

Forouzan

Chapter 4 Digital Transmission

4-1 DIGITAL-TO-DIGITAL CONVERSION

In this section, we see how we can 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.

Topics discussed in this section:

Line Coding
Line Coding Schemes
Block Coding
Scrambling

Figure 4.1 Line coding and decoding

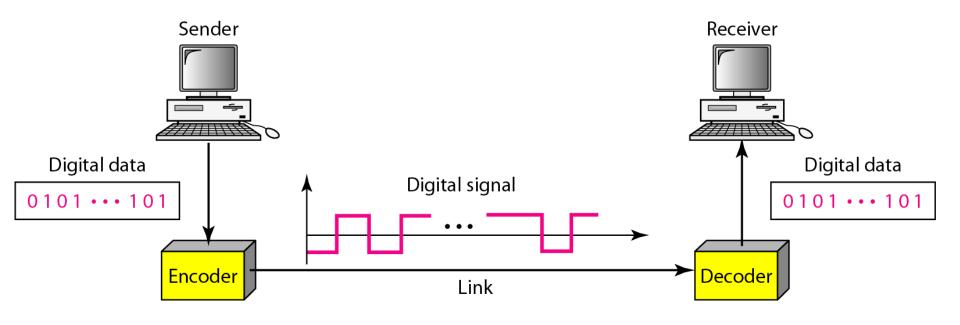
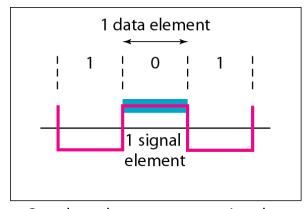
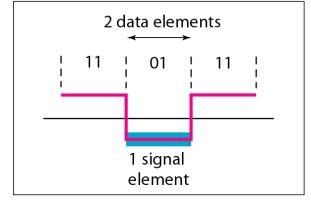


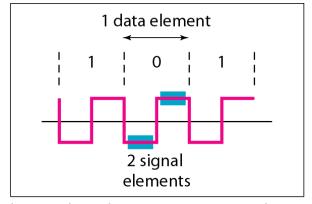
Figure 4.2 Signal element versus data element



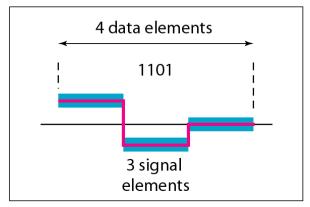
a. One data element per one signal element (r = 1)



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



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



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

Example 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?

Solution

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 bandwidth is finite.

Example 4.2

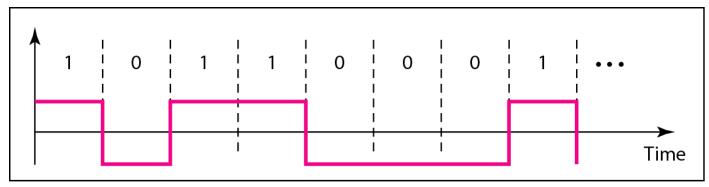
The maximum data rate of a channel (see Chapter 3) is $N_{max} = 2 \times B \times \log_2 L$ (defined by the Nyquist formula). Does this agree with the previous formula for N_{max} ?

Solution

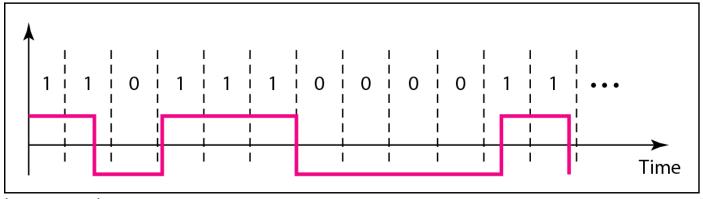
A signal with L levels actually can carry log_2L bits per level. If each level corresponds to one signal element and we assume the average case (c = 1/2), then we have

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

Figure 4.3 Effect of lack of synchronization



a. Sent



b. Received

Example 4.3

In a digital transmission, the receiver clock is 0.1 percent faster than the sender clock. How many extra bits per second does the receiver receive if the data rate is 1 kbps? How many if the data rate is 1 Mbps?

