a communications system for transmitting a digital signal down a line.
- Line coding consists of representing the digital signal to be transported, by a waveform that is appropriate for the specific properties of the physical channel (and of the receiving equipment). The pattern of voltage, current or photons used to represent the digital data on a transmission link is called *line encoding*.

Error

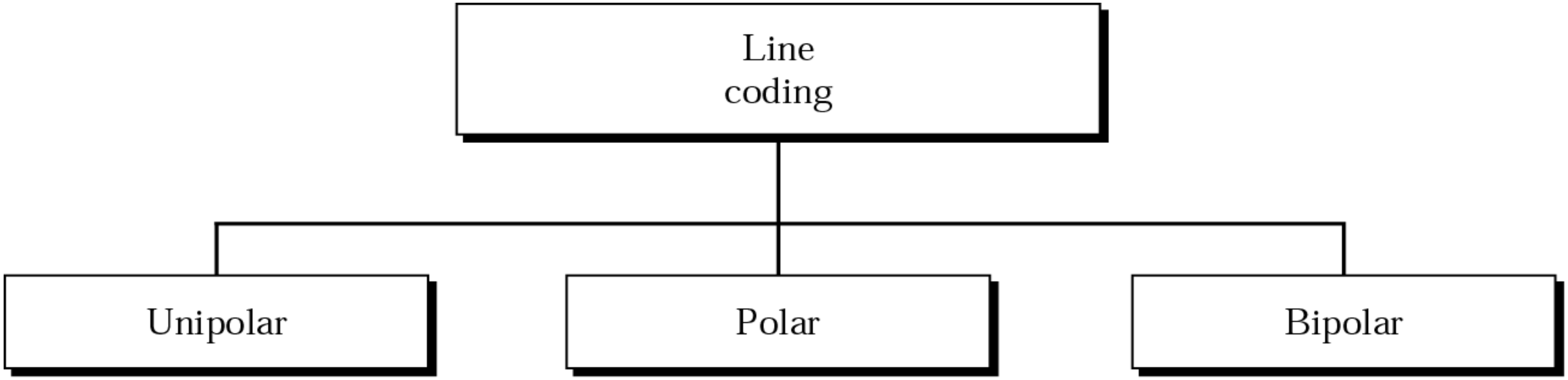
- Error
  - errors occur during transmission due to line impairments.
- Some codes are constructed such that when an error occurs it can be detected.
  - For example: a particular signal transition is not part of the code. When it occurs, the receiver will know that a symbol error has occurred.

Noise and Interference

- Noise and interference
  - there are line encoding techniques that make the transmitted signal “immune” to noise and interference.
  - This means that the signal cannot be corrupted, it is *stronger than error detection*.
- Complexity
  - the more robust and resilient the code, the more complex it is to implement and the price is often paid in *baud rate or required bandwidth*.

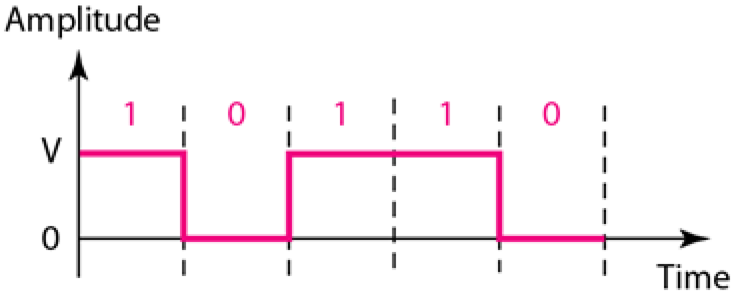
Line coding schemes

- Broad Categories



Unipolar

- Unipolar encoding uses only one polarity of voltage level i.e., only two voltage levels are used as shown in Figure below.



- In this encoding approach, the *bit rate same as data rate*.

Unipolar

- All signal levels are on one side of the time axis
  - either above or below i.e., Unipolar encoding schemes uses single voltage level to represent data. In this case, to represent binary 1 high voltage is transmitted and to represent 0 no voltage is transmitted.

- NRZ (Non Return to Zero) scheme
  - Traditionally, a unipolar scheme is designed as a NRZ scheme in which the +ve voltage defines bit 1 and the zero voltage defines bit 0.
  - The signal level does not return to zero during a symbol transmission. Because there's no rest condition i.e. it either represents 1 or 0.

- What are the issues in Unipolar encoding scheme?

Unipolar

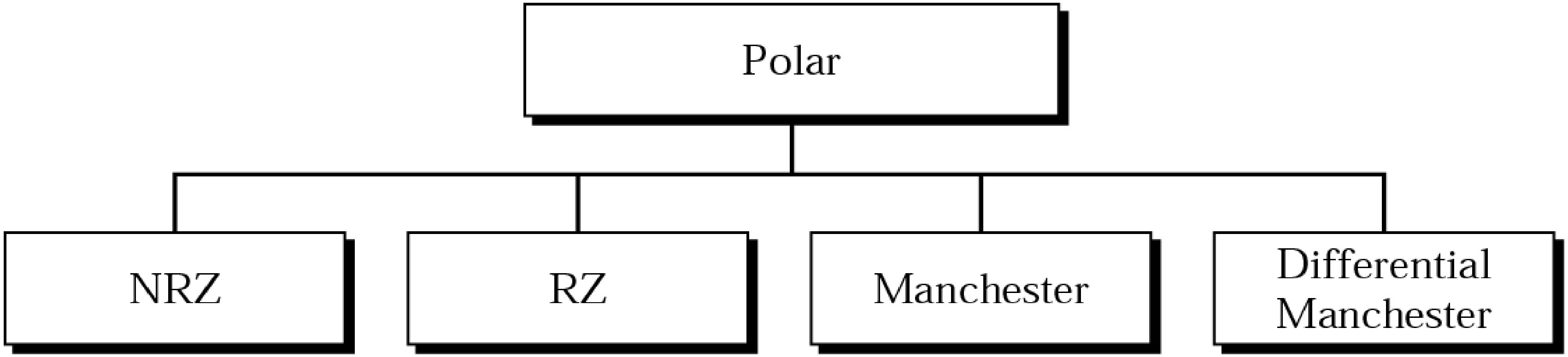
- Scheme is prone to baseline wandering and DC components.

- It has no synchronization or any error detection.

- It is simple but costly in power consumption.

- ***Polar encoding technique uses two voltage levels – one positive and the other one negative.***

Types of Polar Encoding

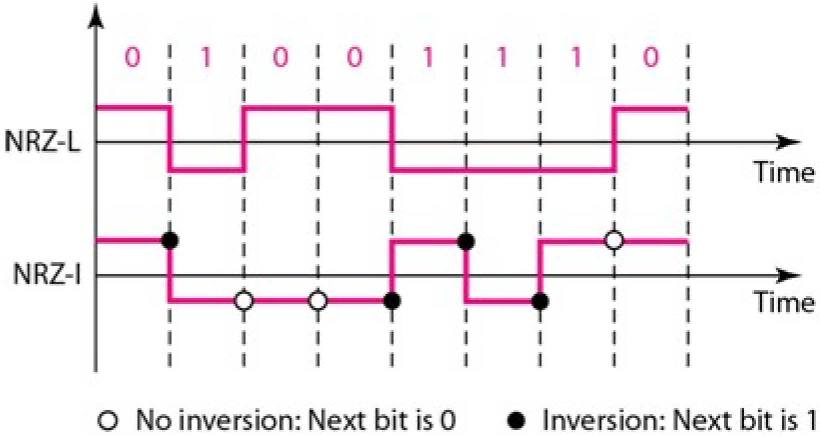


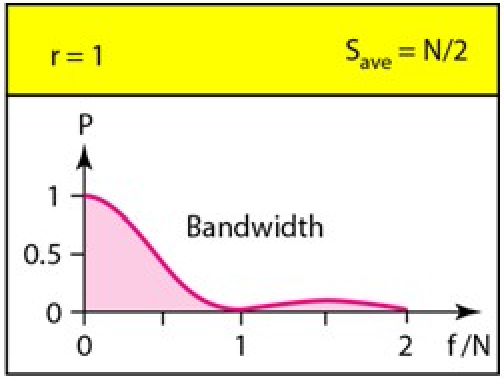
Polar- NRZ

- The voltages are on both sides of the time axis.
- Polar NRZ scheme can be implemented with two voltages. E.g. +V for 0 and -V for 1.
- There are two versions:
  - NZR - Level (NRZ-L)
    - positive voltage for one symbol and negative for the other
  - NRZ - Inversion (NRZ-I)
    - the change or lack of change in polarity determines the value of a symbol. E.g. a “1” symbol inverts the polarity a “0” does not.

NRZ: NRZ-L and NRZ-I

- The data is encoded as the presence or absence of a signal transition at the beginning of the bit time. As shown in the figure below, in NRZ encoding, the signal level remains same throughout the bit-period.





NRZ – L
1 = low level
0 = high level

NRZ – I

- For each 1 in the bit sequence, the signal level is inverted.
- A transition from one voltage level to the other represents a 1.

Note

- In NRZ-L the level of the voltage determines the value of the bit.

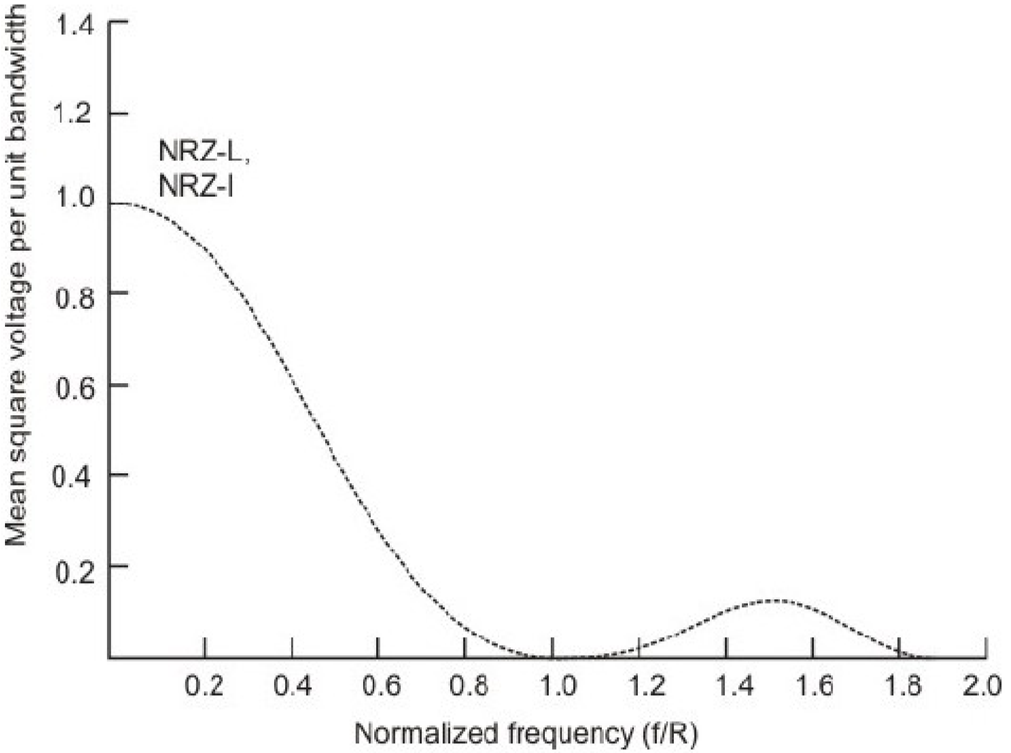
- In NRZ-I the inversion or the lack of inversion determines the value of the bit.

Advantages of NRZ coding

- Detecting a transition in presence of noise is more reliable than to compare a value to a threshold.
- NRZ codes are easy to engineer and it makes efficient use of bandwidth.

Spectrum of NRL-L and NRZ-I

- It may be noted that most of the energy is concentrated between 0 and half the bit rate.
- The main limitations are the presence of a dc component and the lack of synchronization capability. No error detection.
- When there is long sequence of 0’s or 1’s, the receiving side will fail to regenerate the clock and synchronization between the transmitter and receiver clocks will fail.



Note

- NRZ-L and NRZ-I both have an average signal rate of N/2 Bd.

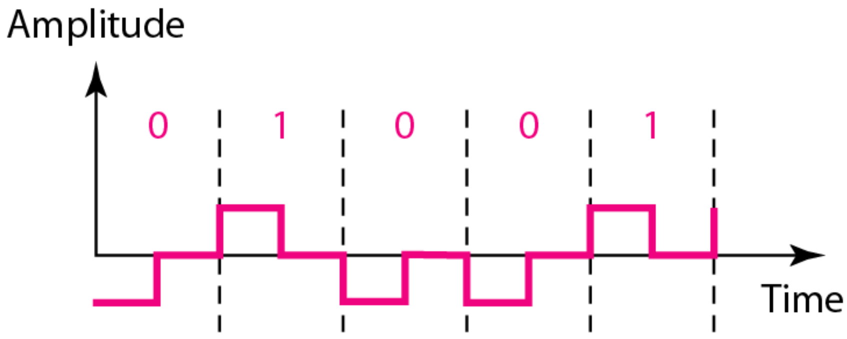
Problem

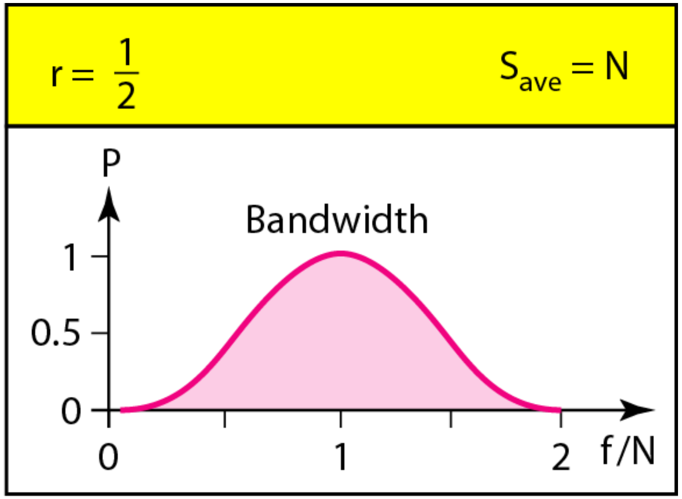
- A system is using NRZ-I to transfer 1-Mbps data. What are the average signal rate and minimum bandwidth?
  - *The average signal rate is S= c x N x 1/r = 1/2 x N x 1 = 500 kbaud. The minimum bandwidth for this average baud rate is Bmin = S = 500 kHz.*

- *Note c = 1/2 for the avg. case as worst case is 1 and best case is 0*

Polar- RZ (Return to Zero)

- The RZ scheme uses three voltage values. (+, 0, -)
- Each symbol has a transition in the middle. Either from high to zero or from low to zero.
- This scheme has more signal transitions (two per symbol) and therefore requires a wider bandwidth.





