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Data Communications

Chapter 04 Digital Transmission

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What are we **going to do** in this **chapter?**

Goal

- Advantages and Disadvantages of Digital Transmission over Analog Transmission
- Learn about various digital transmission schemes
- Digital-to-Digital Conversion Techniques
- Analog-to-Digital Conversion Techniques

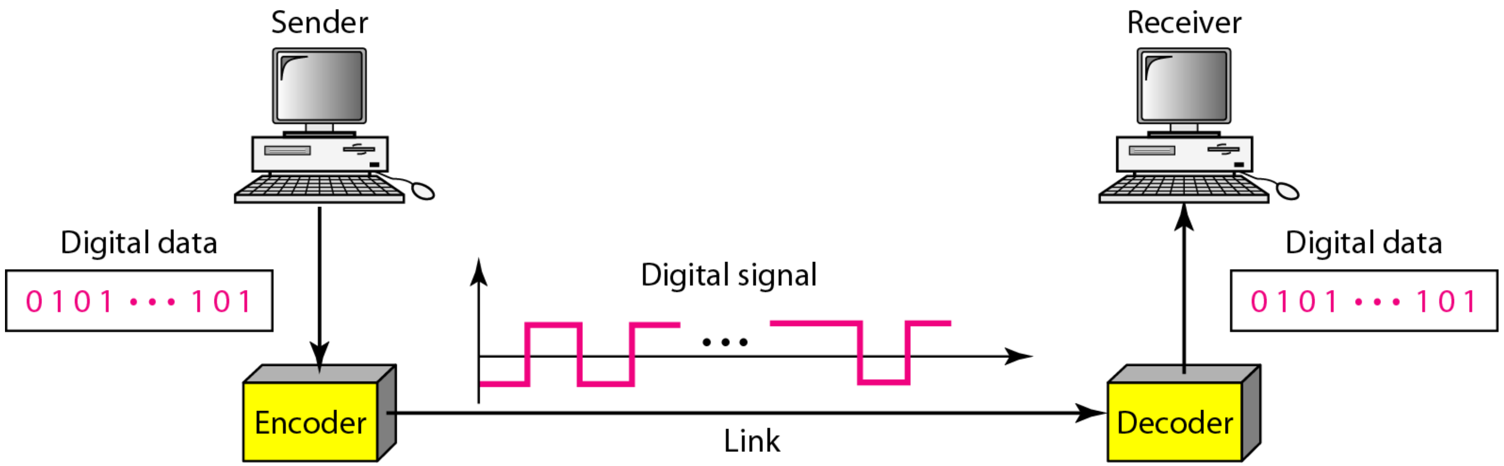
Contents

Line Coding

- the representation of digital information by a digital signal

Characteristics of Line Coding

- Signal Element versus Data Element
- Data Rate versus Signal Rate
- Baseline Wandering
- DC Component
- Self-Synchronization
- etc



4.1 Signal Element versus Data Element

Signal Element

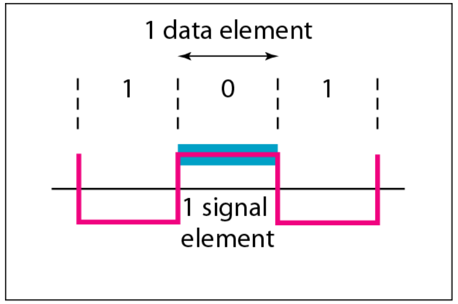
- Shortest unit of digital signal

Data Element

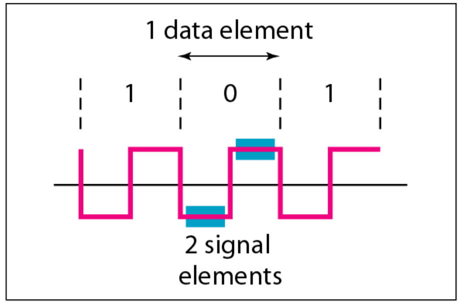
- Smallest entity that can represent a piece of information

Ratio

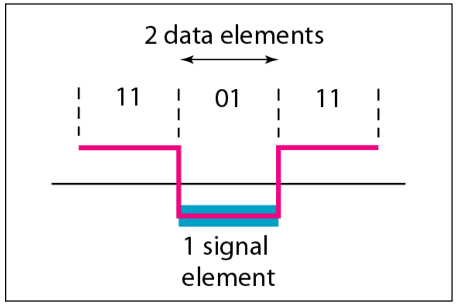
- *r = # of data /# of signal*



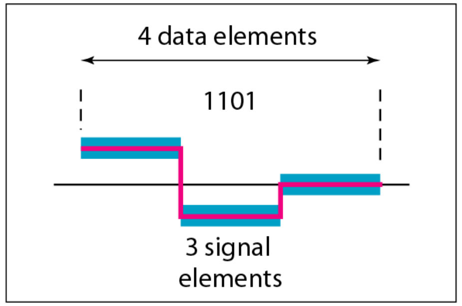
a. One data element per one signal element (r = 1)



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 $\left(r = \frac{4}{3}\right)$

4.1 Data Rate versus Signal Rate

Data Rate

- the number of data(bits) sent in a second.
- Unit → Bit Rate (bps)

Signal Rate

- the number of signal element sent in a second.
- Unit → Pulse Rate, Modulation Rate, Baud Rate.

What we want in Data communication is to Increase Data rate while Decreasing Signal Rate.

***S=c*N*1/r* baud**

Ex. 4.1

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

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

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

The baud rate, not the bit rate, determine the required bandwidth for a digital signal.

- Minimum bandwidth is proportional to the signal rate.
- $B_{min} = c*N*1/r$, $N_{max} = r*B*1/c$

Ex. 4.2

- The maximum data rate of a channel (see Chapter 3) is $N_{max} = 2 \times B \times \log2 L$ (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

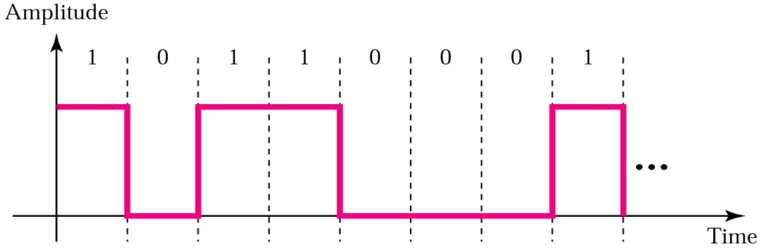
$$N\max = \frac{1}{c} \times B \times r = 2 \times B \times \log_2 L$$

Baseline

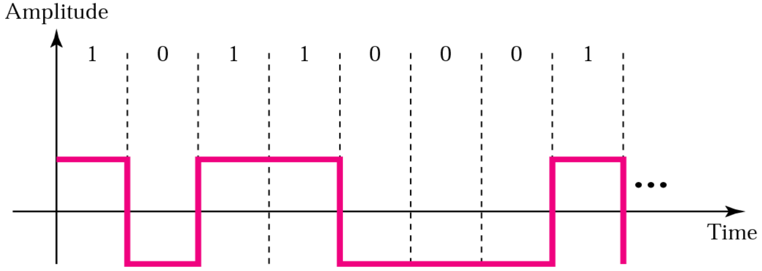
- The average power of received signal
- Long series of 1s and 0s can cause baseline wandering

DC (Direct Current) Component

- Constant digital signal results in a low frequency signal.
- the signal can be distorted and may create an error in the output.
- DC Component is the residual energy in the line and useless.



a. A signal with dc component

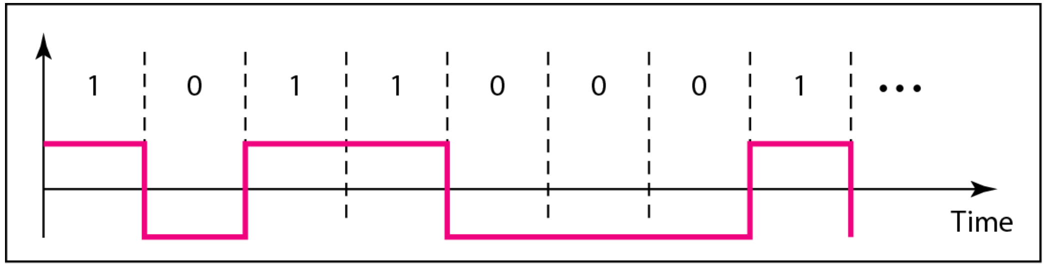


b. A signal without dc component

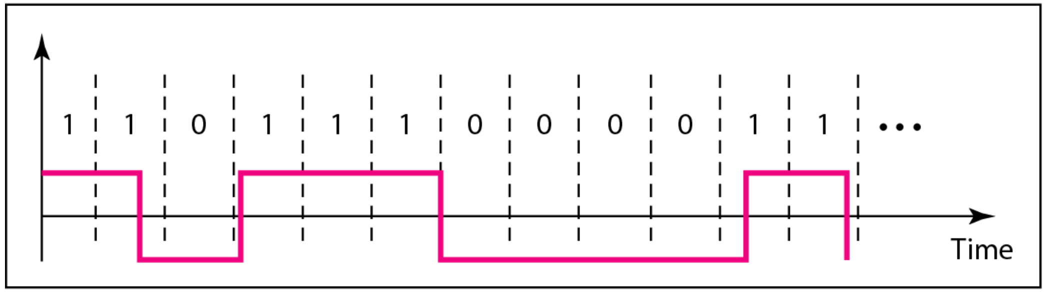
4.1 Self Synchronization

To correctly interpret the signals received from the sender, the receiver's bit intervals must correspond exactly to the sender's bit intervals.