Solution

At 1 kbps, the receiver receives 1001 bps instead of 1000 bps.

1000 bits sent 1001 bits received 1 extra bps

At 1 Mbps, the receiver receives 1,001,000 bps instead of 1,000,000 bps.

1,000,000 bits sent 1,001,000 bits received 1000 extra bps

Figure 4.4 Line coding schemes

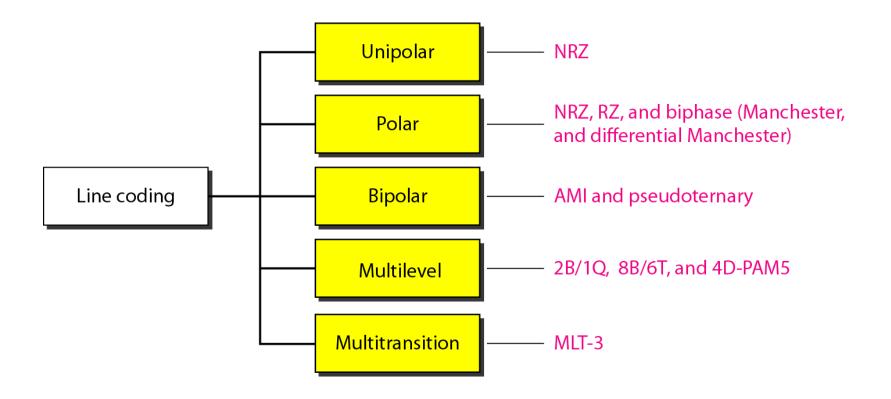
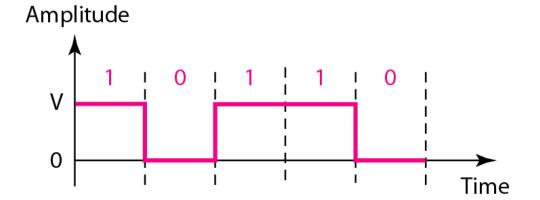


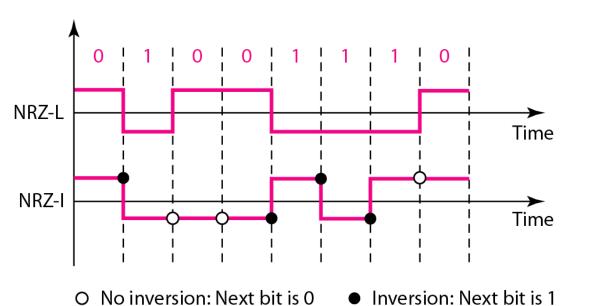
Figure 4.5 Unipolar NRZ scheme

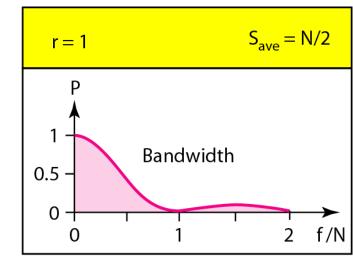


$$\frac{1}{2}V^2 + \frac{1}{2}(0)^2 = \frac{1}{2}V^2$$

Normalized power

Figure 4.6 Polar NRZ-L and NRZ-I schemes





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.



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



NRZ-L and NRZ-I both have a DC component problem.

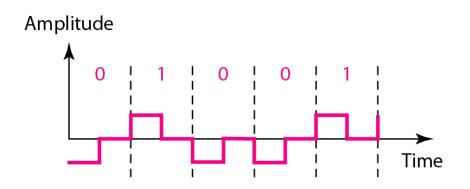
Example 4.4

A system is using NRZ-I to transfer 10-Mbps data. What are the average signal rate and minimum bandwidth?

Solution

The average signal rate is S = N/2 = 500 kbaud. The minimum bandwidth for this average baud rate is $B_{min} = S = 500$ kHz.