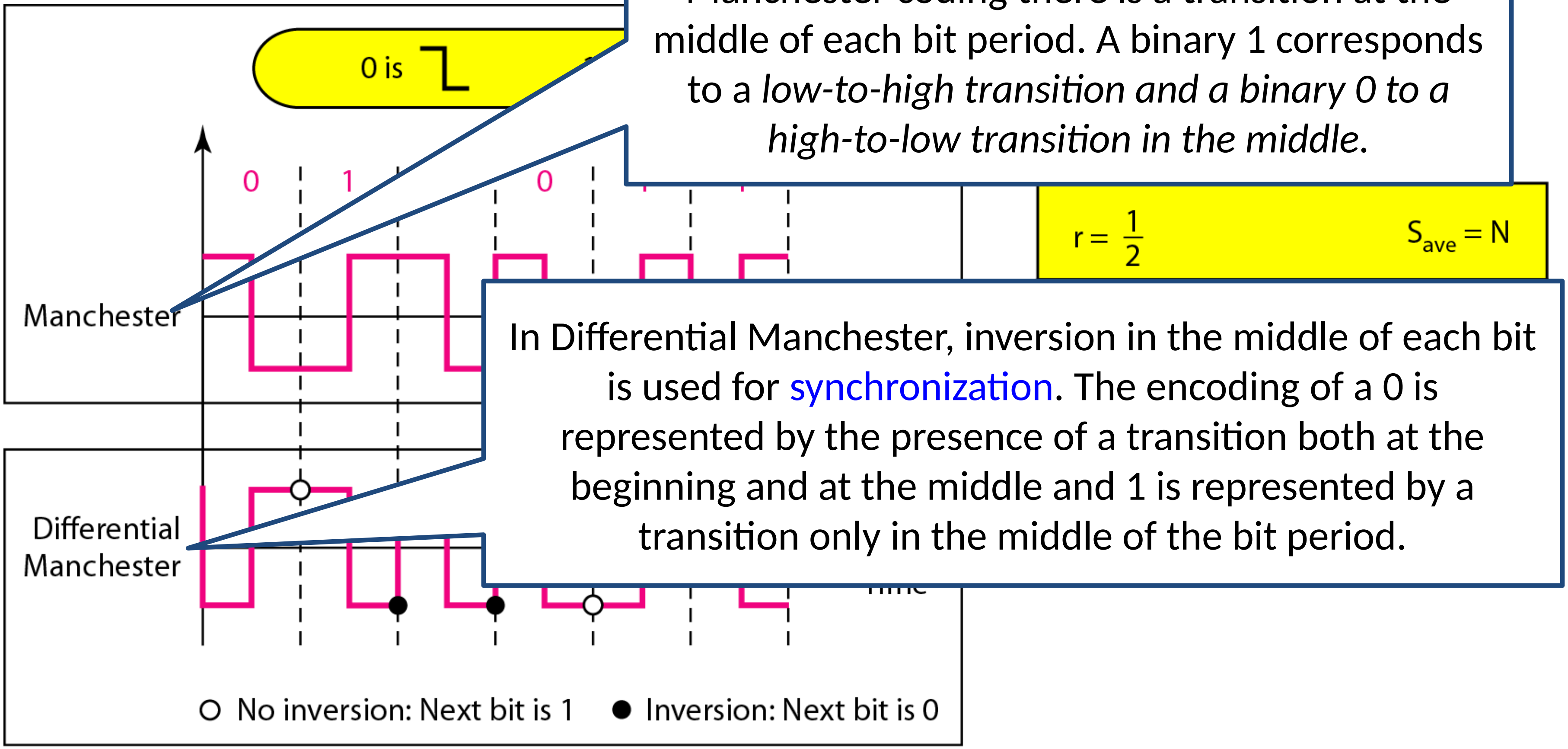
Polar- RZ

- Advantages
  - No DC components or baseline wandering.
  - Self synchronization - transition indicates symbol value.
- Limitations
  - More complex as it uses three voltage level. It has no error detection capability.
  - Increase in bandwidth

Polar-Biphase

- To overcome the limitations of NRZ encoding, biphase encoding techniques can be adopted.
- Manchester coding consists of combining the NRZ-L and RZ schemes.
  - Every symbol has a level transition in the middle: from high to low or low to high. Uses only two voltage levels.
- Differential Manchester coding consists of combining the NRZ-I and RZ schemes.
  - Every symbol has a level transition in the middle. But the level at the beginning of the symbol is determined by the symbol value. One symbol causes a level change the other does not.

Polar biphase: Manchester and differential Manch



In Manchester coding the mid-bit transition serves as a clocking mechanism and also as data. In Manchester coding there is a transition at the middle of each bit period. A binary 1 corresponds to a *low-to-high transition and a binary 0 to a high-to-low transition in the middle.*

In Differential Manchester, inversion in the middle of each bit is used for synchronization. The encoding of a 0 is represented by the presence of a transition both at the beginning and at the middle and 1 is represented by a transition only in the middle of the bit period.

Advantages and Limitations

Advantages

- Two levels
- The is no DC component and no baseline wandering.
- Good synchronization

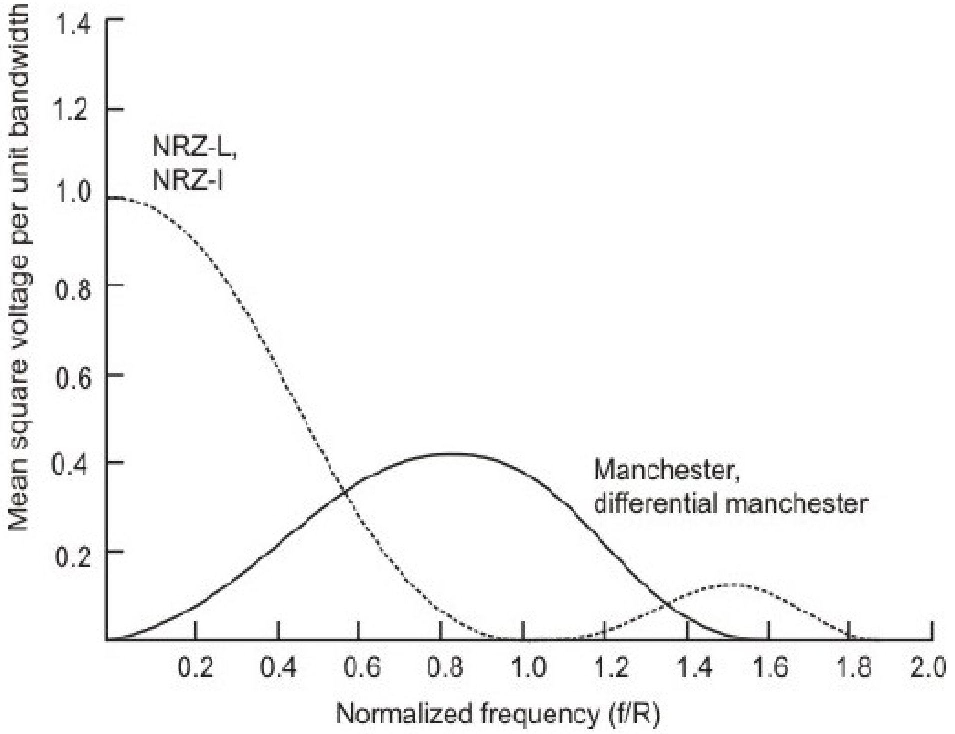
Limitation

- Higher bandwidth due to doubling of bit rate with respect to data rate
  - The minimum bandwidth of Manchester and differential Manchester is 2 times that of NRZ.

- None of these codes has error detection.

Key Features

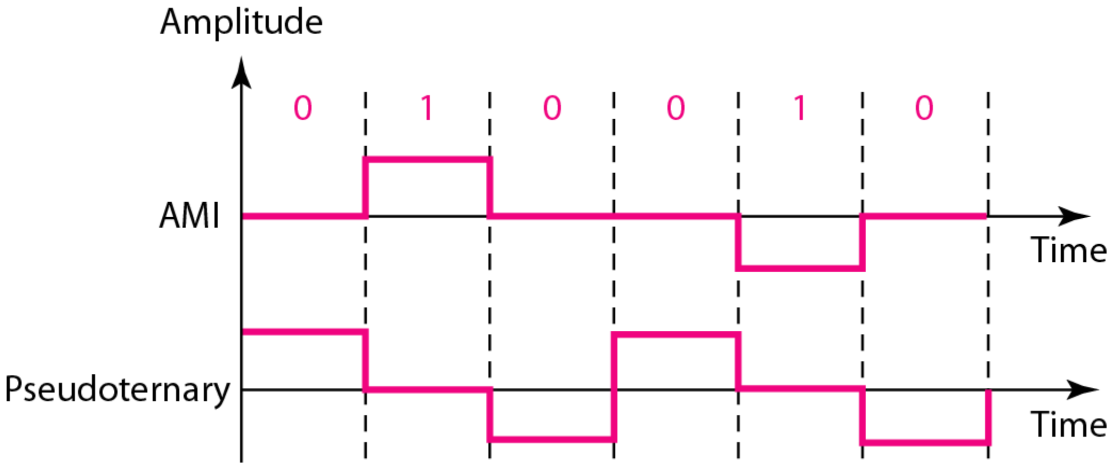
- The bandwidth required for biphase techniques are greater than that of NRZ techniques, but due to the predictable transition during each bit time, the receiver can synchronize properly on that transition.
- Biphase encoded signals have no DC components as shown in Figure.
- A Manchester code is now very popular and has been specified for the IEEE 802.3 standard for base band coaxial cables and twisted pair CSMA/CD bus LANs.

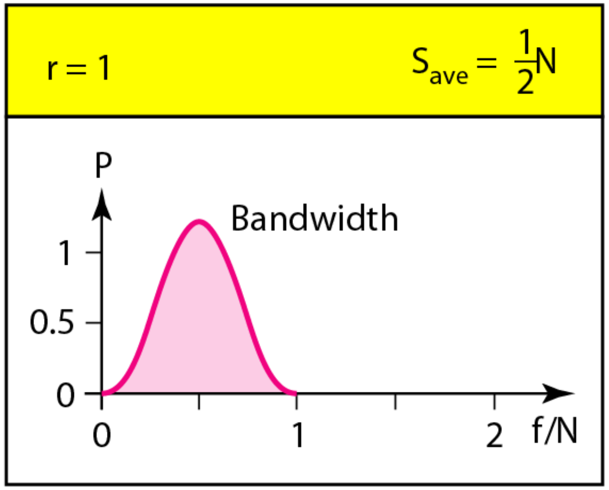


Bipolar – AMI and Pseudoternary

- Code uses 3 voltage levels: (+, 0, -) to represent the symbols (note not transitions to zero as in RZ).
- Voltage level for one symbol is at "0" and the other alternates between + & -.
- Bipolar Alternate Mark Inversion (AMI) - the "0" symbol is represented by zero voltage and the "1" symbol alternates between +V and -V.
- Pseudoternary is the reverse of AMI. Alternating +ve and –ve pulses occur for binary 0 instead of binary 1.

Bipolar schemes:
AMI and Pseudoternary





Bipolar

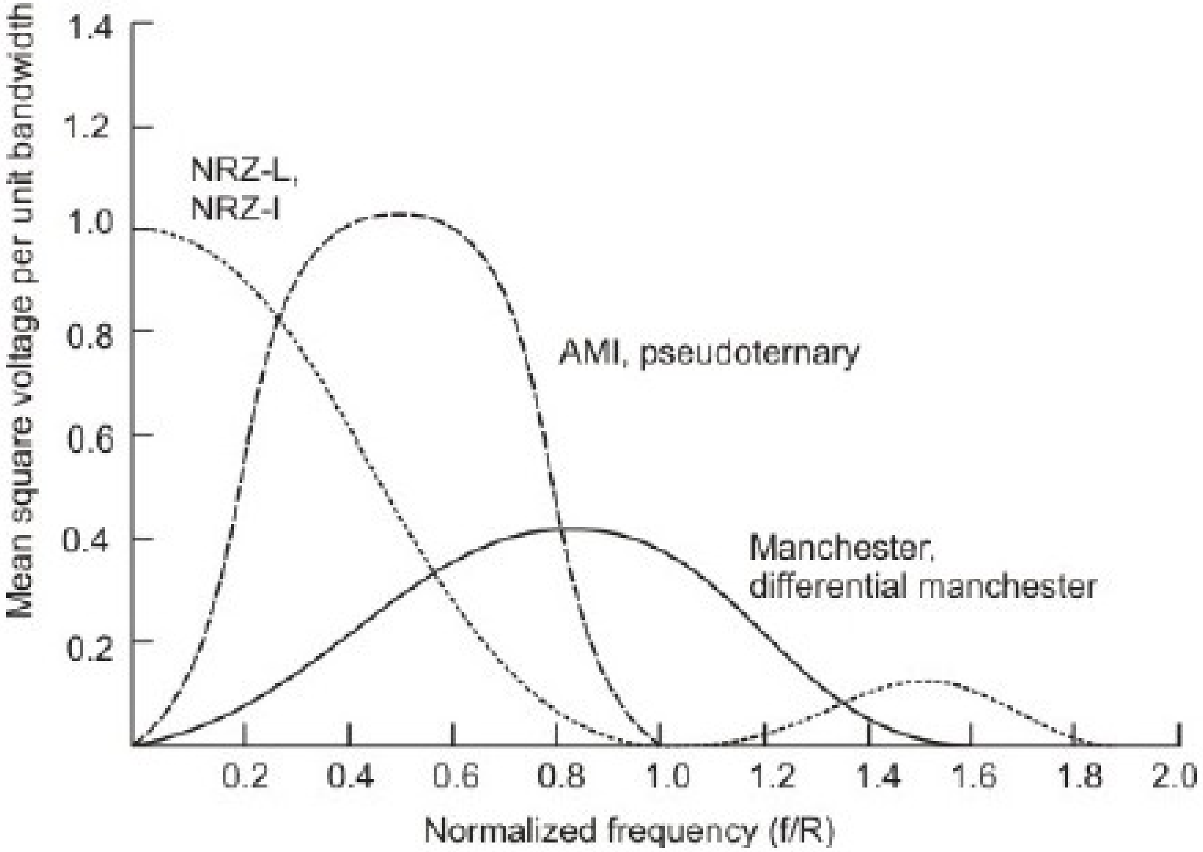
Advantages

- It is a better alternative to NRZ.
- Has no DC component or baseline wandering.
- Lesser bandwidth

Limitation

- Loss of synchronization for long sequences of 0’s i.e., it has no self synchronization because long runs of “0”s results in no signal transitions.
- No error detection.