Self synchronizing digital signal includes timing information in the data being transmitted.



a. Sent



b. Received

Built in Error Detection

- Whether the code has an ability to detect an error that occurred during transmission.

Immunity to Noise and Interference

- A code that is immune to the noise and interference.

Complexity

- Complex code cost more.

Five broad categories

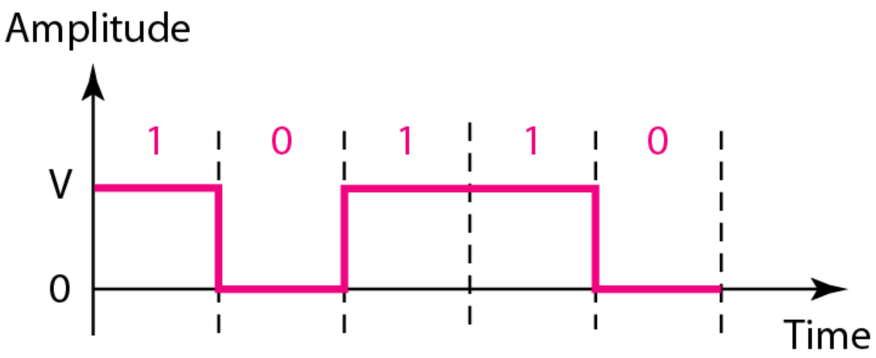
- **Line coding**
  - **Unipolar** — NRZ
  - **Polar** — NRZ,RZ, and biphase (Manchester, and differential Manchester)
  - **Bipolar** — AMI and pseudoternary
  - **Multilevel** — 2B/1Q, 8B/6T, and 4D-PAM5
  - **Multitransition** — MLT-3

uses only one level of value (1: positive value, 0: idle)

Simple and inexpensive

Unipolar encoding problems

- almost obsolete today
- Having DC Component :cannot travel through media that cannot handle DC Components, such as microwave.
- Baseline wandering
- Synchronization : The receiver has to rely on a timer.



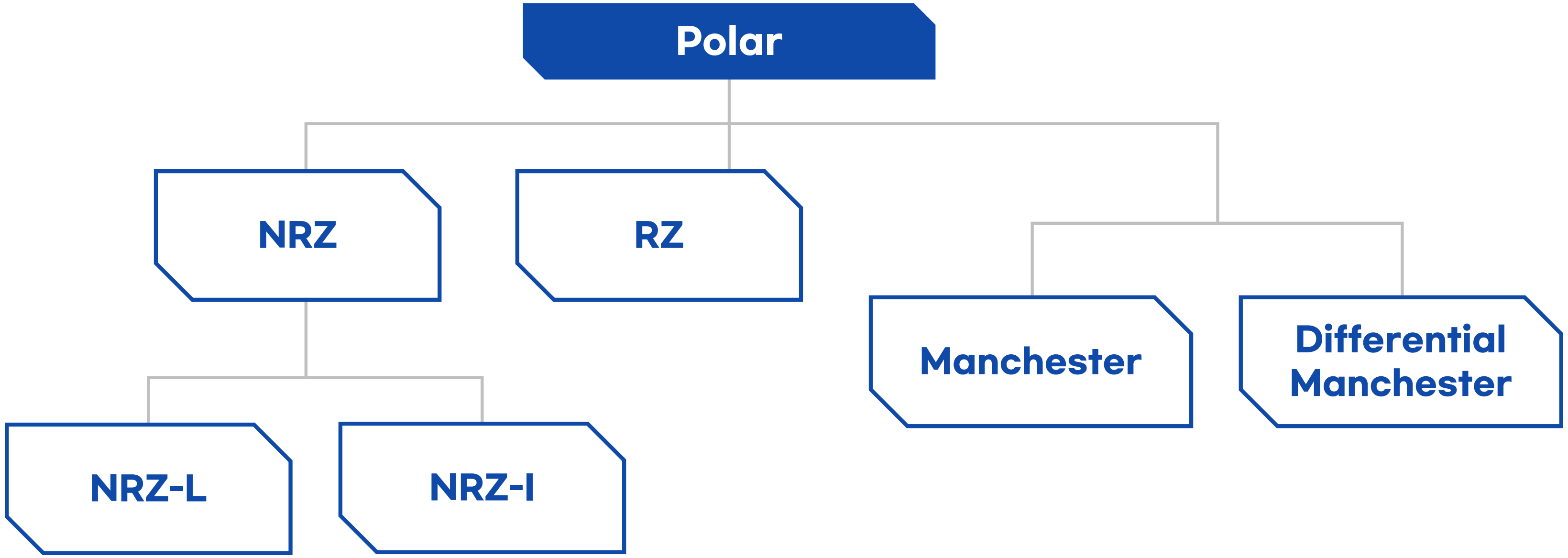
$$\frac{1}{2}V^2 + \frac{1}{2}(0)^2 = \frac{1}{2}V^2$$

Normalized power

4.1 Polar Encoding

Uses two levels (positive and negative) of amplitude.

Types of polar encoding



the value of the signals is always either positive or negative.

NRZ-L (Level)

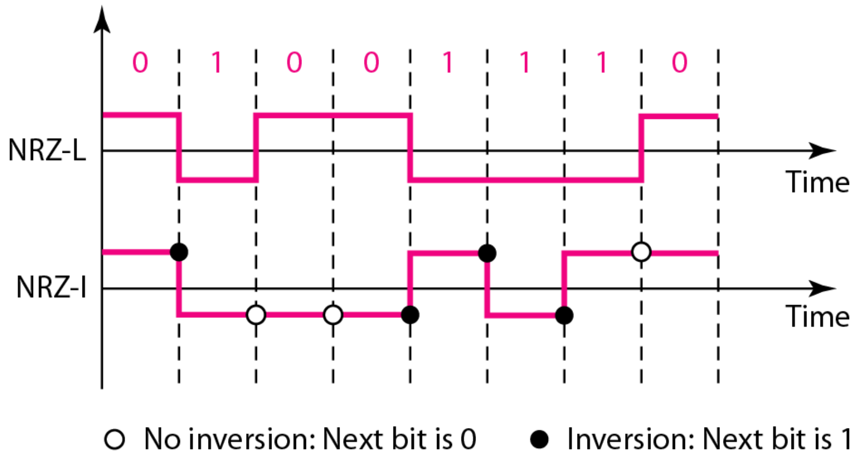
- the level of the signal is dependent upon the state of the bit.
- positive means 0, negative means 1.

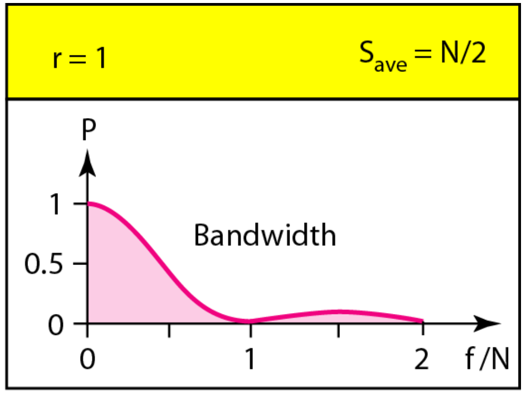
NRZ-I (Invert)

- the signal is inverted if a 1 is encountered.

DC component?, Baseline?

NRZ-I is superior to NRZ-L due to the synchronization provided by the signal change each time a 1 bit is encountered.



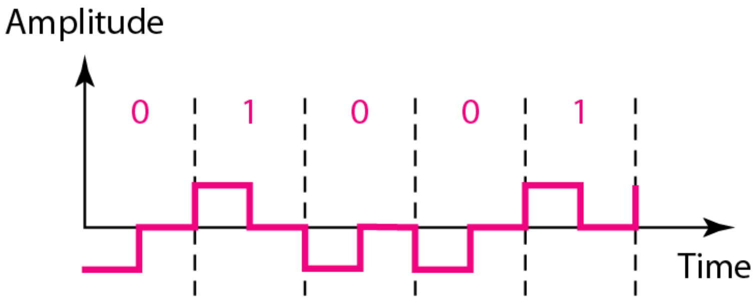


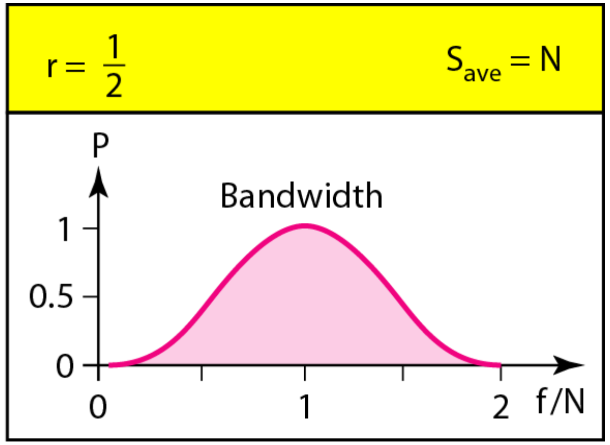
To ensure synchronization, there must be a signal change for each bit in NRZ.