Figure 4.7 Polar RZ scheme



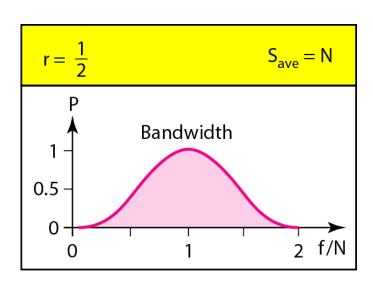
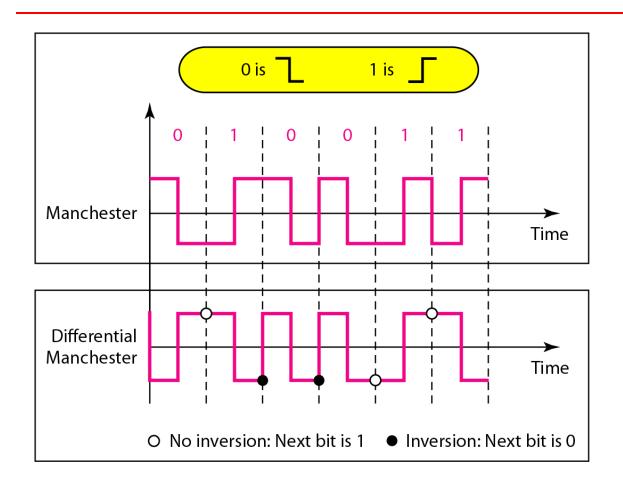
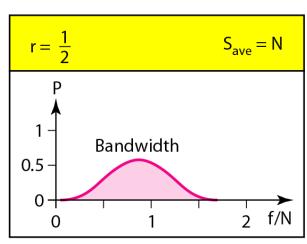


Figure 4.8 Polar biphase: Manchester and differential Manchester schemes





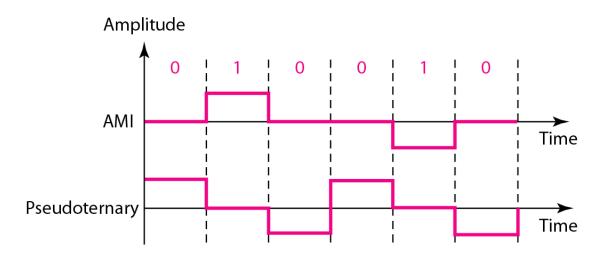
In Manchester and differential Manchester encoding, the transition at the middle of the bit is used for synchronization.

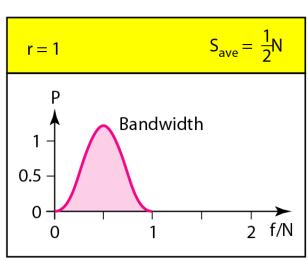
The minimum bandwidth of Manchester and differential Manchester is 2 times that of NRZ.



In bipolar encoding, we use three levels: positive, zero, and negative.

Figure 4.9 Bipolar schemes: AMI and pseudoternary





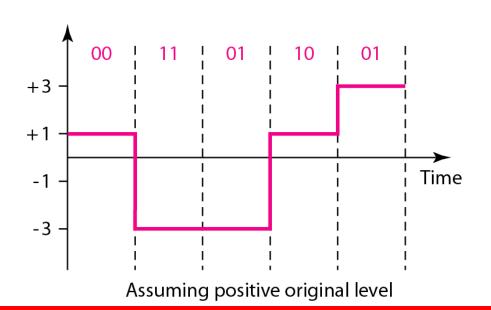
In *m*B*n*L schemes, a pattern of *m* data elements is encoded as a pattern of *n* signal elements in which 2^m ≤ Lⁿ.