Frequency spectrum of different encoding techniques



Summary of Line Coding Schemes

Category	**Scheme**	**Bandwidth (average)**	**Characteristics**
Unipolar	NRZ	B=N/2	Costly, no self synchronization if long 0s and 1s, DC
Polar	NRZ-L	B=N/2	No self synchronization if long 0s and 1s, DC
	NRZ-I	B=N/2	No self synchronization for long 0s, DC
	Biphase	B=N	Self synchronization, no DC, high bandwidth
Bipolar	AMI	B=N/2	No self synchronization for long 0s, DC

BLOCK CODING

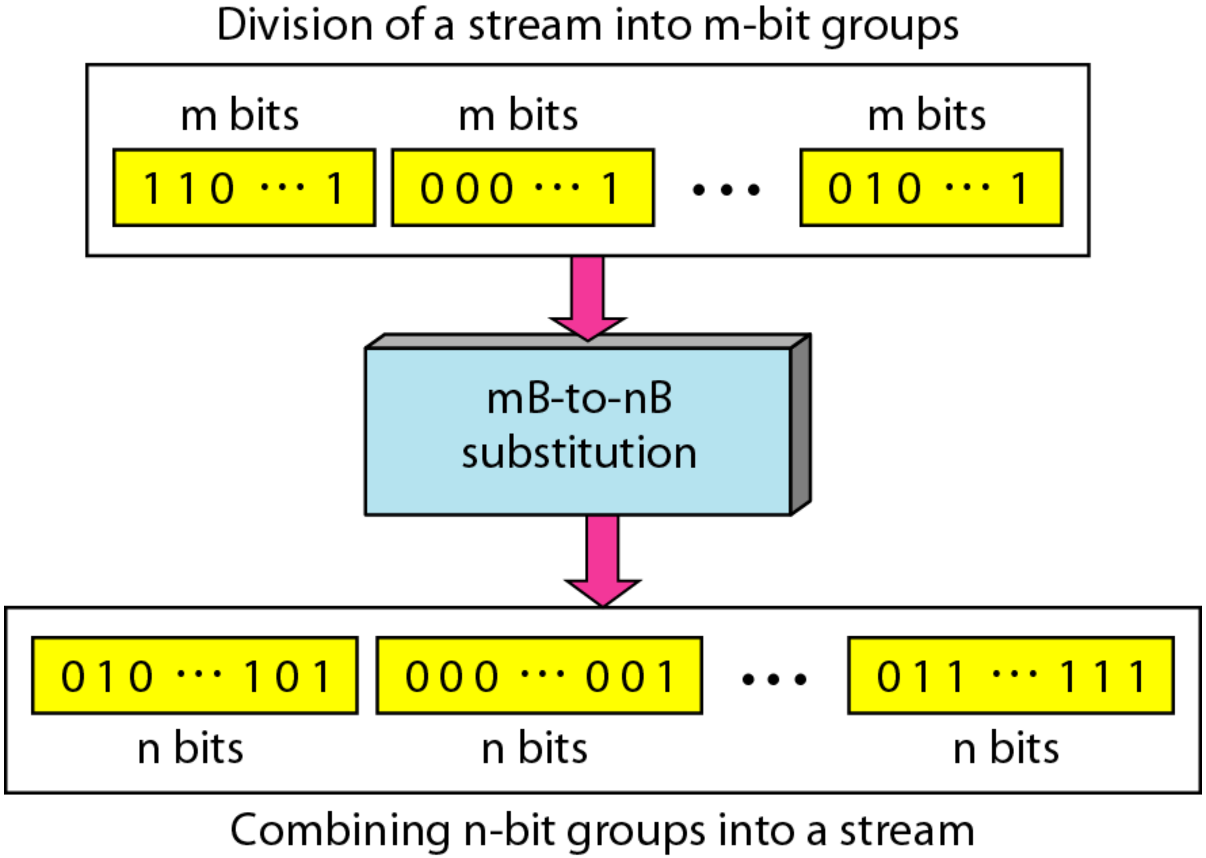
Block Coding

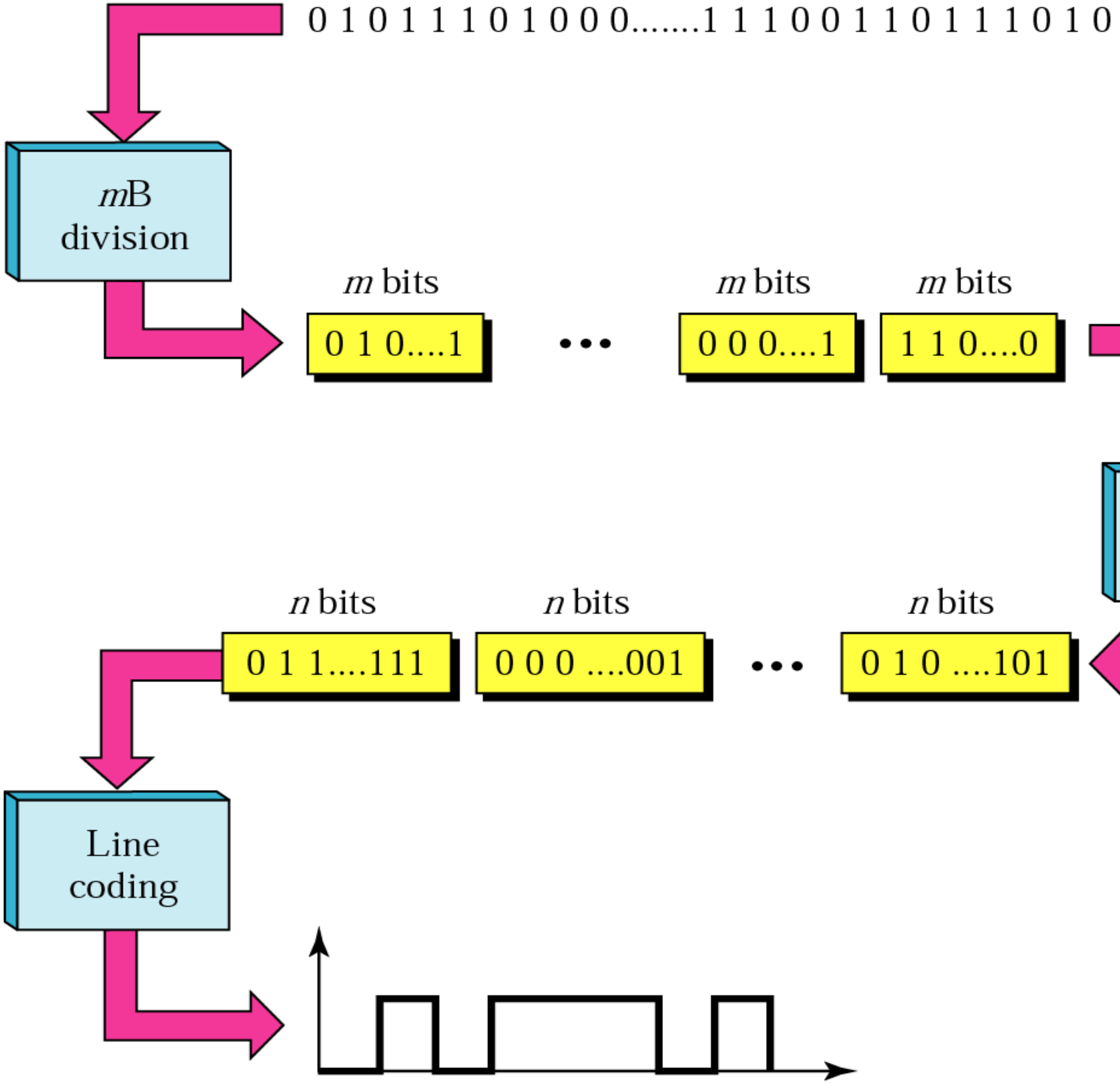
- To ensure accuracy of data frame received, redundant bits are used.
  - For example, in even parity one parity bit is added to make the count of 1s in the frame even. This way the original number of bits are increased. It is called Block Coding.

Block Coding

- For a code to be capable of error detection, we need to add redundancy, i.e., extra bits to the data bits.
- Synchronization also requires redundancy - transitions are important in the signal flow and must occur frequently.
- Block coding is done in three steps: division, substitution and combination.
- Block coding is represented by slash notation: mB/nB.
  - it replaces each m-bit group with an n-bit group.
- The resulting bit stream prevents certain bit combinations that when used with line encoding would result in DC components or poor sync. quality.
- After block coding is done it is line coded for transmission.

Block coding concept

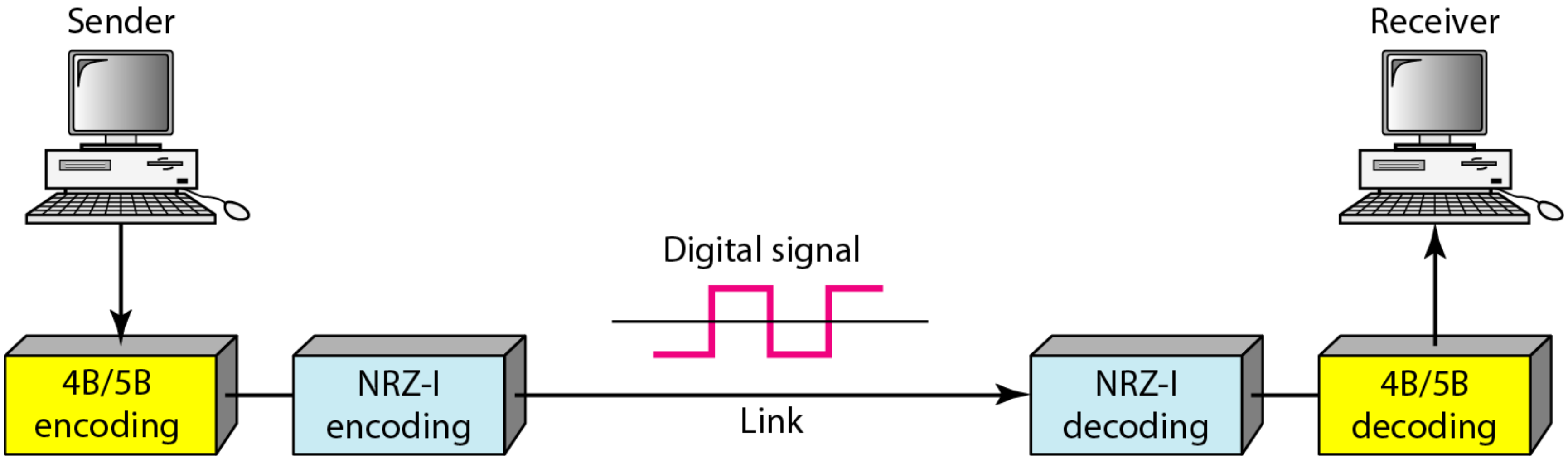



0 1 0 1 1 1 0 1 0 0 0.......1 1 1 0 0 1 1 0 1 1 1 0 1 0
*m*B
division
m bits
0 1 0....1
• • •
m bits
0 0 0....1
m bits
1 1 0....0
*m*B to *n*B
substitution
n bits
0 1 1....111
n bits
0 0 0 001
• • •
n bits
0 1 0 101
Line
coding

4B/5B in combination with NRZ-I

- NRZ-I
  - It has a good signal rate, one half that of the biphase
  - But it has a synchronization problem
    - A long sequence of 0s can make the receiver clock lose synchronization
- One solution is to change the bit stream, prior to encoding with NRZ-I, so that it does not have a long stream of 0s.
  - The 4B/5B scheme achieves this goal.

Using block coding 4B/5B with NRZ-I line coding scheme



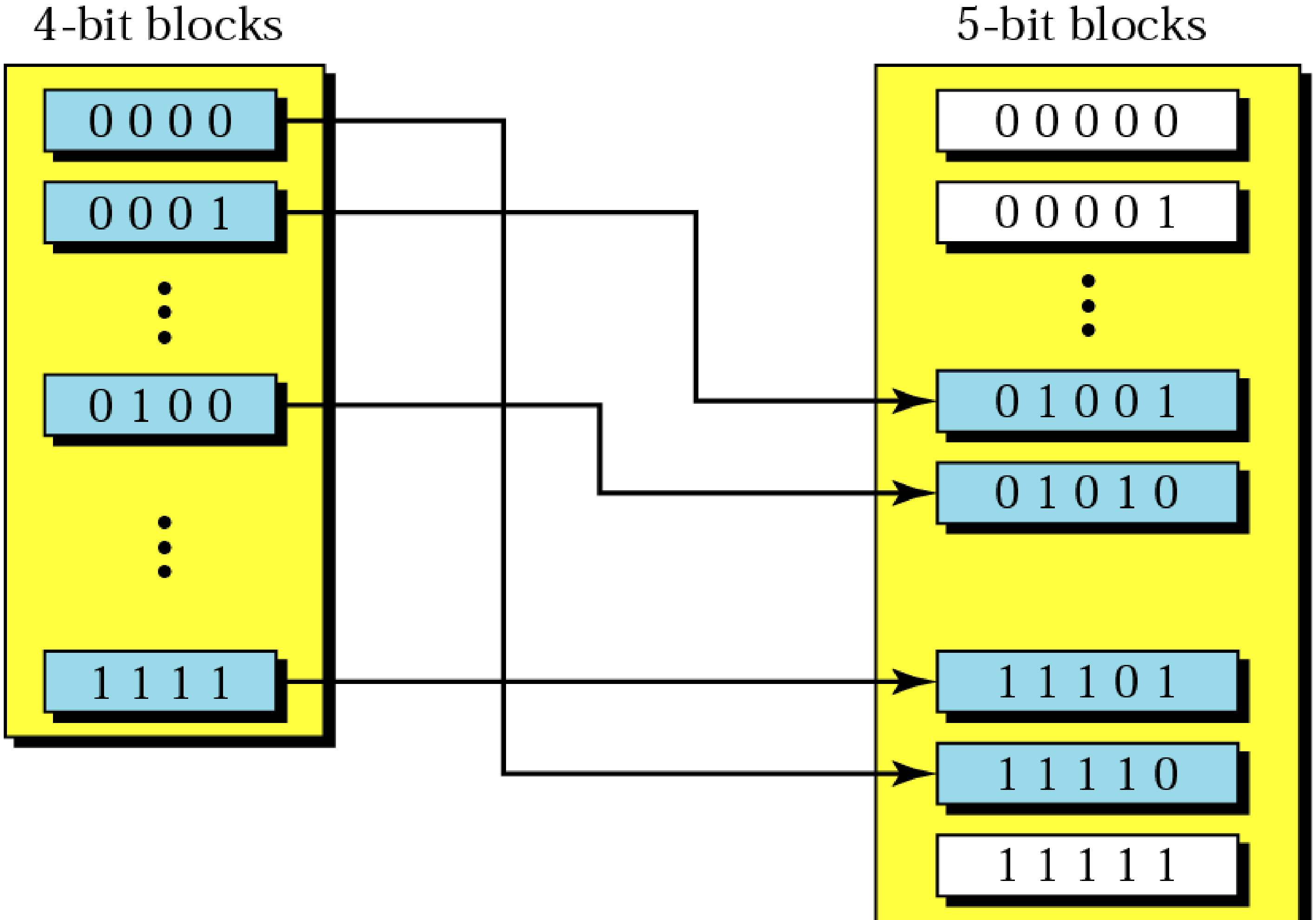
4B/5B solving the synchronization issue

- In 4B/5B the 5 bit output that replaces the 4 bit input has
  - no more than one leading zero (left bit) and
  - no more that two trailing zeros (right bits)
- So when different groups are combined to make a new sequence, there are never more than 3 consecutive 0s.
- Note: NRZ-I has no problem with sequences of 1s.

Redundancy

- A 4 bit data word can have 16 combinations.
- A 5 bit word can have 32 combinations.
- There are 16 groups that are not used for 4B/5B encoding.
- Some of these groups are used for control purposes.