RZ (Return to Zero)

- using three values (positive, negative, zero)
  - 1 : positive-to-zero
  - 0 : negative-to-zero

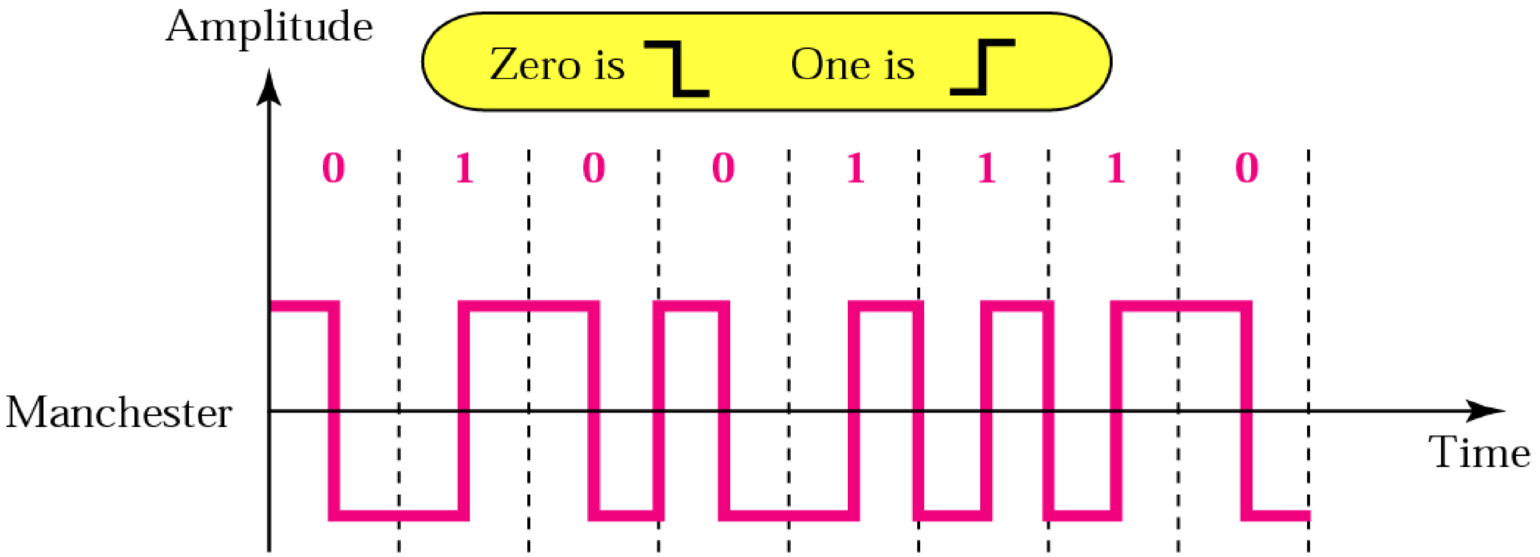
- The signal changes not between bits but during each bit
  → More bandwidth





4.1 Manchester Encoding

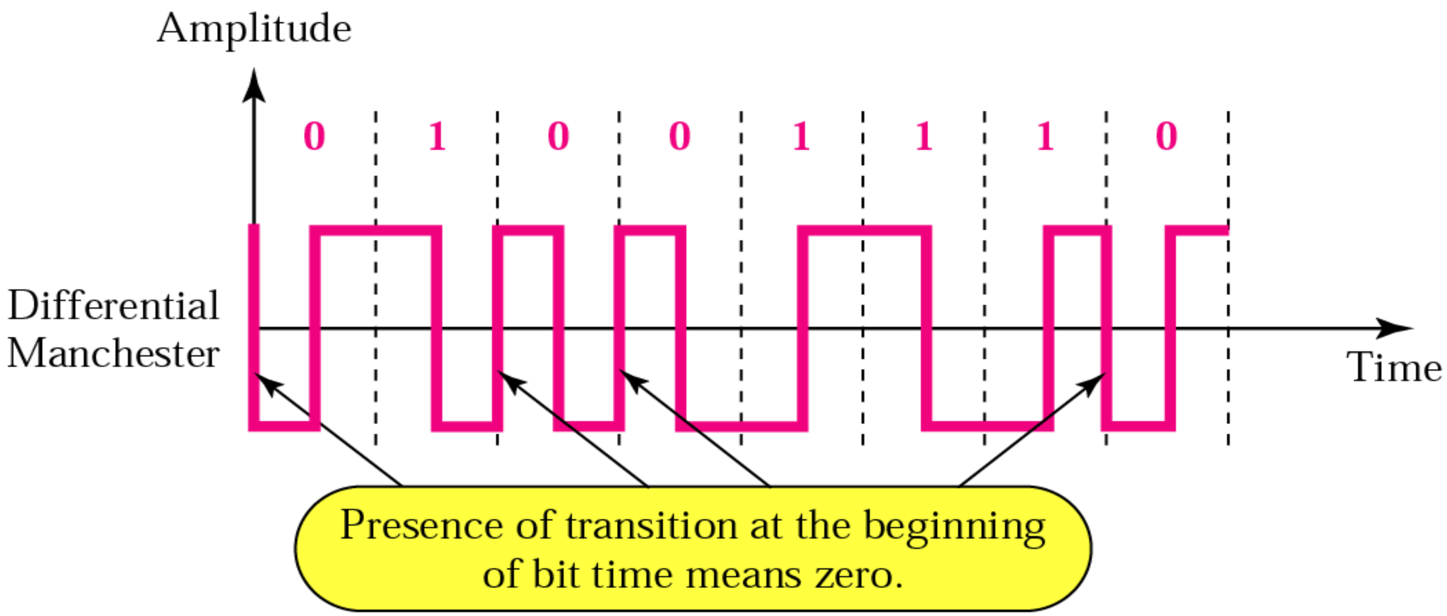
uses an inversion at the middle of each bit interval for both synchronization and bit representation.



4.1 Differential Manchester Encoding

In differential Manchester encoding, the transition at the middle of the bit is used only for synchronization.

The bit representation is defined by the inversion or noninversion at the beginning of the bit.

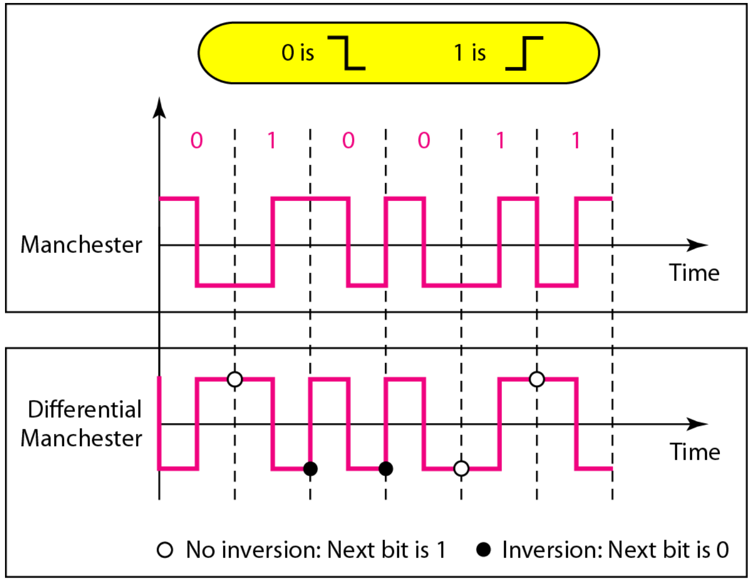


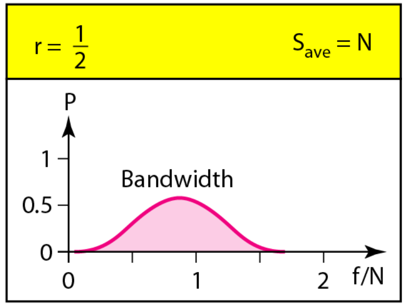
Manchester & Differential Manchester

- No baseline wandering
- No DC component
- Good Synchronization

Only drawback is the signal rate

- Because there is always one transition at the middle of the bit





uses three voltage levels

- **positive, negative, zero.**
  - **zero level : binary 0**
  - **positive and negative voltage : 1(alternate)**

AMI (Alternate Mark Inversion)

- **the simplest type of bipolar encoding**
- **AMI changes poles with every 1 it encounters**
  - Make DC component 0
  - These changes provide the synchronization needed by the receiver
- **But, there is no mechanism to ensure the synchronization of a long string of 0s**

Pseudo ternary

- The 1 bit is encoded as a zero voltage and the 0 is encoded as alternating positive and negative voltages.