Figure 4.10 Multilevel: 2B1Q scheme

Previous level:	Previous level:	
positive	negative	

Next bits	Next level	Next level
00	+1	-1
01	+3	-3
10	-1	+1
11	-3	+3

Transition table



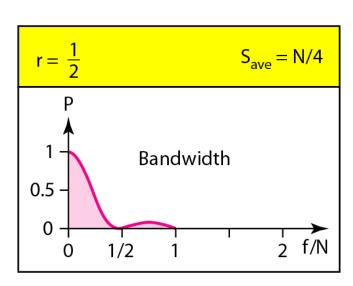


Figure 4.11 Multilevel: 8B6T scheme

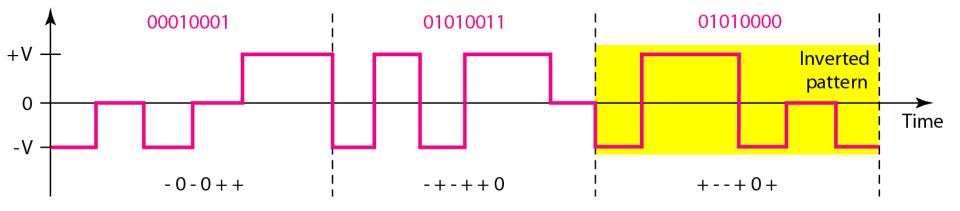


Figure 4.12 Multilevel: 4D-PAM5 scheme

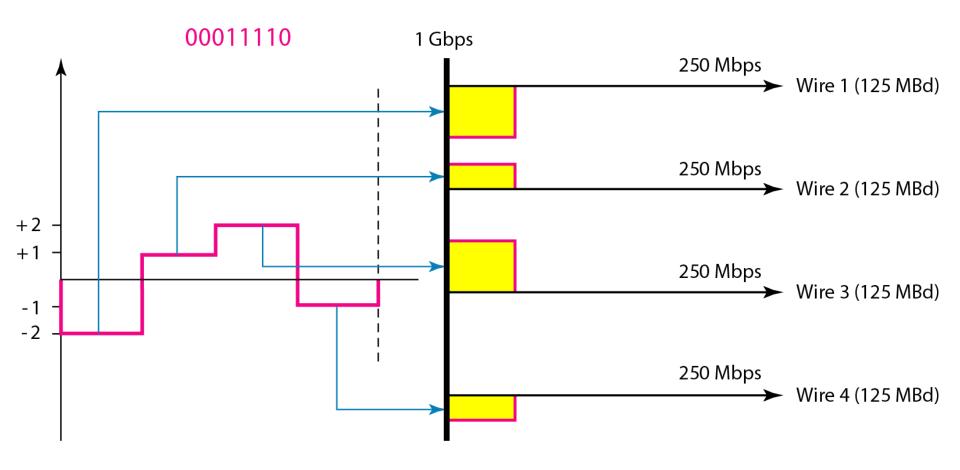
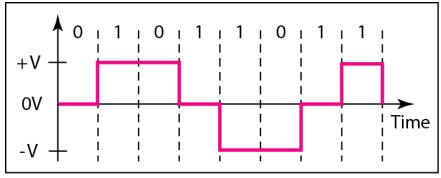
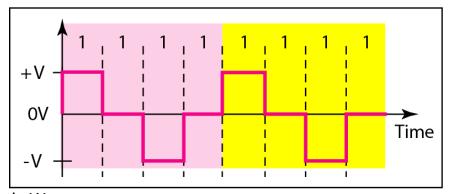


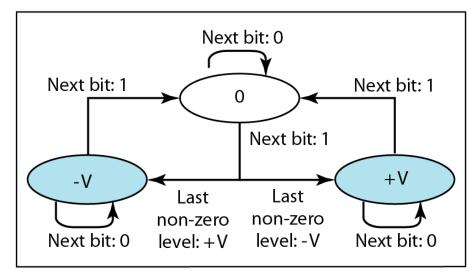
Figure 4.13 Multitransition: MLT-3 scheme



a. Typical case



b. Worse case



c. Transition states

 Table 4.1
 Summary of line coding schemes

Category	Scheme	Bandwidth (average)	Characteristics	
Unipolar	NRZ	B = N/2	Costly, no self-synchronization if long 0s or 1s, DC	
	NRZ-L	B = N/2	No self-synchronization if long 0s or 1s, DC	
Unipolar	NRZ-I $B = N/2$ No self-synchronization for long 0s, DC		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	
2B1Q B = N/4 No self-synchronic		No self-synchronization for long same double bits		
Multilevel	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	



Block coding is normally referred to as mB/nB coding; it replaces each m-bit group with an n-bit group.