Substitution in 4B/5B block coding



4B/5B encoding

Data	Code	Data	Code
0000	11110	1000	10010
0001	01001	1001	10011
0010	10100	1010	10110
0011	10101	1011	10111
0100	01010	1100	11010
0101	01011	1101	11011
0110	01110	1110	11100
0111	01111	1111	11101

Data	Code
Q (Quiet)	00000
I (Idle)	11111
H (Halt)	00100
J (start delimiter)	11000
K (start delimiter)	10001
T (end delimiter)	01101
S (Set)	11001
R (Reset)	00111

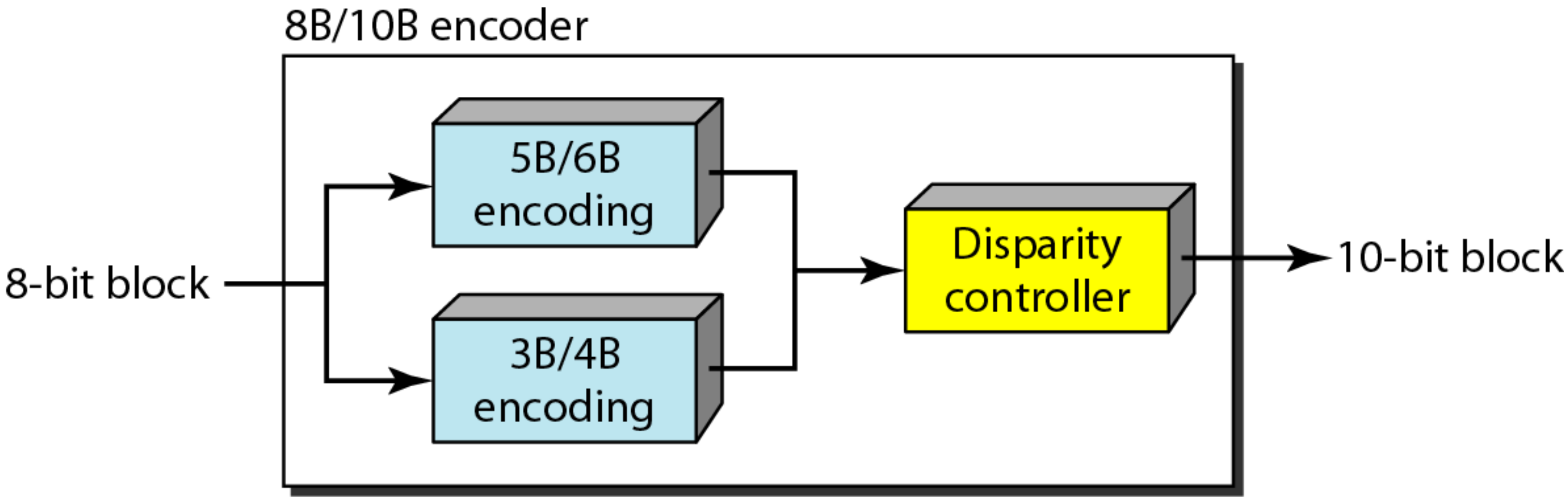
Problem

- We need to send data at a 1-Mbps rate. What is the minimum required bandwidth, using a combination of 4B/5B and NRZ-I or Manchester coding?
  - First 4B/5B block coding increases the bit rate to 1.25 Mbps. The minimum bandwidth using NRZ-I is N/2 or 625 kHz. The Manchester scheme needs a minimum bandwidth of 1.25 MHz. The first choice needs a lower bandwidth, but has a DC component problem; the second choice needs a higher bandwidth, but does not have a DC component problem.

Note

- 4B/5B solves the problem of synchronization and overcomes one of the deficiencies of NRZ-I.
- However
  - It increases the signal rate of NRZ-I
  - The redundant bits add 20% more baud
  - Still the result is less than the biphase scheme which has a signal rate of 2 times that of NRZ-I
- But 4B/5B block encoding does not solve the DC component problem of NRZ-I.
- So, if a DC component is unacceptable, we need to use biphase or bipolar encoding.

8B/10B encoding



8B/10B Key Features

- 8B/10B provides greater error detection capability than 4B/5B.
- The 8B/10B block coding is actually a combination of 5B/6B and 3B/4B encoding.
- To prevent a long run of consecutive 0s and 1s, the code uses a disparity controller which keeps track of excess 0s over 1s (or 1s over 0s)
  - If the bits in the current block create a disparity that contributes to the previous disparity then each bit in the code is complemented (a 0 is changed to 1 and 1 is changed to 0).
- The coding has 2^10-2^8=768 redundant groups that can be used for disparity checking and error detection.
- In general, 8B/10B is superior to 4B/5B because of better built-in-error checking capability and better synchronization.

SCRAMBLING TECHNIQUES

Issues

- Biphase schemes are suitable for dedicated links between stations in a LAN
- But not suitable for long-distance communication because of their wide bandwidth requirement.
- The combination of block coding and NRZ line coding is not suitable for long-distance communication because of the DC component.
- On the other hand, Bipolar AMI encoding has a narrow bandwidth and does not create a DC component.
- However, a long sequence of 0’s upsets the synchronization.
- Looking for a technique that
  - does not increase the number of bits and does provide synchronization
  - That is a solution that substitutes long zero level pulses with a combination of other levels to provide synchronization

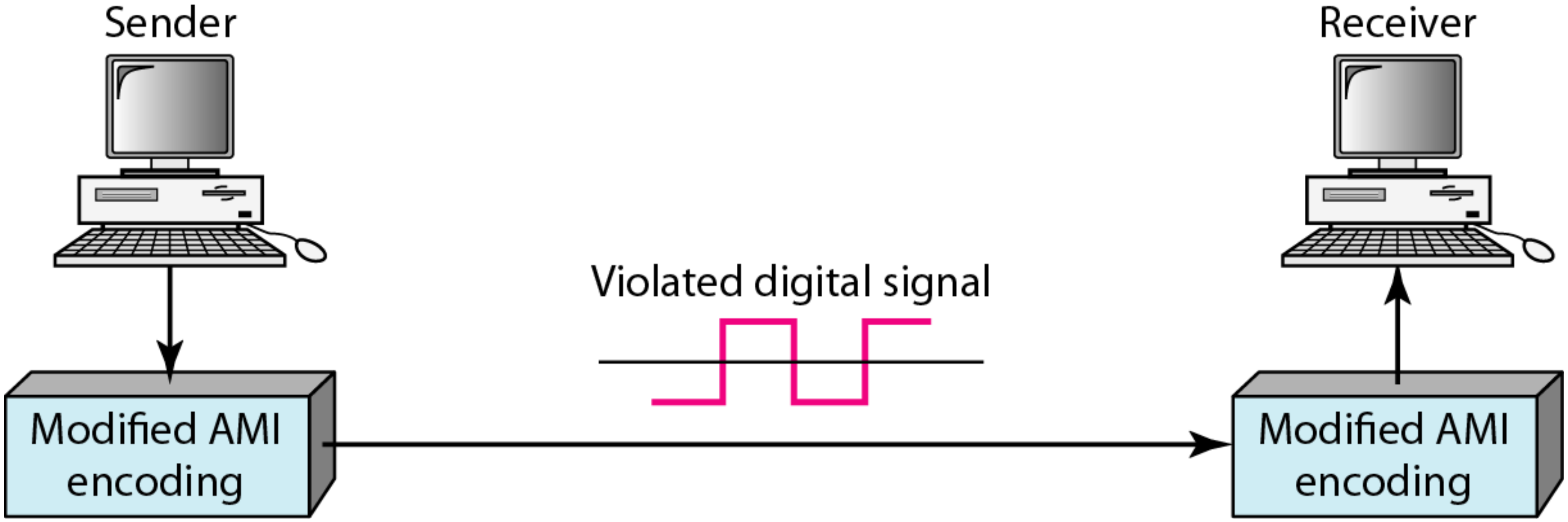
Solution: Scrambling

- Use scrambling to replace sequences that would produce constant voltage

- Main idea:
  - Sequences that would result in a constant voltage are replaced by filling sequences that will provide sufficient transitions for the receiver's clock to maintain synchronization.
  - Filling sequences must be recognized by receiver and replaced with original data sequence.
  - Filling sequence is the same length as original sequence.
- Design goals:
  - No dc component
  - No long sequences of zero-level line signals
  - No reduction in data rate
  - Error detection capability

AMI used with scrambling

- Scrambling is a technique used to create a sequence of bits - self clocking, no low frequencies, no wide bandwidth.
- It is implemented at the same time as encoding, the bit stream is created on the fly.
- It replaces ‘unfriendly’ runs of bits with a violation code.

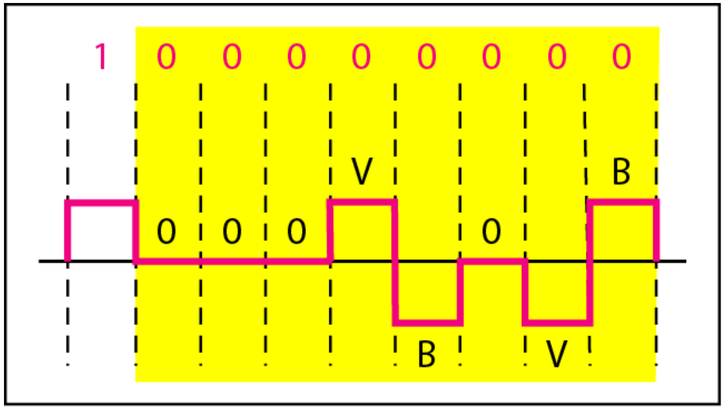


Scrambling Techniques

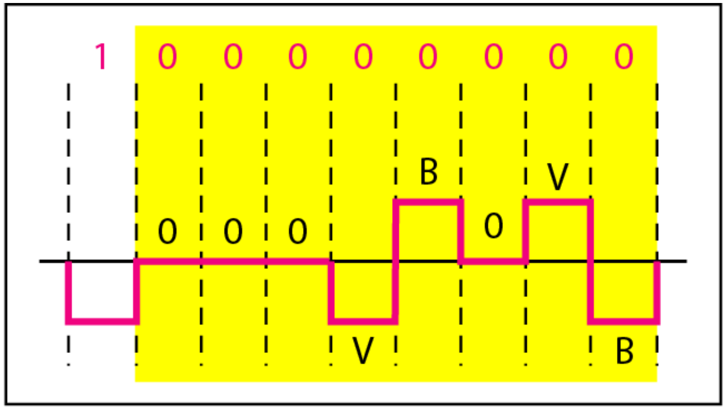
- Two common scrambling techniques are
  - Bipolar with 8-zero substitution (B8ZS)
  - High density bipolar 3-zero (HDB3)

B8ZS

- B8ZS substitutes eight consecutive zeros with 000VB0VB.
- The V (nonzero voltage) stands for violation , it violates the line encoding rule (i.e., breaks an AMI rule of encoding).
- B stands for bipolar, it implements the bipolar line encoding rule (i.e., a nonzero level voltage in accordance with the AMI rule).
- V and B here is relative, V means the same polarity as the polarity of previous nonzero pulse; B means the polarity opposite to the polarity of previous nonzero pulse
- This technique does not change the bit rate rather balances the +ve and –ve voltage levels (2 +ves and 2-ves)
- Substitution may change the polarity of a 1 as AMI needs to follow its rules.



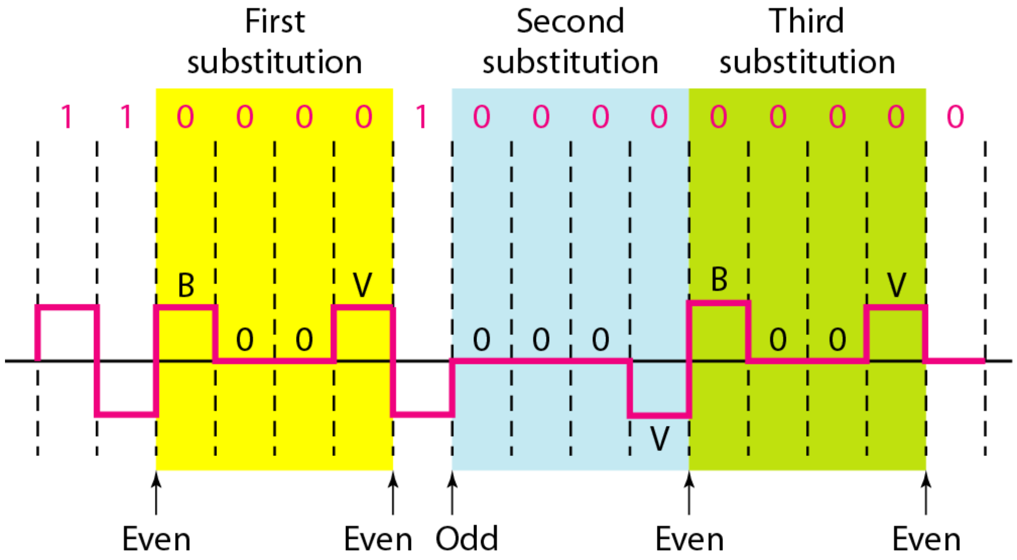
a. Previous level is positive.



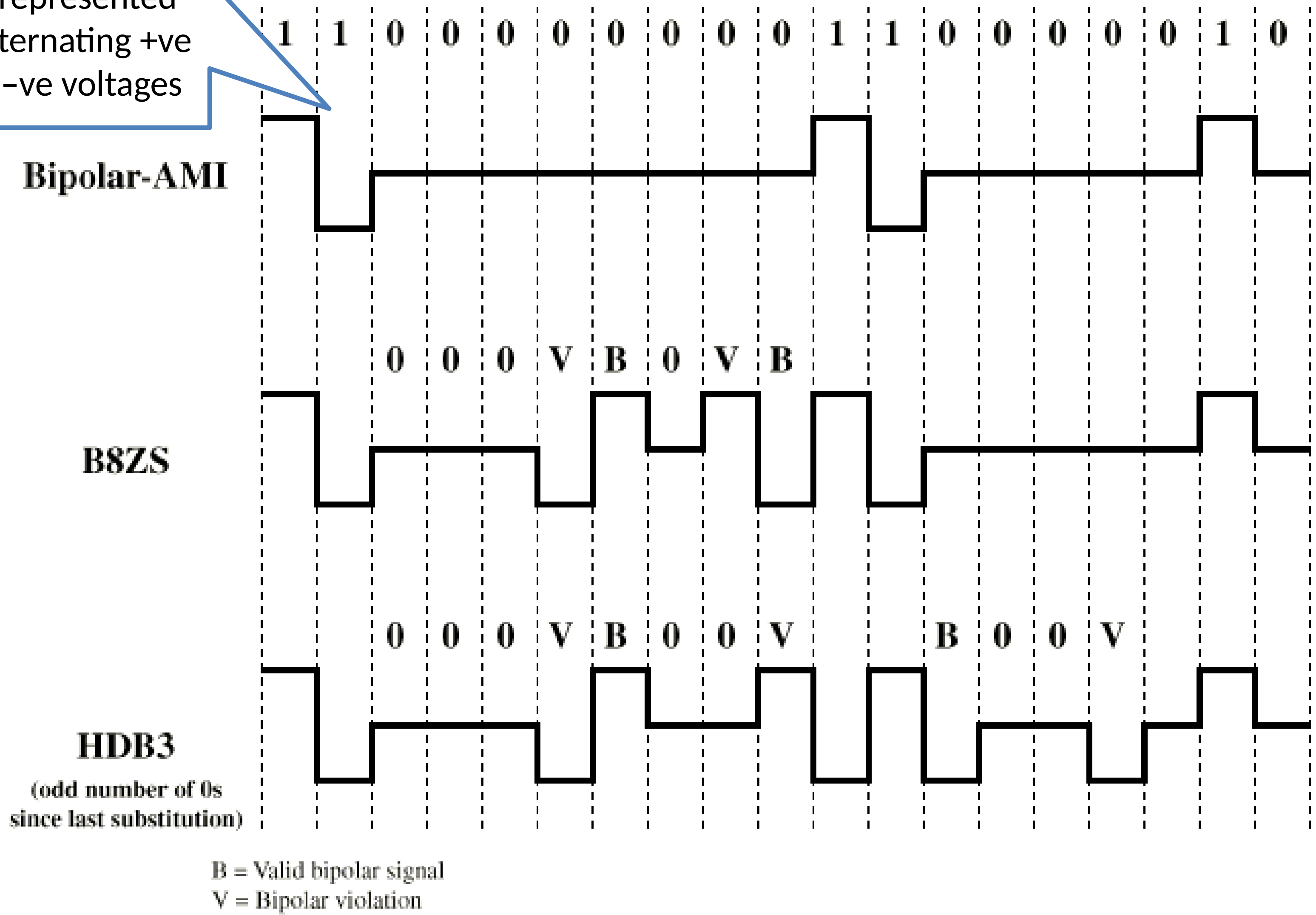
b. Previous level is negative.