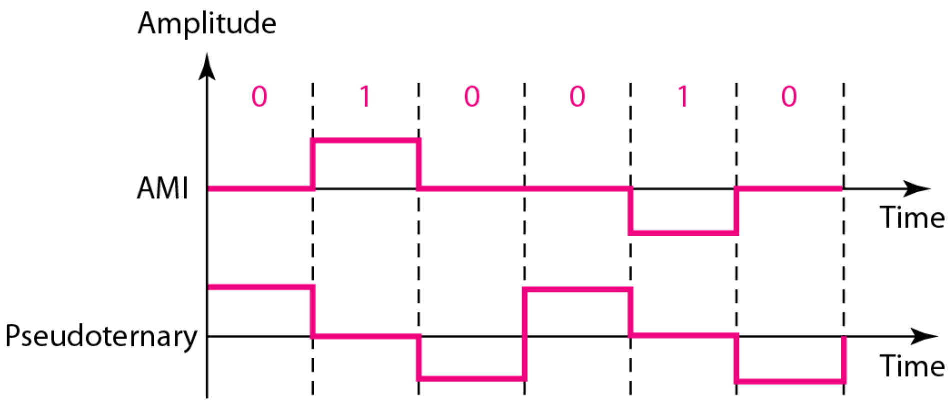
No DC Component

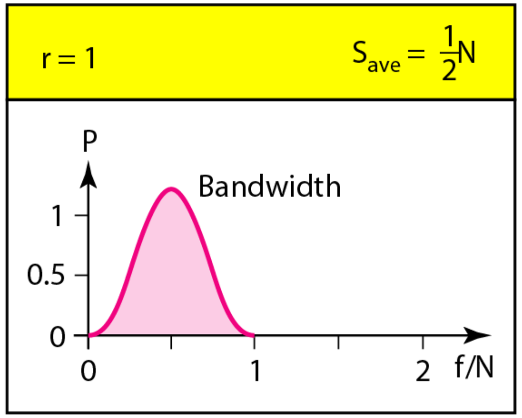
Same signal rate with NRZ

Energy around frequency N/2

Synchronization problem when long 0s

Commonly used for a long distance communication



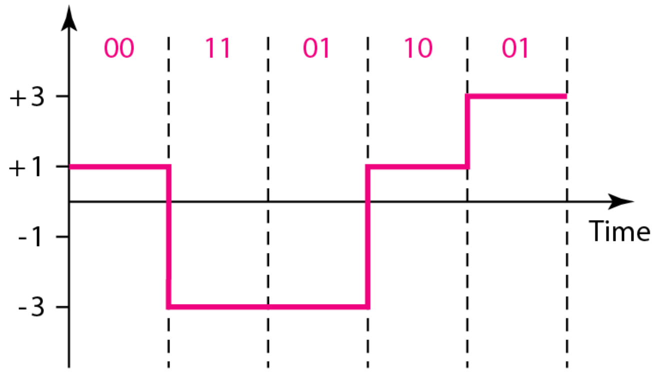


2B1Q (two binary one quaternary)

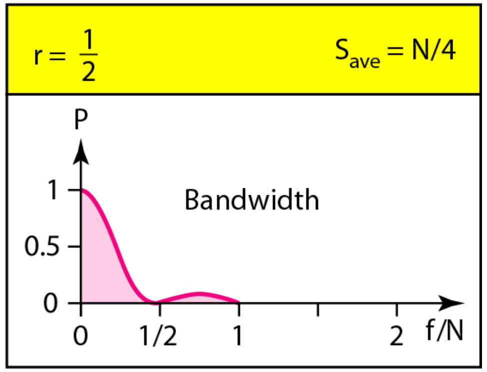
- uses four voltage levels
- Each pulse can represent 2bits

	Previous level: positive	Previous level: negative
Next bits	Next level	Next level
00	+1	-1
01	+3	-3
10	-1	+1
11	-3	+3

Transition table

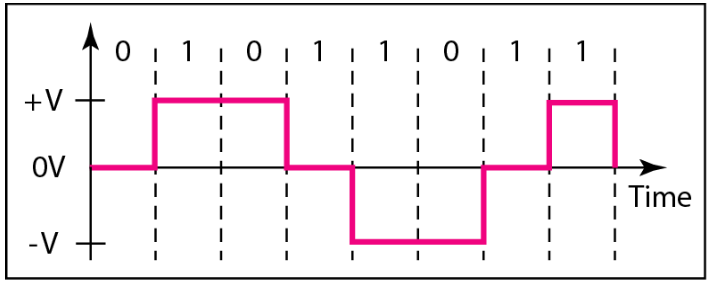


Assuming positive original level

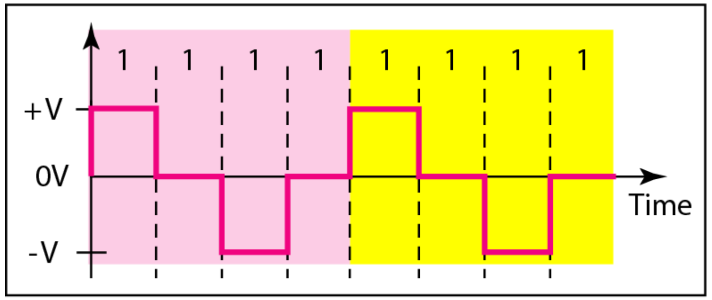


MLT-3 (Multiline Transmission, three level)

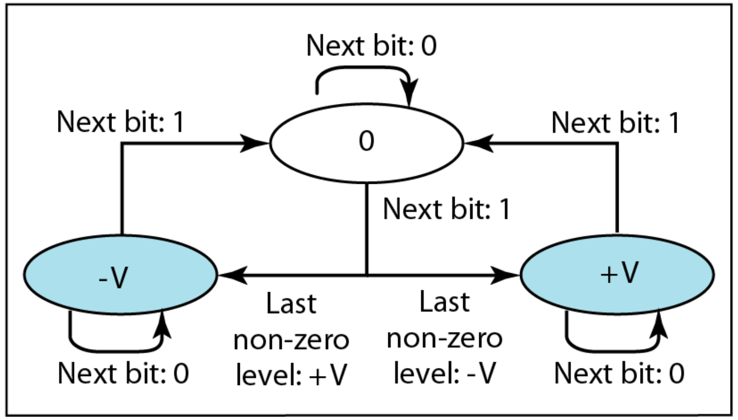
- similar to NRZ-I
- uses three level of signals (+1, 0, -1)



a. Typical case



b. Worse case



c. Transition states

4.1 Summary of Line Coding Scheme

Category	Scheme	Bandwidth (average)	Characteristics
Unipolar	NRZ	B=N/2	Costly, no self-synchronization if long 0s or 1s, DC
Unipolar	NRZ-L	B=N/2	No self-synchronization if long 0s or 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
Multilevel	2B1Q	B=N/4	No self-synchronization for long same double bits
	8B6T	B=3N/4	Self-synchronization, no DC
	4D-PAM5	B=N/8	Self-synchronization, no DC
Multiline	MLT-3	B=N/3	No self-synchronization for long 0s

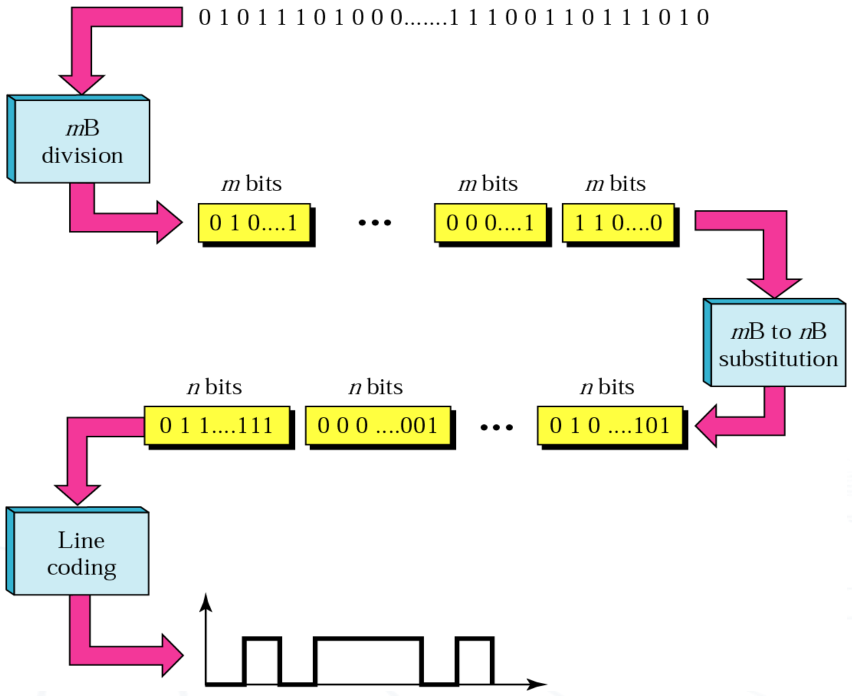
4.1 Block Coding

Block coding was introduced to improve the performance of the line coding.

with some redundancies, block code can provide synchronization and error detection.

Block coding is normally referred to as mB/nB coding.

- it replaces each m-bit group with an n-bit group.