Figure 4.14 Block coding concept

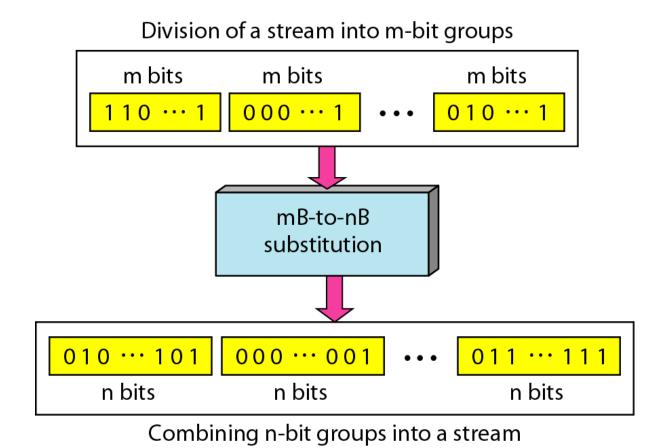


Figure 4.15 Using block coding 4B/5B with NRZ-I line coding scheme

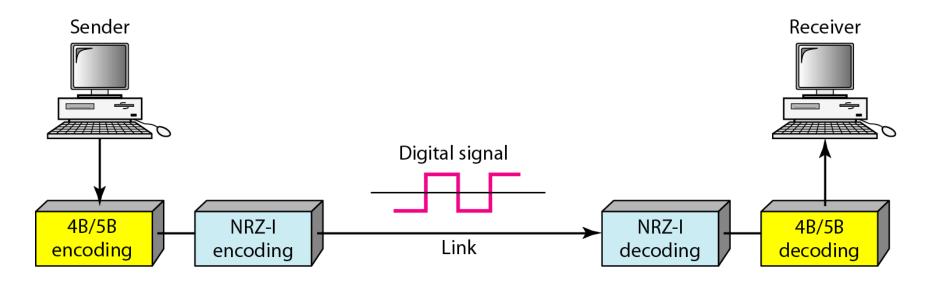
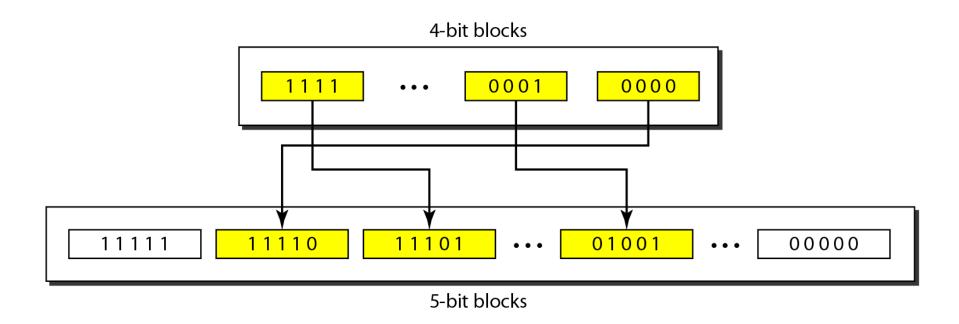


Table 4.2 4B/5B mapping codes

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		

Figure 4.16 Substitution in 4B/5B block coding



Example 4.5

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?

Solution

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 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.