HDB3

- HDB3 substitutes four consecutive zeros with 000V or B00V depending on the number of nonzero pulses after the last substitution.
- The reason for two different substitutions is to maintain the even number of non zero pulses after each substitution. Two rules are:
  - If the number of non zero pulses after the last substitution is even then the substitution is B00V to make total number of non zero pulse even.
  - If the number of non zero pulses after the last substitution is odd then the substitution is 000V to make total number of non zero pulses even.



A neutral 0 voltage represents binary 0 and binary 1s are represented by alternating +ve and –ve voltages



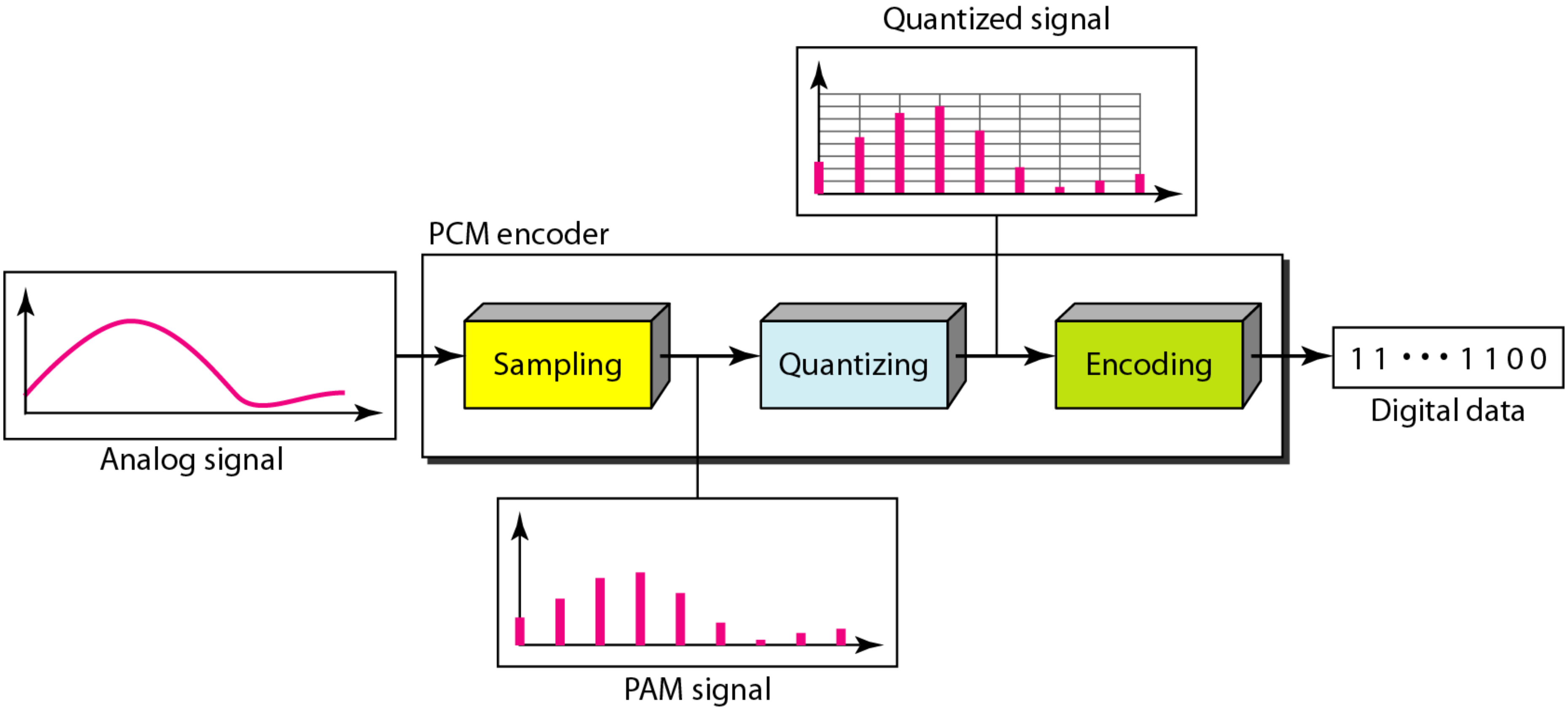
ANALOG TO DIGITAL CONVERSION

- A digital signal is superior to an analog signal
  - because it is more robust to noise and can easily be recovered, corrected and amplified.
- For this reason, the tendency today is to change an analog signal to digital data.
- In this section we discuss two techniques, pulse code modulation and delta modulation.

Pulse code modulation (PCM)

- PCM consists of three steps to digitize an analog signal:
    1. **Sampling**
    2. **Quantization**
    3. **Binary encoding**
- Before we sample, we have to filter the signal to limit the maximum frequency of the signal as it affects the sampling rate.
- Filtering should ensure that we do not distort the signal, i.e., remove high frequency components that affect the signal shape.

Components of PCM encoder

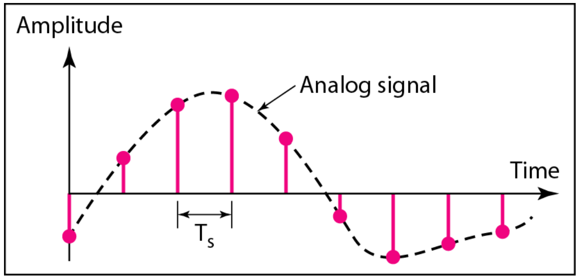


Sampling Theorem

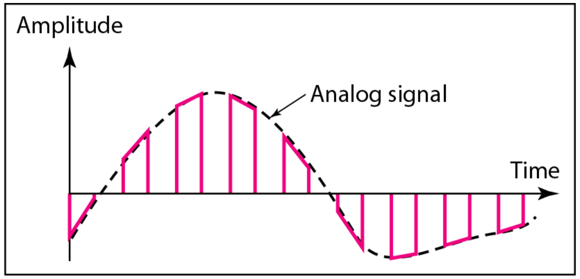
- Sampling Theorem: If a signal is sampled at regular intervals of time and at a rate higher than twice the highest signal frequency, then the samples contain all the information of the original signal.
- For example, voice data are limited to below 4000Hz
  - 8000 samples per second is sufficient to characterize the voice signal.
- Samples are analog samples, called Pulse Amplitude Modulation (PAM) samples.
- To convert to digital, each analog sample must be assigned a binary code.

Sampling

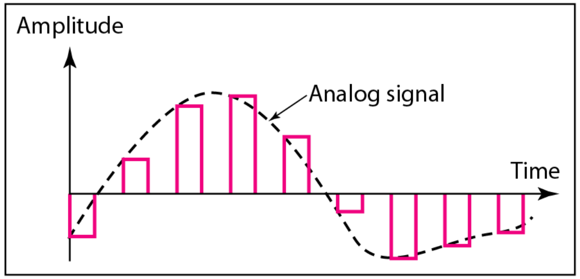
- Analog signal is sampled every T_S secs.
- T_s is referred to as the sampling interval.
- $f_s = 1/T_s$ is called the sampling rate or sampling frequency.
- There are 3 sampling methods:
  - **Ideal** - an impulse at each sampling instant
  - **Natural** - a pulse of short width with varying amplitude
  - **Flattop** - sample and hold, like natural but with single amplitude value



a. Ideal sampling



b. Natural sampling

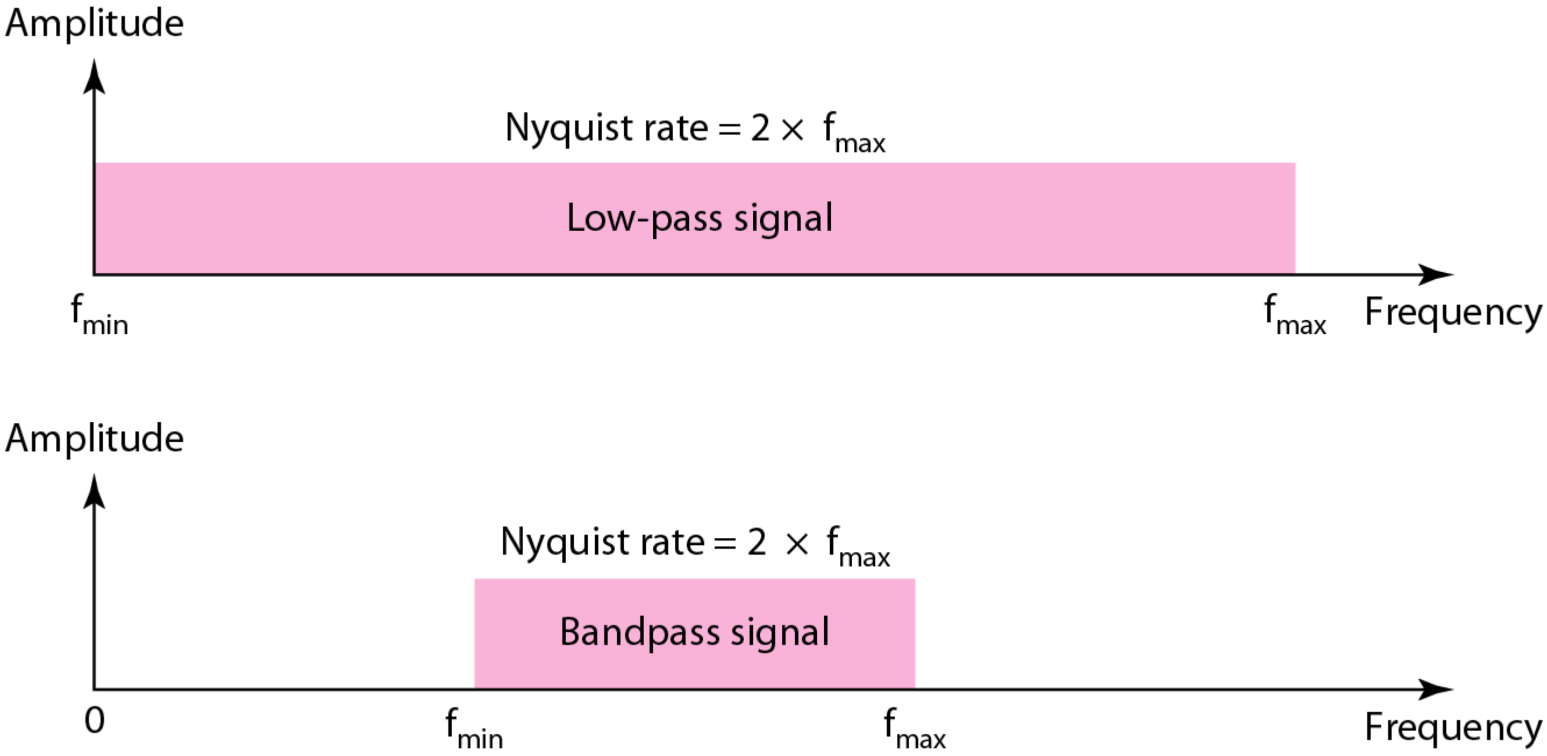


c. Flat-top sampling

Sampling rate

- What are the restrictions on T_s ?
  - According to the **Nyquist theorem**, the sampling rate must be at least 2 times the highest frequency contained in the signal.
- We can sample a signal only if the signal is band limited. In other words, a signal with an infinite bandwidth cannot be sampled.
- The sampling rate must be at least 2 times the highest frequency, not the bandwidth.
  - If the analog signal is low-pass, the bandwidth and the highest frequency are the same value.
  - If the analog signal is bandpass, the bandwidth value is lower than the value of the maximum frequency.

Nyquist sampling rate for low-pass and bandpass signals



Condition on the rate of sampling

- In general, in order to exploit the possibilities of bandpass sampling, the requirement is to determine a base band $(-B,B)$, measured in hertz, such that $(f_L, f_U) \in ([n-1]B, nB)$, where $n \in \{0, 1, 2 \ldots\}$ has an integer value.
- The condition that (f_L, f_U) lies in such an interval implies that

$$f_U \leq nB, f_L \geq (n-1)B$$

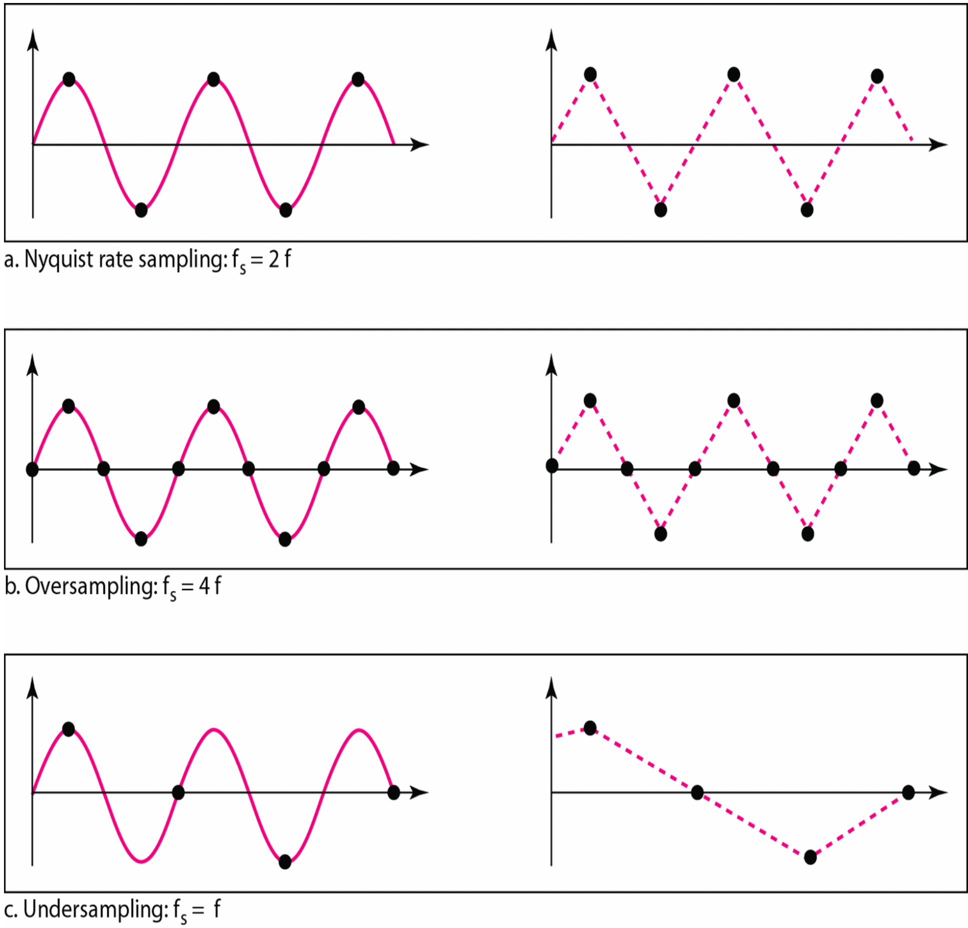
- The condition on the rate of sampling $f_S = 2B$ can be written concisely as

$$2f_U/ n \leq f_S \leq 2f_L/n - 1$$

Example: Recovery of a sampled sine wave for different sampling rates

- For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: f_s = 4f (2 times the Nyquist rate), f_s = 2f (Nyquist rate), and f_s = f (one-half the Nyquist rate).