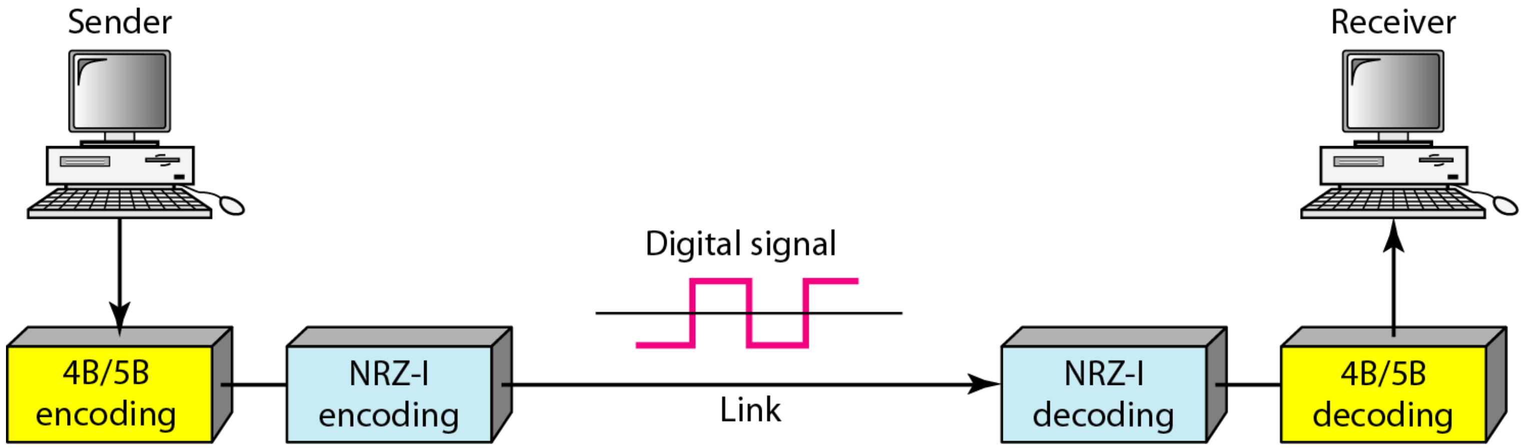
4.1 4Binary/5Binary

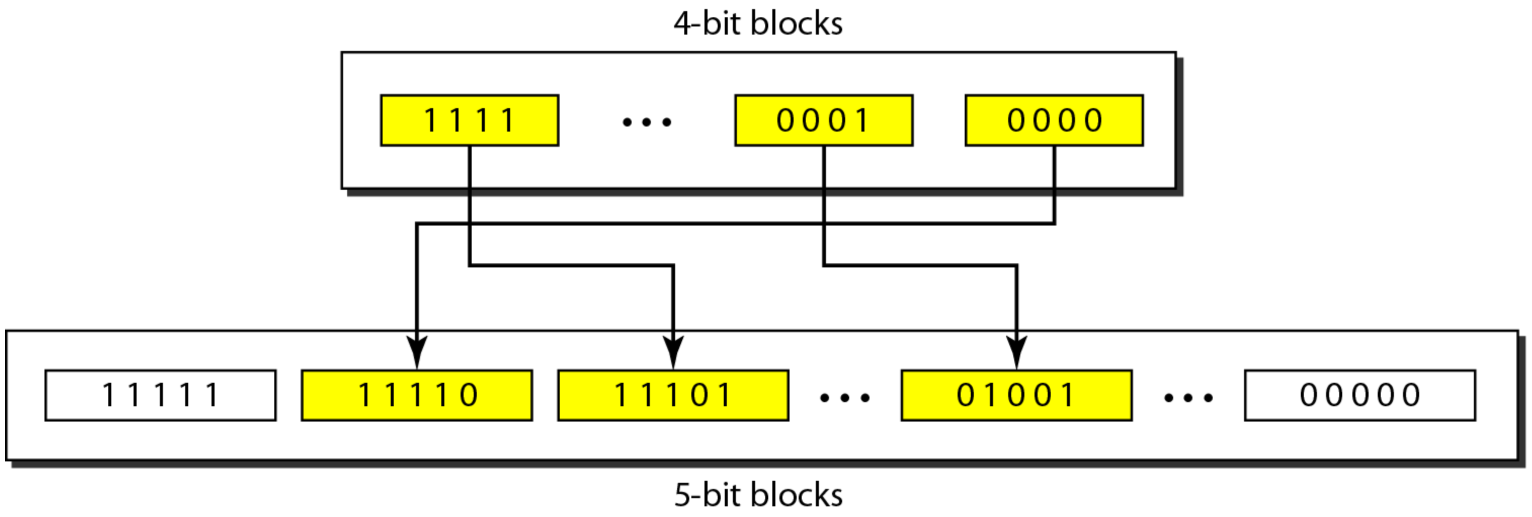
Designed to be used in combination with NRZ-I.

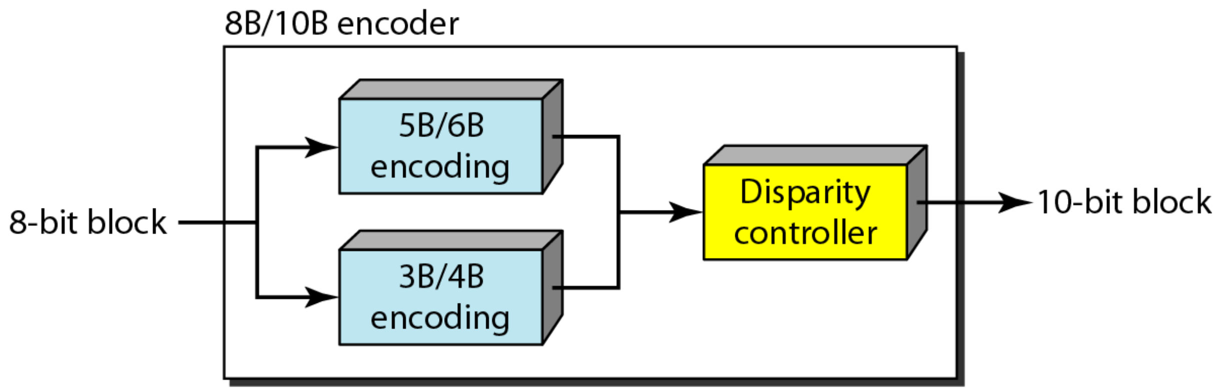
A long sequence of 0s can make the receiver clock lose synchronization.

Block coded stream does not have more than three consecutive 0s.

Some of the unused groups are used for control purpose;
the others are not used at all.




4-bit blocks
1111
• • •
0001
0000
11111
11110
11101
• • •
01001
• • •
00000
5-bit blocks


8B/10B encoder
8-bit block
5B/6B encoding
3B/4B encoding
Disparity controller
10-bit block

4.1 4B/5B Block Codes

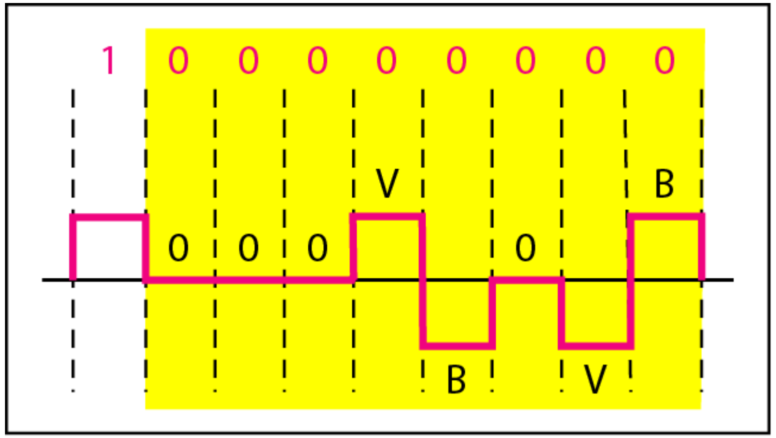
Data Sequence	Encoded Sequence	Control Sequence	Encoded Sequence
0000	11110	Q (Quiet)	00000
0001	01001	I (Idle)	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11010		
1101	11011		
1110	11100		
1111	11101		

To send the data, the encoded data has no DC component and relatively narrow bandwidth.

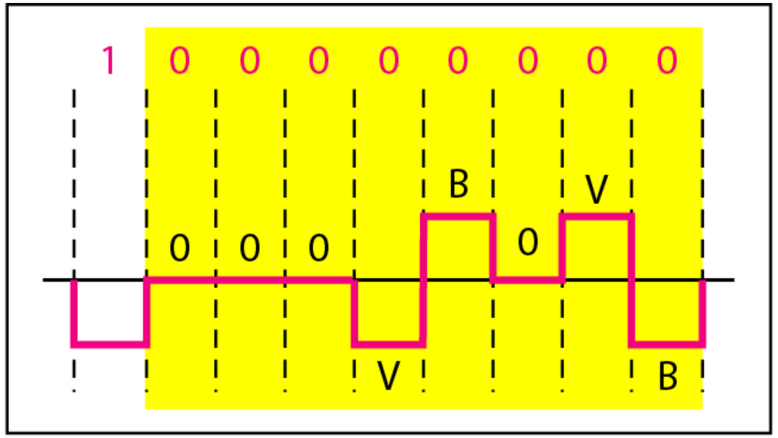
If we can find a way to avoid a long sequence of 0s in the original stream, we can use bipolar AMI for long distance.

B8ZS (Bipolar with 8-zero substitution)

- 8 zero → 000VB0VB
- Does not change bit rate and balance the voltage level



a. Previous level is positive.



b. Previous level is negative.

4.2 Analog-to-Digital Conversion

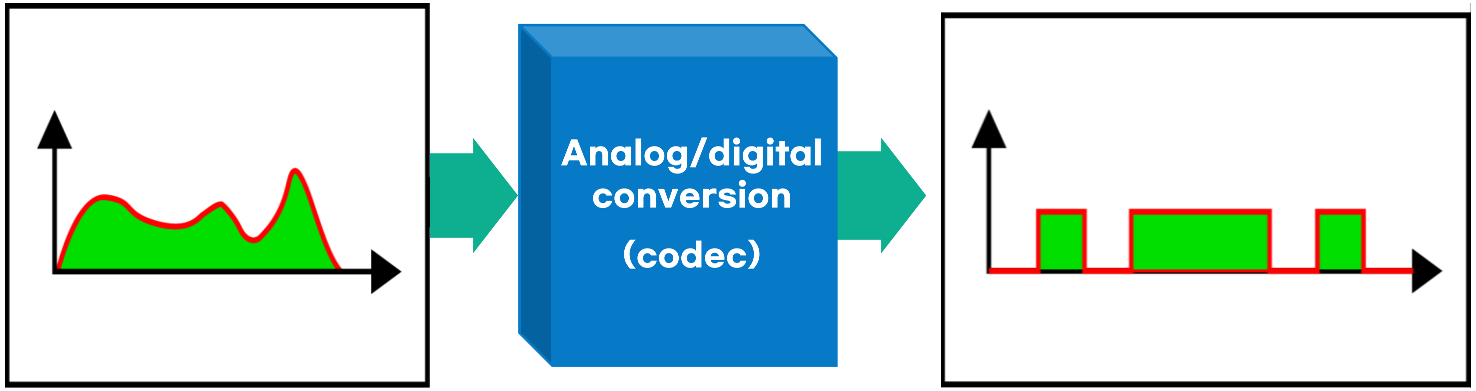
The process changing analog to digital data.

the representation of analog information by a digital signal.

→ **Digitization**

- recording singer's voice onto a compact disc

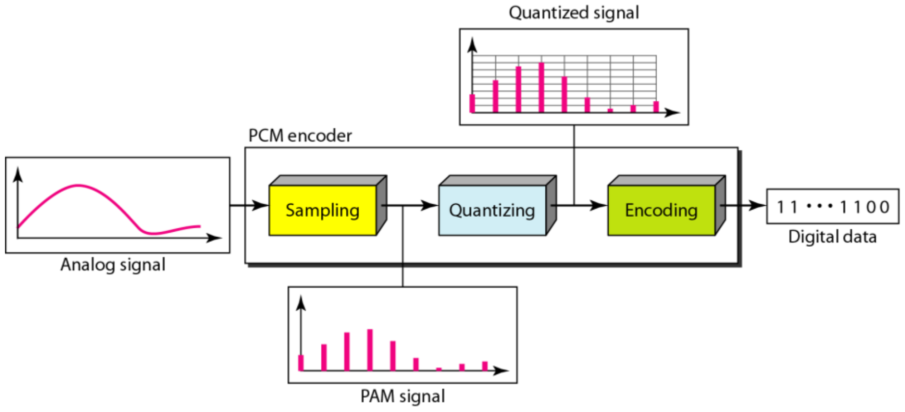
- **After the digital data are created, we can use one of techniques previously to convert the digital data to digital signal.**

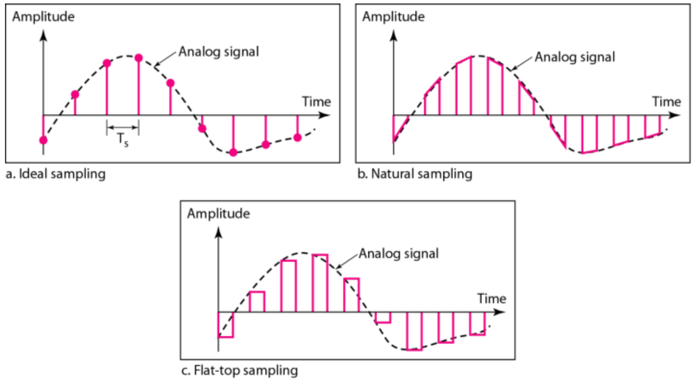


Sampling → PAM (Pulse Amplitude Modulation)

- The first step of PCM
- Term **sampling** means measuring the amplitude of the signal at **equal intervals.**
- takes analog information every Ts (fs = 1/Ts)
- Three sampling methods: Ideal, Natural and Flat-top

According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.





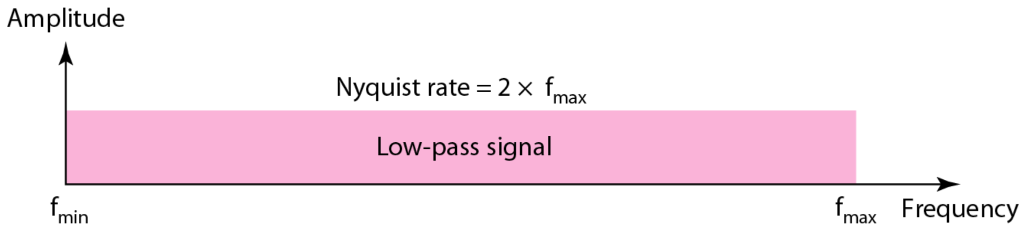
4.2 Sampling Requirements

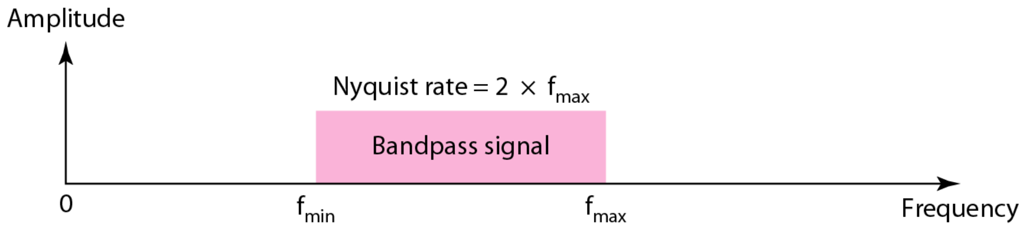
1. **We can sample a signal only if the signal is band limited.**
2. **Sampling rate must be at least 2 times the highest frequency, not the bandwidth.**

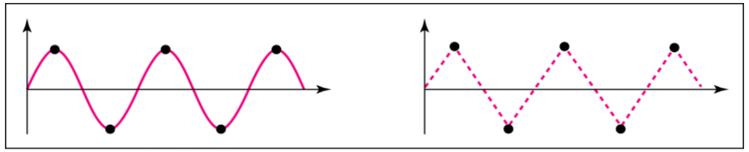
Ex. 4.6

- For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: fs = 4f (2 times the Nyquist rate), fs = 2f (Nyquist rate), and fs = f (one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.
- It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave (part a). Oversampling in part b can also create the same approximation, but it is redundant and unnecessary. Sampling below the Nyquist rate (part c) does not produce a signal that looks like the original sine wave.

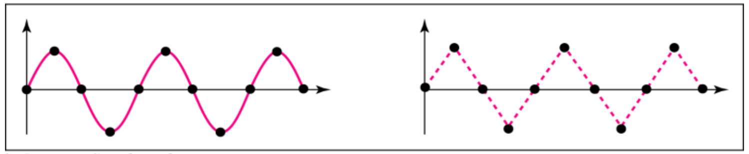
4.2 Sampling Requirements



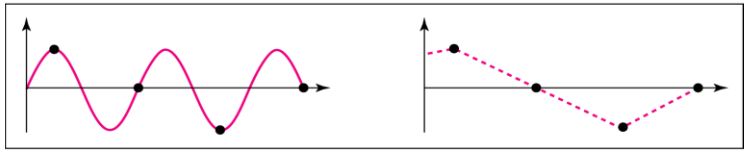




a. Nyquist rate sampling: $f_s = 2\,f$



b. Oversampling: $f_s = 4\,f$



c. Undersampling: $f_s = f$

Ex. 4.10

- A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?
- The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

Ex. 4.11

- A complex bandpass signal has a bandwidth of 200 kHz.
  What is the minimum sampling rate for this signal?
- We cannot find the minimum sampling rate in this case because we do not know where the bandwidth starts or ends. We do not know the maximum frequency in the signal.

4.2 Quantization

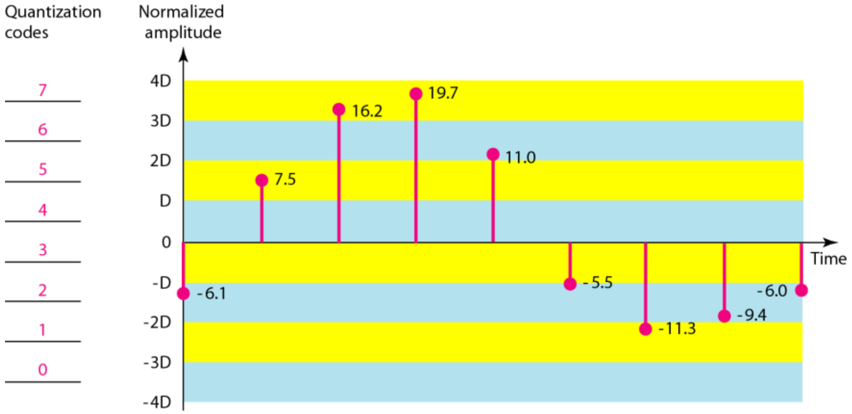
We assume that the original signal has instantaneous amplitude between V_{min} and V_{max}.