- It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave (a). Oversampling in (b) can also create the same approximation, but it is redundant and unnecessary. Sampling below the Nyquist rate (c) does not produce a signal that looks like the original sine wave.



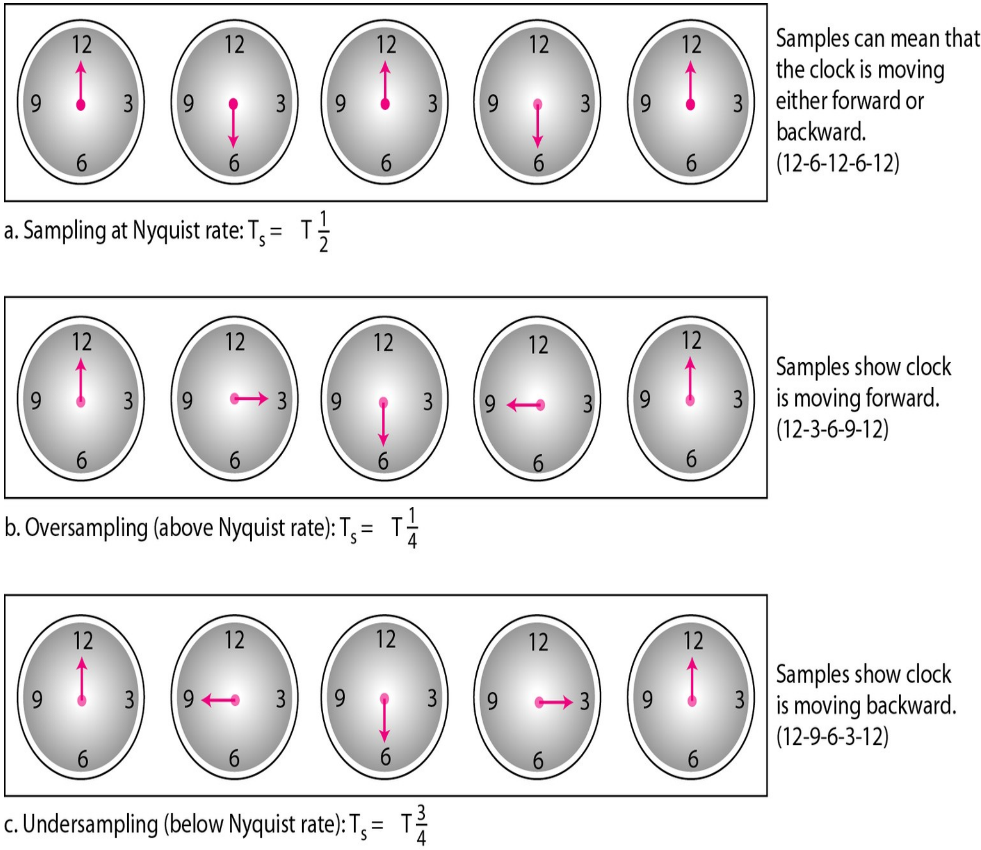
a. Nyquist rate sampling: $f_s = 2$ f

b. Oversampling: $f_s = 4$ f

c. Undersampling: $f_s =$ f

Example: Sampling of a clock with only one hand

- Consider the revolution of a hand of a clock. The second hand of a clock has a period of 60 s. According to the Nyquist theorem, we need to sample the hand every 30 s ($T_s = T/2$ or $f_s = 2f$).
- In (a), the sample points, in order, are 12, 6, 12, 6, 12, and 6. The receiver of the samples cannot tell if the clock is moving forward or backward.
- In (b), we sample at double the Nyquist rate (every 15 s). The sample points are 12, 3, 6, 9, and 12. The clock is moving forward.
- In (c), we sample below the Nyquist rate ($T_s = 3T/4$ or $f_s = 4f/3$). The sample points are 12, 9, 6, 3, and 12. Although the clock is moving forward, the receiver thinks that the clock is moving backward.



Samples can mean that the clock is moving either forward or backward. (12-6-12-6-12)

a. Sampling at Nyquist rate: $T_s = T\frac{1}{2}$

Samples show clock is moving forward. (12-3-6-9-12)

b. Oversampling (above Nyquist rate): $T_s = T\frac{1}{4}$

Samples show clock is moving backward. (12-9-6-3-12)

c. Undersampling (below Nyquist rate): $T_s = T\frac{3}{4}$

Quantization

- Sampling results in a series of pulses of varying amplitude values ranging between two limits: a min and a max.
- The amplitude values are infinite between the two limits.
- We need to map the *infinite* amplitude values onto a finite set of known values.
- This is achieved by dividing the distance between min and max into L zones, each of height ㅌ ㄱㅅ

ㅌ = (max - min)/L

Quantizing

- Each sample is quantized into some level
  - The original signal is now only approximated and cannot be recovered exactly
  - This effect is called quantizing error or quantizing noise
- For example, 8 bit sample gives 256 levels
- 8000 samples per second and 8 bits per sample gives 64kbps, for a single voice signal.

Quantization Levels

- The midpoint of each zone is assigned a value from 0 to L-1 (resulting in L values)
- Each sample falling in a zone is then approximated to the value of the midpoint.

Quantization Zones

- Assume we have a voltage signal with amplitudes V_{min}=-20V and V_{max}=+20V.
- We want to use L=8 quantization levels.
- Zone width $_{\sqsubset}{}^{\mathrm{E}}$ = (20 – (-20))/8 = 5
- The 8 zones are: -20 to -15, -15 to -10, -10 to -5, -5 to 0, 0 to +5, +5 to +10, +10 to +15, +15 to +20
- The midpoints are: -17.5, -12.5, -7.5, -2.5, 2.5, 7.5, 12.5, 17.5

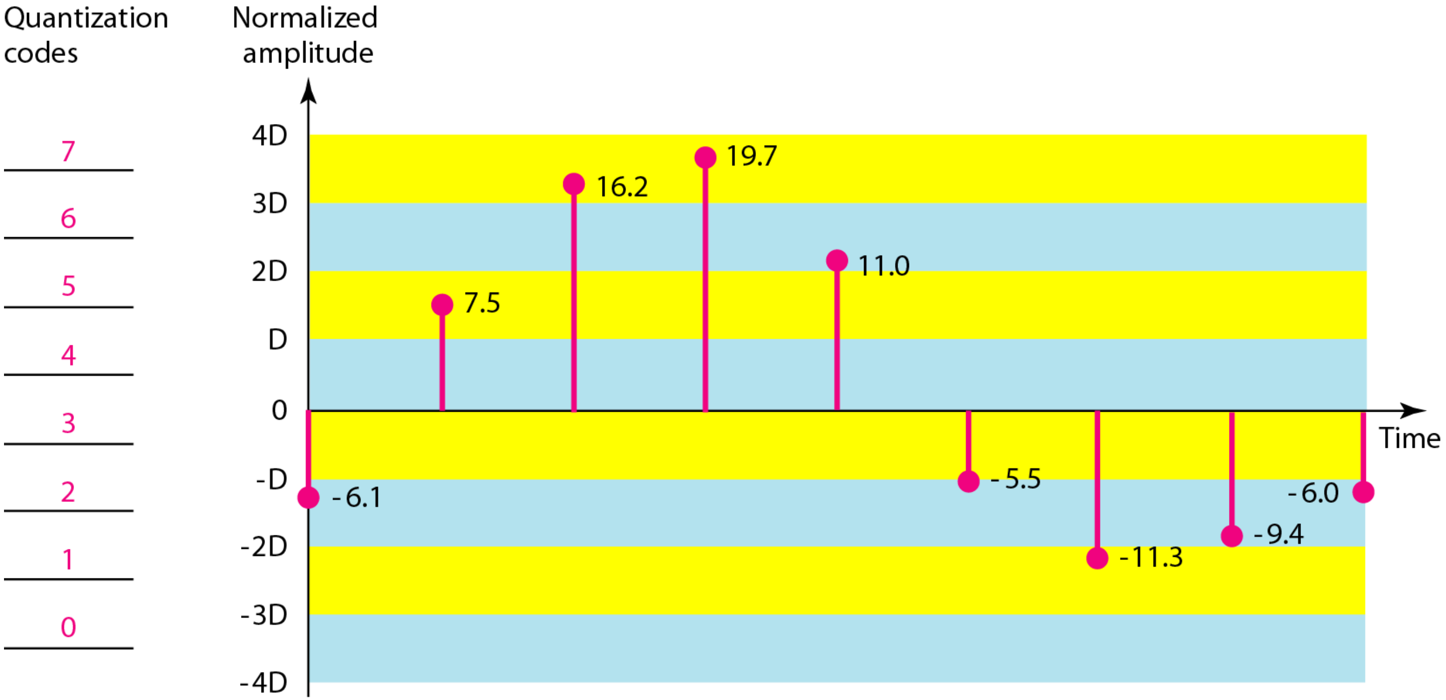
Assigning Codes to Zones

- Each zone is then assigned a binary code.
- The number of bits required to encode the zones, or the number of bits per sample as it is commonly referred to, is obtained as follows:

$$n_b = \log_2 L$$

- Given our example, $n_b = 3$
- The 8 zone (or level) codes are therefore: 000, 001, 010, 011, 100, 101, 110, and 111
- Assigning codes to zones:
  - 000 will refer to zone -20 to -15
  - 001 to zone -15 to -10, etc.

Quantization and encoding of a sampled signal



Normalized PAM values	-1.22	1.50	3.24	3.94	2.20	-1.10	-2.26	-1.88	-1.20
Normalized quantized values	-1.50	1.50	3.50	3.50	2.50	-1.50	-2.50	-1.50	-1.50
Normalized error	-0.38	0	+0.26	-0.44	+0.30	-0.40	-0.24	+0.38	-0.30
Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

Quantization Error

- When a signal is quantized, we introduce an error - the coded signal is an approximation of the actual amplitude value.
- The difference between actual and coded value (midpoint) is referred to as the quantization error.
- The more zones, the smaller E which results in smaller errors.
- BUT, the more zones the more bits required to encode the samples -> higher bit rate

Quantization Error and SN$_Q$R

- Signals with lower amplitude values will suffer more from quantization error as the error range: E /2, is fixed for all signal levels.
- Non linear quantization is used to alleviate this problem. Goal is to keep SN$_Q$R fixed for all sample values.
- Two approaches:
  - The quantization levels follow a logarithmic curve. Smaller E 's at lower amplitudes and larger ⊏ E 's at higher amplitudes.
  - Companding and Expanding: The sample values are compressed at the sender into logarithmic zones, and then expanded at the receiver. The zones are fixed in height.

Bit rate and bandwidth requirements of PCM

- The bit rate of a PCM signal can be calculated form the number of bits per sample x the sampling rate

$$\text{Bit rate} = n_b \times f_s$$

- The bandwidth required to transmit this signal depends on the type of line encoding used. Refer to previous section for discussion and formulas.
- A digitized signal will always need more bandwidth than the original analog signal. Price we pay for robustness and other features of digital transmission.

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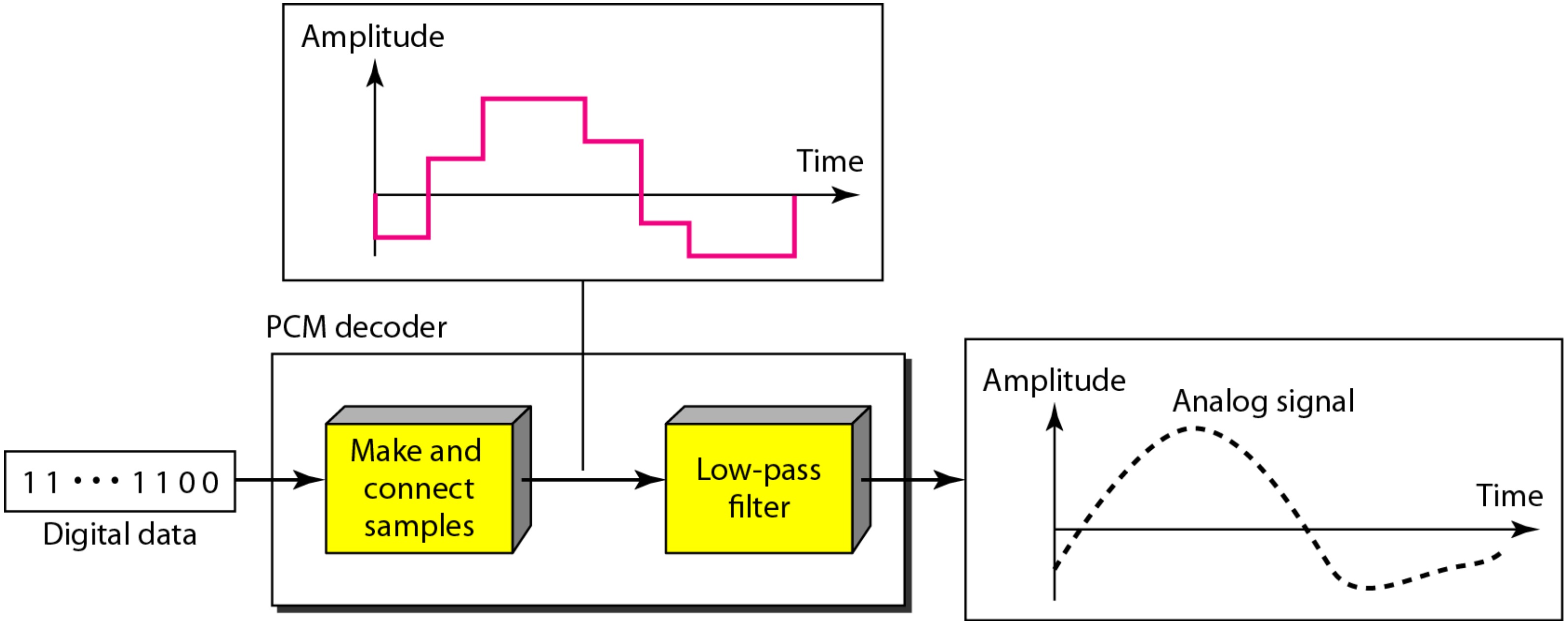
PCM Decoder: Original Signal Recovery

- To recover an analog signal from a digitized signal we follow the following steps:
  - We use a hold circuit that holds the amplitude value of a pulse till the next pulse arrives.
  - We pass this signal through a low pass filter with a cutoff frequency that is equal to the highest frequency in the pre-sampled signal.
- The higher the value of L, the less distorted a signal is recovered.



PCM Bandwidth

- Suppose we are given the bandwidth of a low-pass analog signal. If we then digitize the signal, what is the new minimum bandwidth of the channel that can pass this digitized signal?
  - The minimum bandwidth of a line encoded signal is

    B_{min}= c X N X (1/r)   [slide 11]

    Substitute the value of N,

    B_{min}= c X n_b x f_s X (1/r) = c X n_b X 2 X B_{analog} X 1/r

    Now, when 1/r= 1 (for a NRZ or bipolar signal) and c=1/2 (the average situation), the minimum bandwidth is

    B_{min}= n_b X B_{analog}

    i.e., the minimum bandwidth of the digital signal is n_b times greater than the bandwidth of the analog signal.

This is the price we pay for digitization.

Maximum Data Rate of a Channel

- According to the Nyquist theorem, the data rate of a channel is

  $N_{max} = 2 \text{ X } B \text{ X } \log_2 L$ bps

- We can deduce this rate from the Nyquist sampling theorem by using the following arguments.

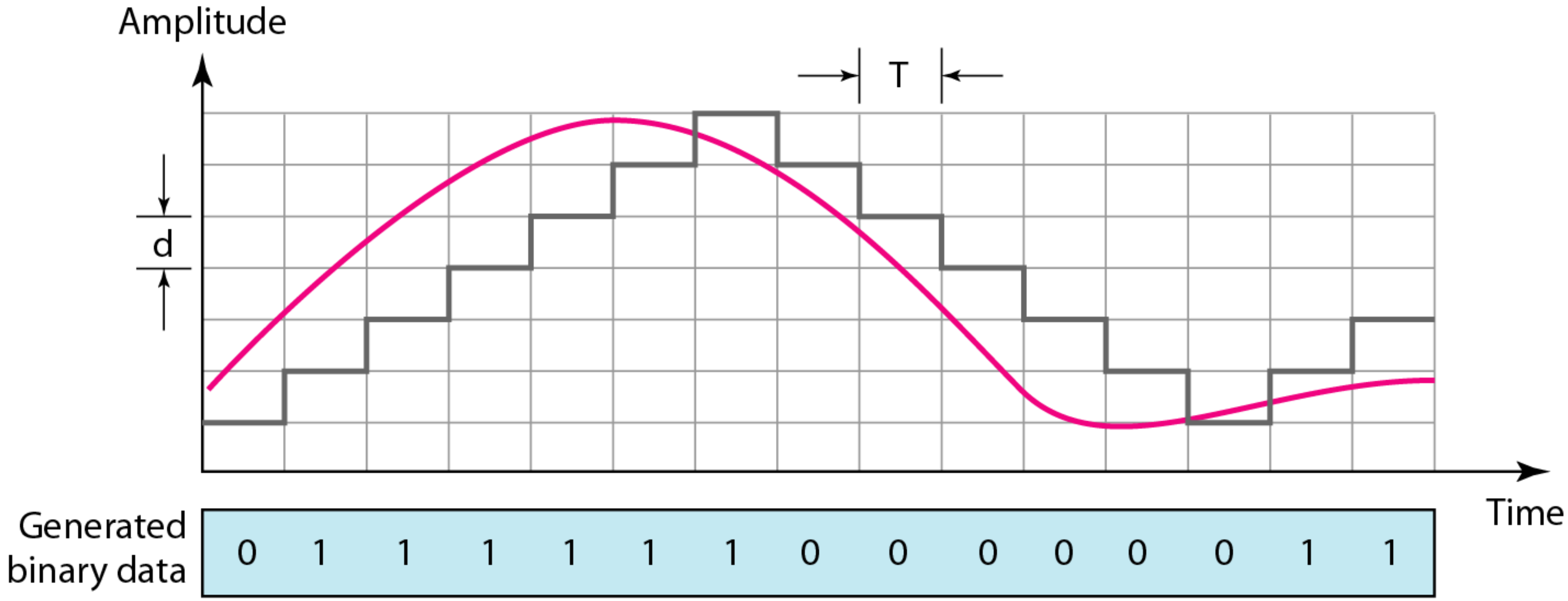
1. We assume that available channel is low pass with bandwidth B
2. We assume that the digital signal we want to send has L levels, where each level is a signal element. This means $r=1/\log_2 L$
3. We first pass the digital signal through a low pass filter to cut off the frequencies above B Hz.
4. We treat the resulting signal as an analog signal and sample it at 2 X B samples per second and quantize it using L levels. Additional quantization levels are useless because the signal originally had L levels
5. The resulting bit rate is $N= n_b \text{ x } f_s = 2 \text{ X } B \text{ X } \log_2 L$ This is the maximum bandwidth.. Thus $N_{max} = 2 \text{ X } B \text{ X } \log_2 L$ bps
6. If the date rate and the number of signal levels are fixed then the minimum bandwidth is $B=N/ (2 \text{ X } \log_2 L)$ Hz

DELTA MODULATION

Delta Modulation (DM)

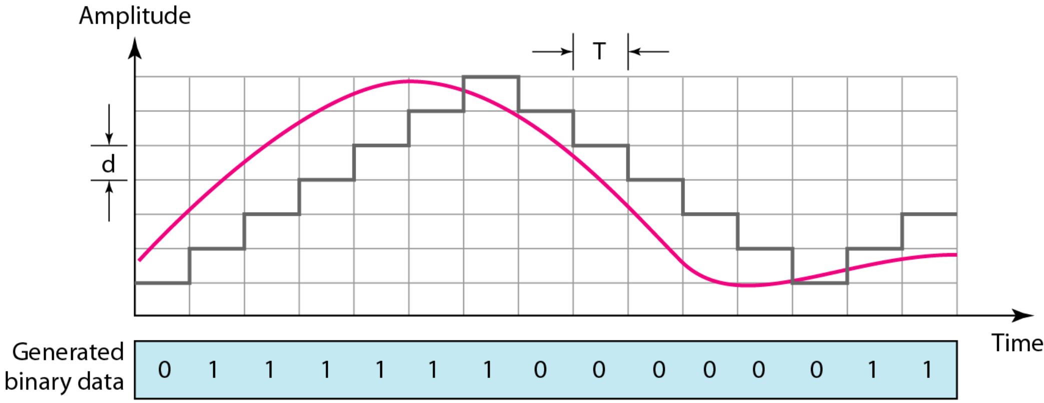
- PCM is a very complex technique. Other techniques have been developed to reduce the complexity of PCM. The simplest is delta modulation.
- PCM finds the value of the signal amplitude for each sample
- DM finds the change from the previous sample.

Bits are sent one after another i.e., no code words here



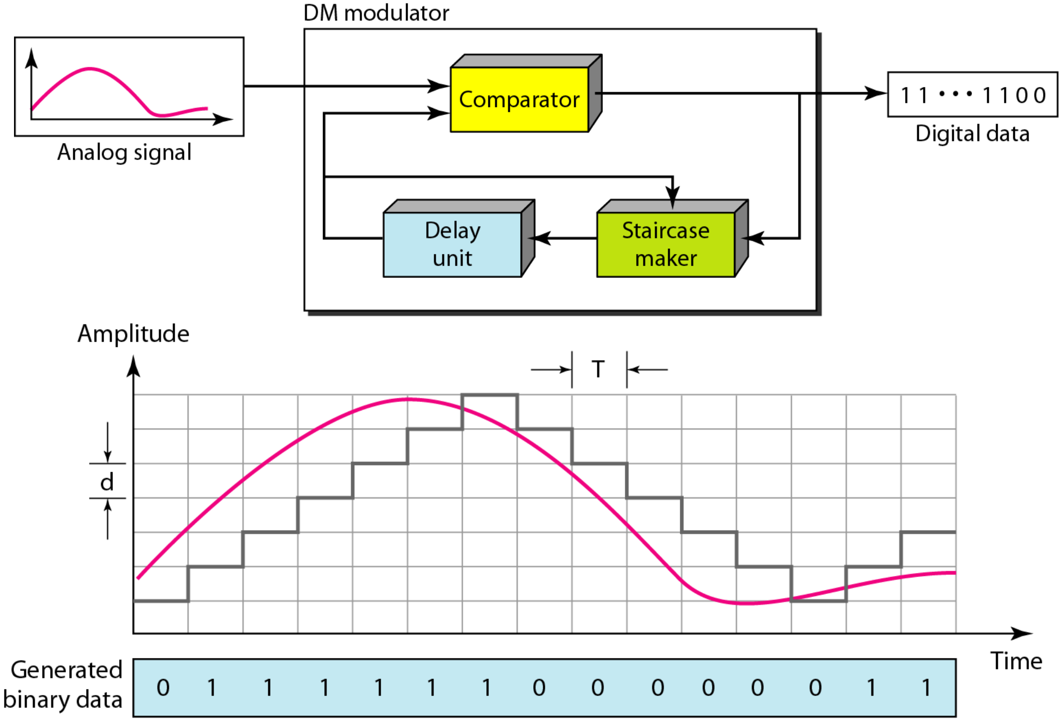
Delta Modulation

- This scheme sends only the difference between pulses, if the pulse at time t_{n+1} is higher in amplitude value than the pulse at time t_n, then a single bit, say a “1”, is used to indicate the positive value.
- If the pulse is lower in value, resulting in a negative value, a “0” is used.
- This scheme works well for small changes in signal values between samples.
- If changes in amplitude are large, this will result in large errors.



DM Modulation Components

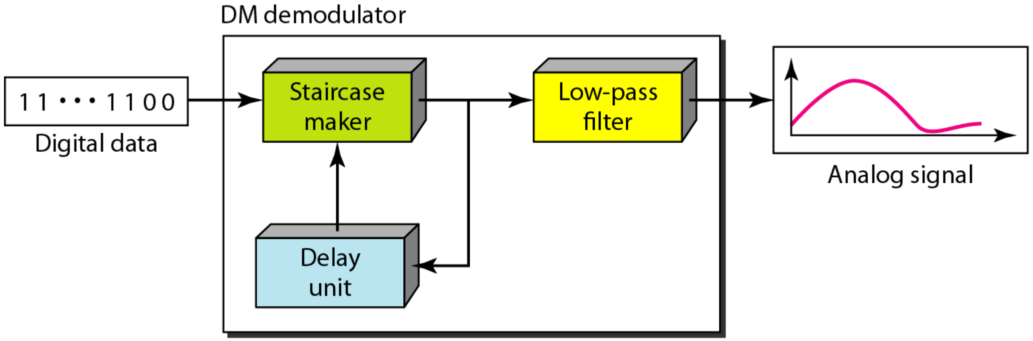
- The modulator is used at the sender site to create a stream of bits from an analog signal.
- The process records the small +ve or –ve changes, called delta (d).
- If the d is +ve, the process records a 1; if it is –ve, the process records a 0.
- However, the process needs a base against which the analog signal is compared.
- The modulator builds a second signal that resembles a staircase.
- Finding the change is then reduced to comparing the input signal with the gradually made staircase signal.



The modulator, at each sampling interval, compares the value of the analog signal with the last value of the staircase signal. If the amplitude of the analog signal is larger, the next bit in the digital data is 1 otherwise 0. Here a delay unit required to hold the staircase function for a period between two comparisons

DM Demodulation Components

- The demodulator takes the digital data and using, the staircase maker and the delay unit, creates the analog signal.
- The created analog signal, however needs to pass through a low-pass filter for smoothing.



- A better performance can be achieved if the value of delta is not fixed. In **adaptive DM**, the value of delta (d) changes according to the amplitude of the analog signal.

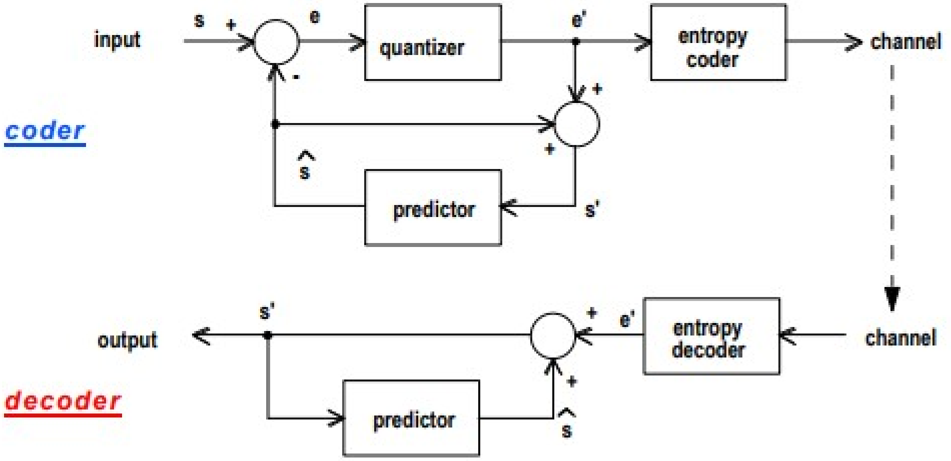
- DM is also not perfect. **Quantization error** is always introduced in the process. The quantization error of DM is much less than that for PCM.

Delta PCM (DPCM)

- Instead of using one bit to indicate positive and negative differences, we can use more bits -> quantization of the difference.
- Each bit code is used to represent the value of the difference.
- The more bits the more levels -> the higher the accuracy.

Differential Pulse Code Modulation (DPCM)

- DPCM is a procedure of converting an analog into a digital signal in which an analog signal is sampled and then the difference between the actual sample value and its predicted value (predicted value is based on previous sample or samples) is quantized and then encoded forming a digital value.
- DPCM code words represent differences between samples unlike PCM where code words represented a sample value.
- Basic concept of DPCM - coding a difference, is based on the fact that most source signals show significant correlation between successive samples so encoding uses redundancy in sample values which implies lower bit rate.