We divide the range into L zones, each of height Δ (delta).

- $\Delta = (V_{max} - V_{min})/L$

We assign quantized values of 0 to L-1 to the midpoint of each zone.

We approximate the value of the sample amplitude to the quantized values.



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 Level

- Depends on the range of the analog signal and how accurately we need to recover the signal.
- For audio, 256. For video, thousands.

Quantization Error

- The difference between the quantized signal and the original signal.
- $-\Delta/2 < \text{error} < \Delta/2$

Uniform and Non-Uniform Quantization

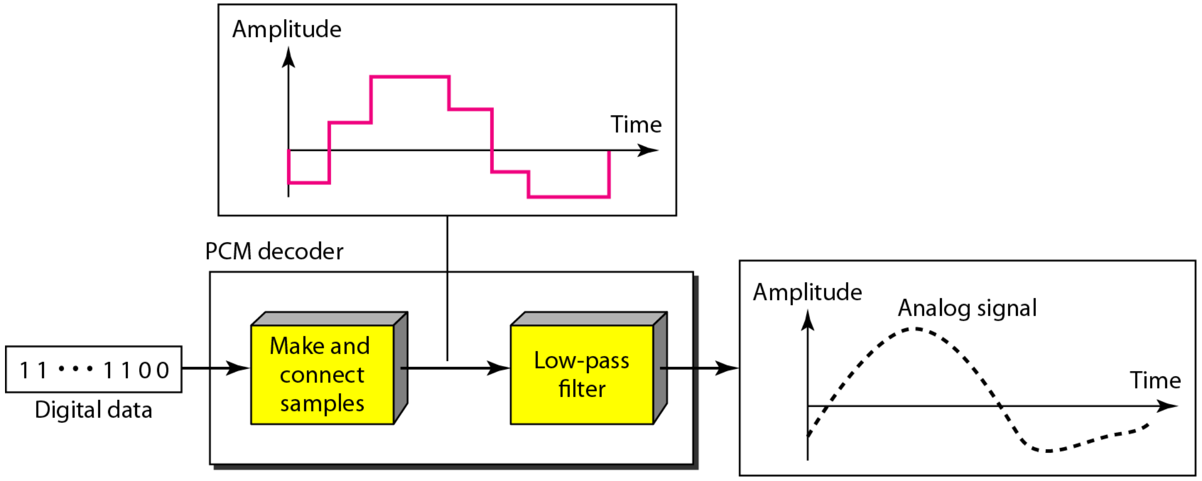
- Changes in amplitude often occur more frequently in lower amplitudes than in the higher ones.
- Non-Uniform quantization is achieved by two processes: **Companding and Expanding**.

Encoding

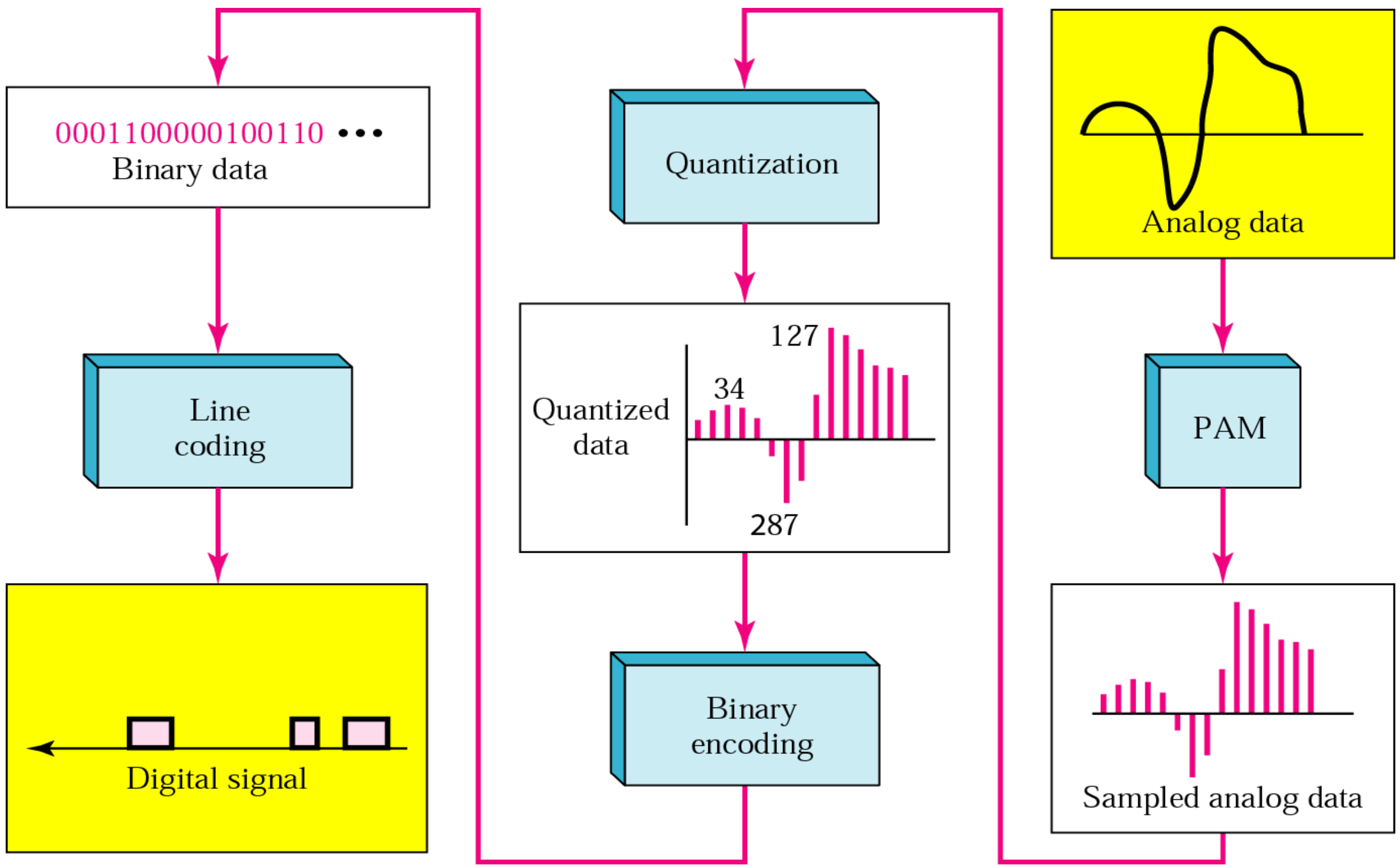
- After each sample is quantized and the number of bits per sample is decided, each sample can be changed to an n_b-*bit* code word.
- **Bit Rate = Sample rate x number of bits per sample = $f_s x\ n_b$**

Original Signal Recovery

- First change the code words into a pulse.
- Make the staircase signal into an analog signal by passing through a low pass filter.



4.2 From analog signal to digital signal



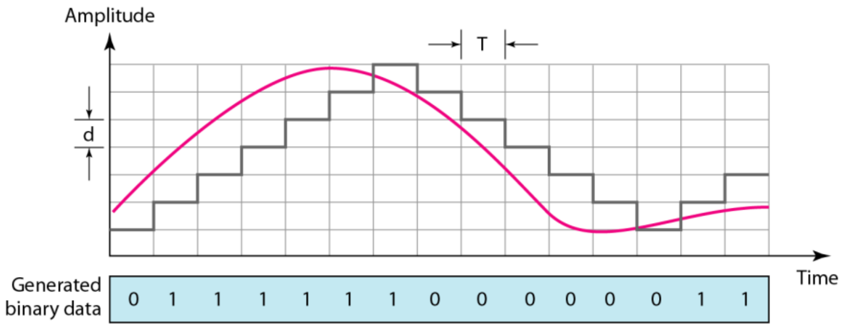
PCM is complex technique. We want simple one.