Prediction error

$$\boxed{e = s - \hat{s}}$$

Reconstruction

$$\boxed{s' = e' + \hat{s}}$$

Reconstruction error = quantization error

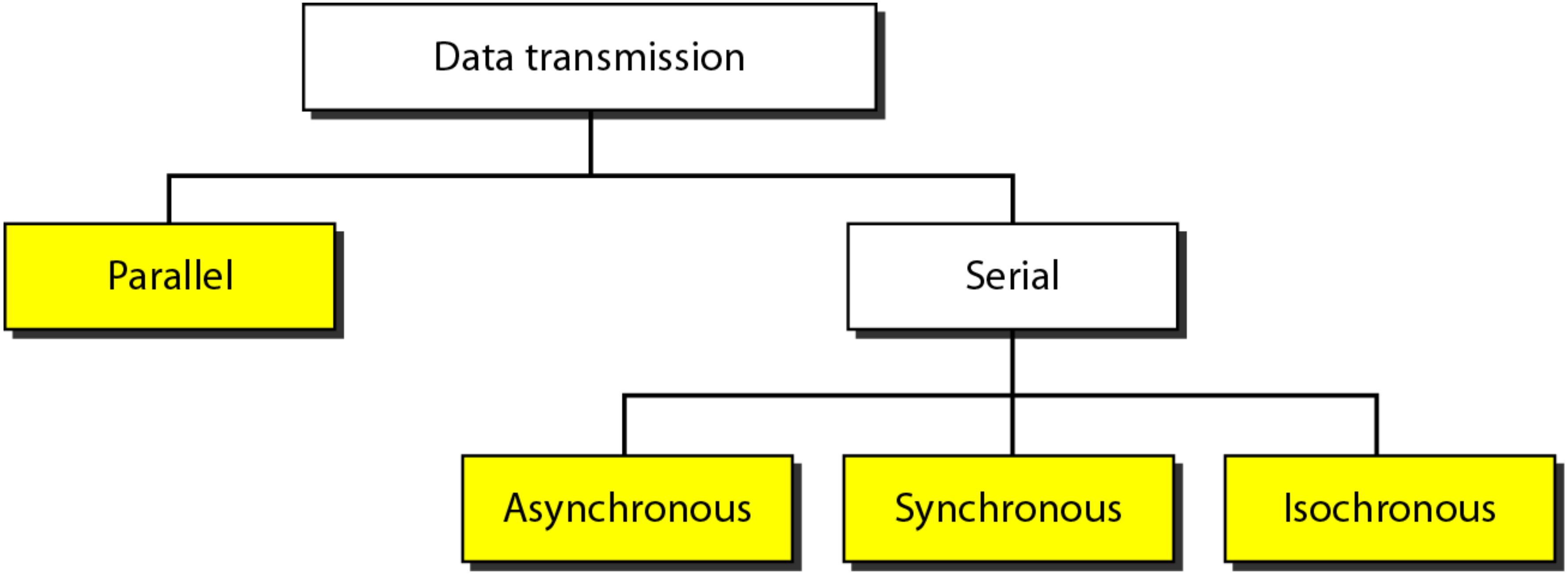
$$\boxed{s' - s = e' - e = q}$$

Transmission modes

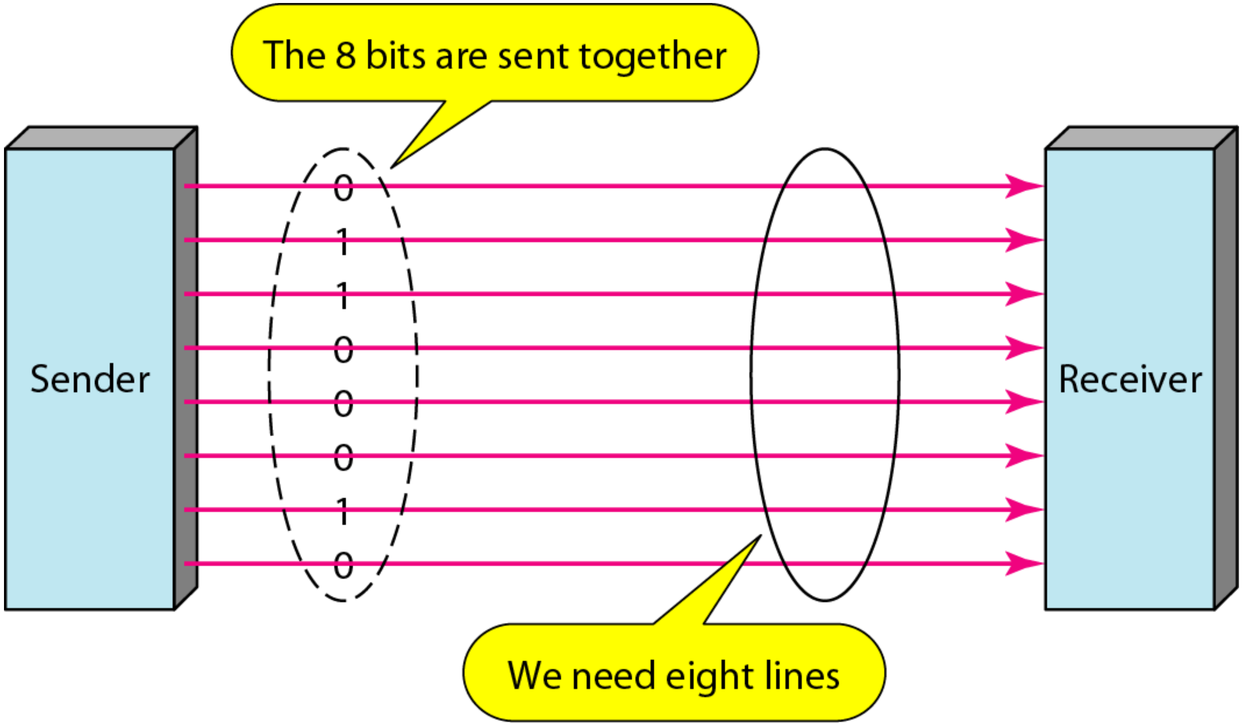
- The transmission of binary data across a link can be accomplished in either parallel or serial mode.
  - In parallel mode, multiple bits are sent with each clock tick.
  - In serial mode, 1 bit is sent with each clock tick.

  While there is only one way to send parallel data, there are three subclasses of serial transmission: *asynchronous*, *synchronous*, and *isochronous*.

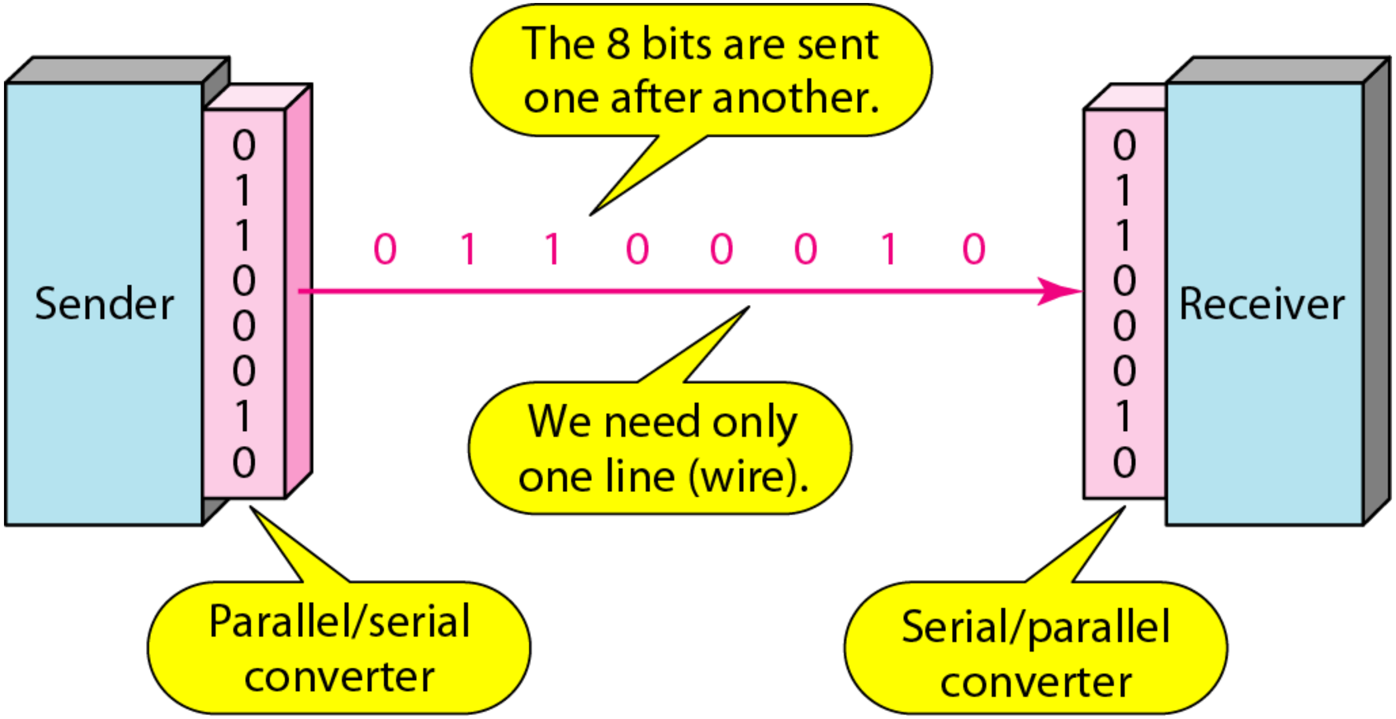
Data transmission and modes



Parallel Transmission



Serial Transmission



The 8 bits are sent one after another.

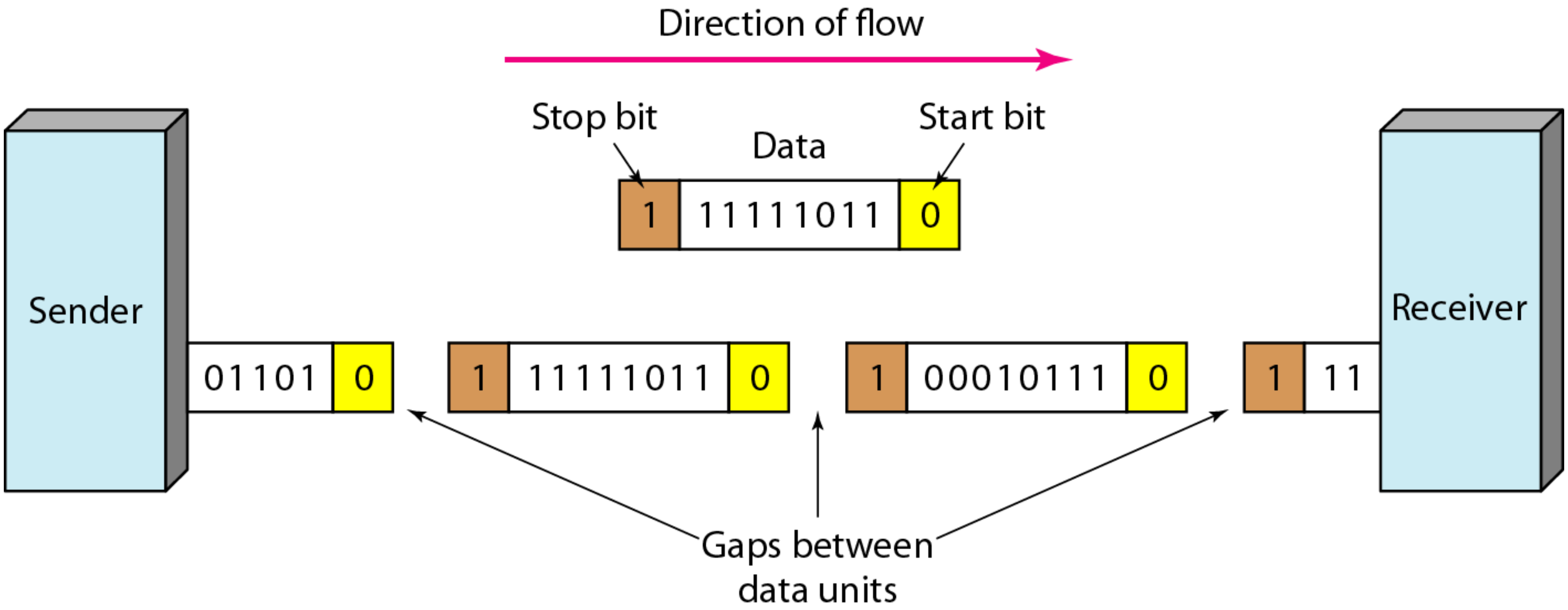
We need only one line (wire).

Parallel/serial converter

Serial/parallel converter

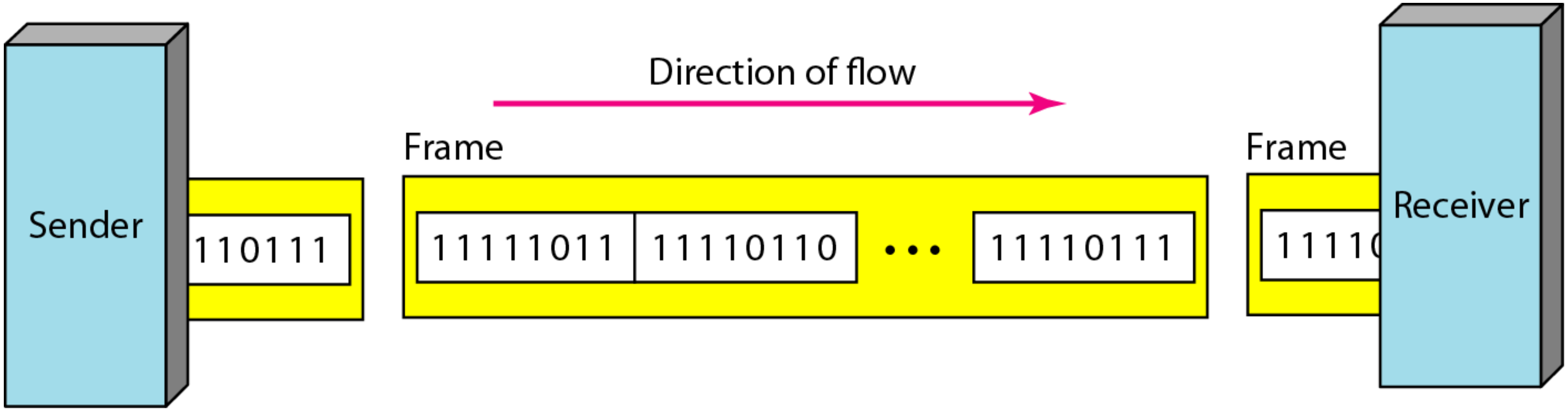
Asynchronous Transmission

- In asynchronous transmission, we send one start bit (0) at the beginning and one or more stop bits (1s) at the end of each byte. There may be a gap between each byte.
- Asynchronous here means "asynchronous at the byte level," but the bits are still synchronized; their durations are the same.



Synchronous Transmission

- In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits. The bits are usually sent as bytes and many bytes are grouped in a frame. A frame is identified with a start and an end byte.



Isochronous

- In isochronous transmission we cannot have uneven gaps between frames.
  - Transmission of bits is fixed with equal gaps.

- In real time audio and video, in which uneven delays between frames are not acceptable, synchronous transmission fails.
  - For example, TV images are broadcast at the rate of 30 images per second (say); they must be viewed at the same rate. If each image is sent by using one or more frames, there should be no delays between frames. For this type of application, the entire stream of bits must be synchronized.
- Thus, the isochronous transmission guarantees that the data arrive at a fixed rate.