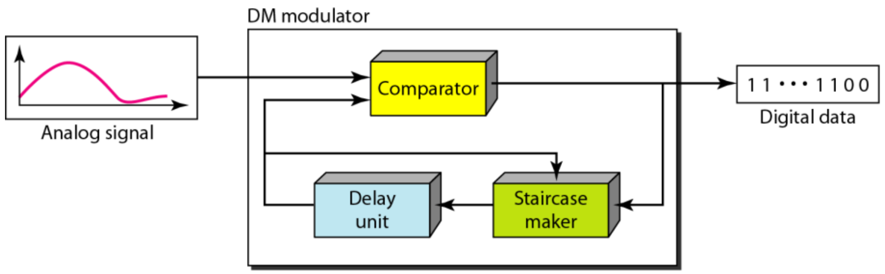
Modulator

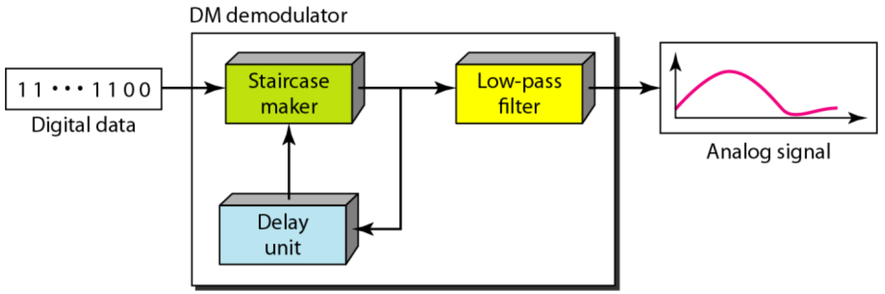
- Compares the value of the analog signal with the last value of the staircase signal.
- If the amplitude of the signal is larger, the staircase signal goes δ up; otherwise, δ down.

Demodulator

- Takes the digital data and creates the analog signal by using the staircase maker and the delay unit.

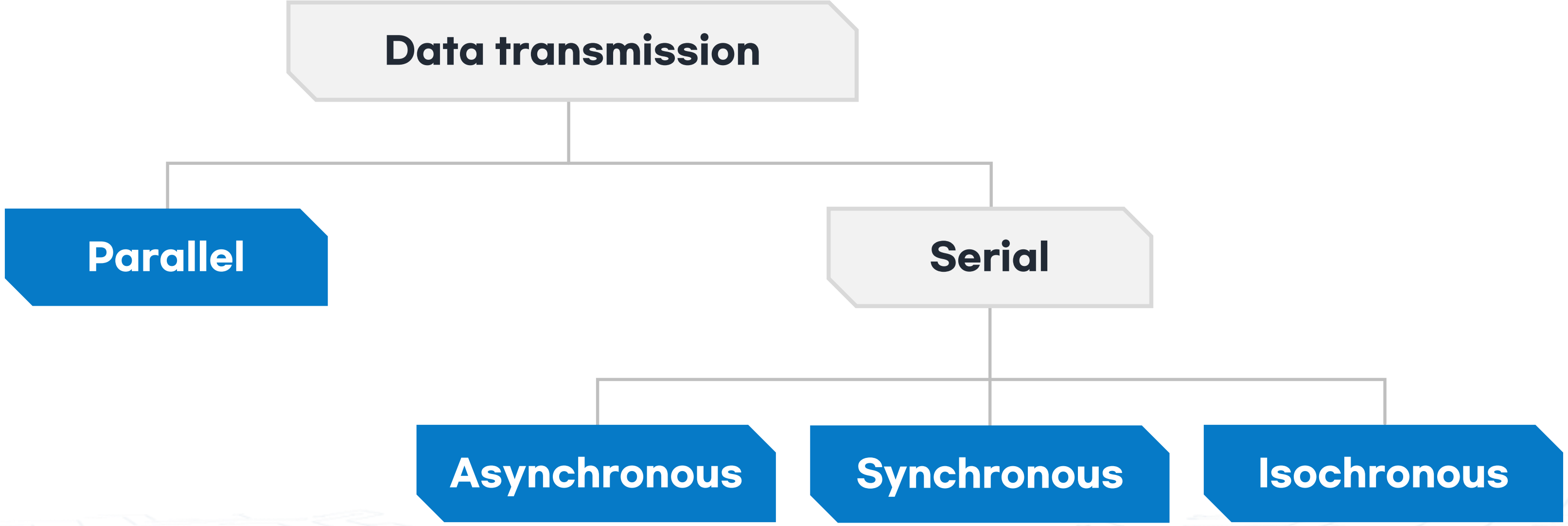






When we are considering the wiring, the data stream is of the primary concern.

Do we send 1 bit a time? Or do we group bits into larger groups?



4.3 Parallel Transmission

By grouping data, we can send data n bits at a time instead of 1.

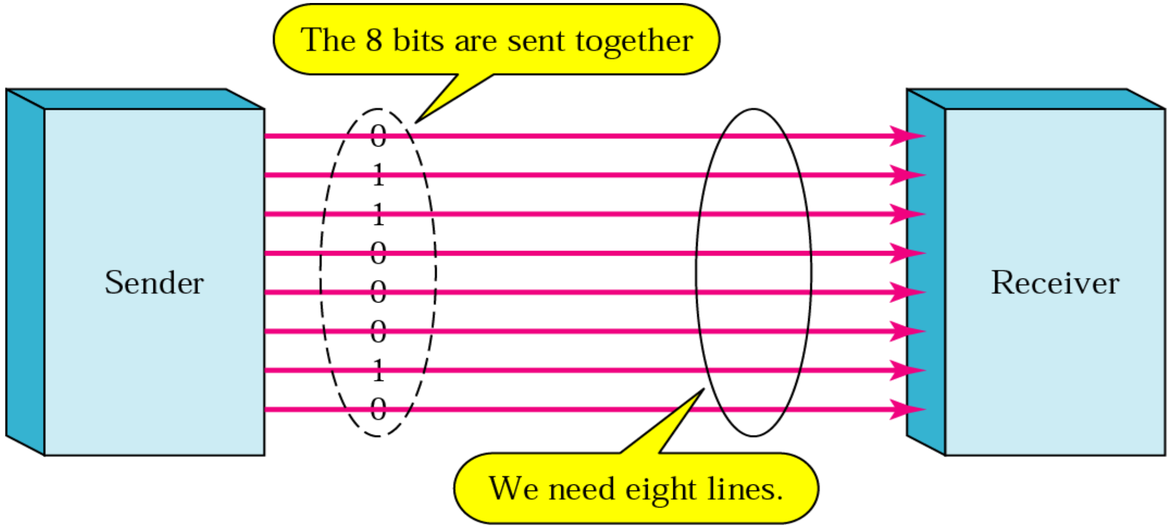
Use n wires instead of 1 wire.

Advantages : **Speed**

- Increase transmission speed by n times

Disadvantages : **Cost**

- Needs n communication wires



4.3 Serial Transmission

Asynchronous Transmission

- Timing is unimportant
- In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. There may be a gap between each byte.

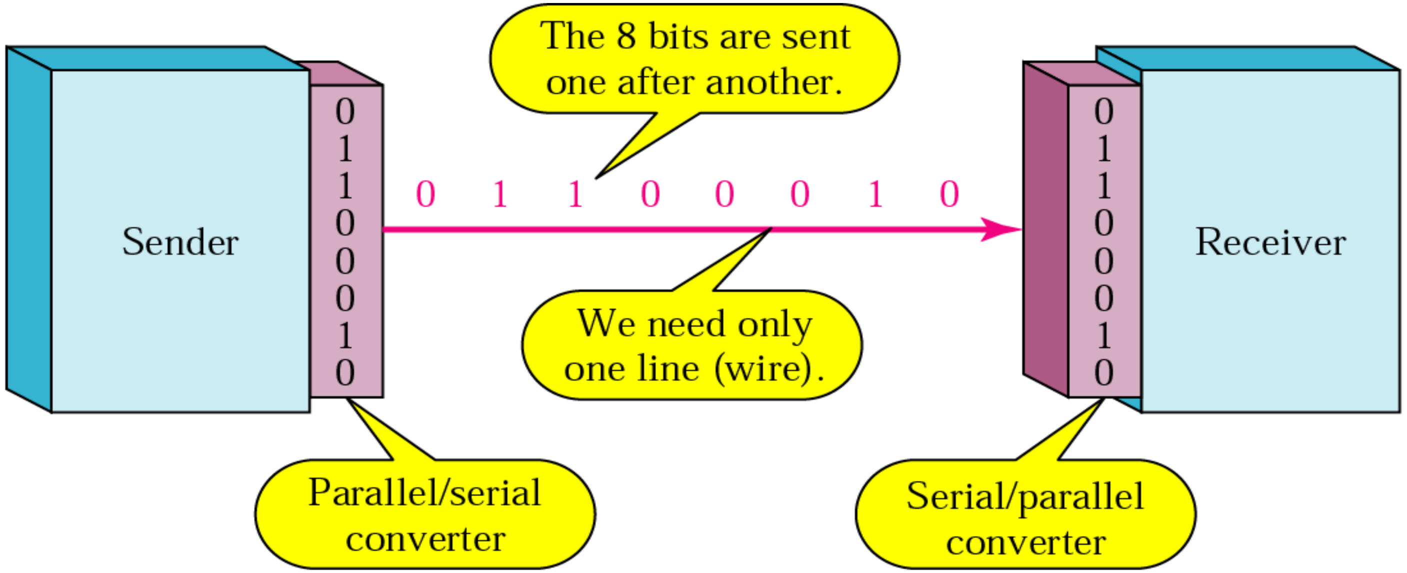
Synchronous Transmission

- Bit stream is combined into longer frames.
- In synchronous transmission, we send bits one after another without start/stop bits or gaps.
- It is the responsibility of the receiver to group the bits.
- Timing is very important~!

Isochronous Transmission

- Even delay between frames are highly required. Ex. TV, Real-time App. Etc.

4.3 Serial Transmission



4.3 Serial Transmission