Figure 4.17 8B/10B block encoding

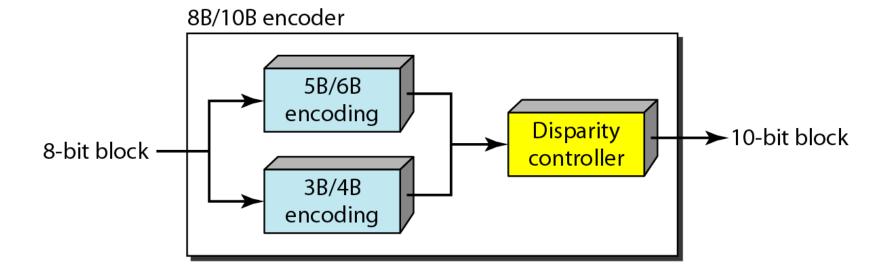


Figure 4.18 AMI used with scrambling

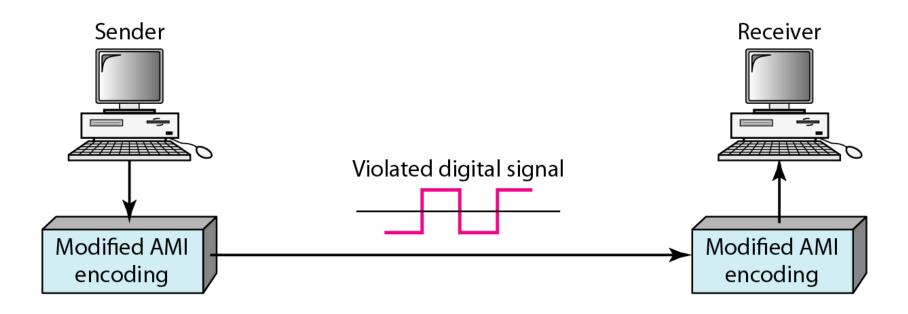
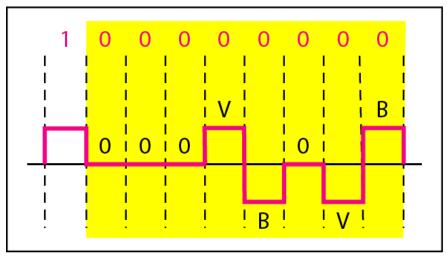
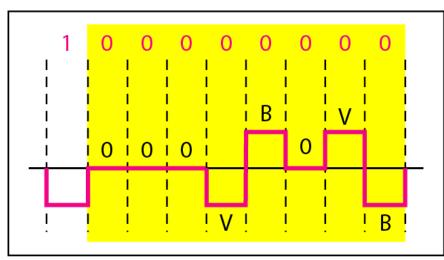


Figure 4.19 Two cases of B8ZS scrambling technique



a. Previous level is positive.



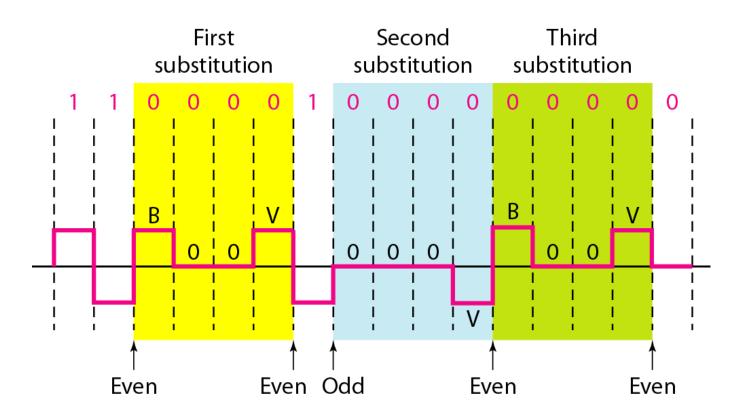
b. Previous level is negative.



Note

B8ZS substitutes eight consecutive zeros with 000VB0VB.

Figure 4.20 Different situations in HDB3 scrambling technique



Note

HDB3 substitutes four consecutive zeros with 000V or B00V depending on the number of nonzero pulses after the last substitution.

4-2 ANALOG-TO-DIGITAL CONVERSION

We have seen in Chapter 3 that a digital signal is superior to an analog signal. The tendency today is to change an analog signal to digital data. In this section we describe two techniques, pulse code modulation and delta modulation.

Topics discussed in this section:

Pulse Code Modulation (PCM) Delta Modulation (DM)

Figure 4.21 Components of PCM encoder

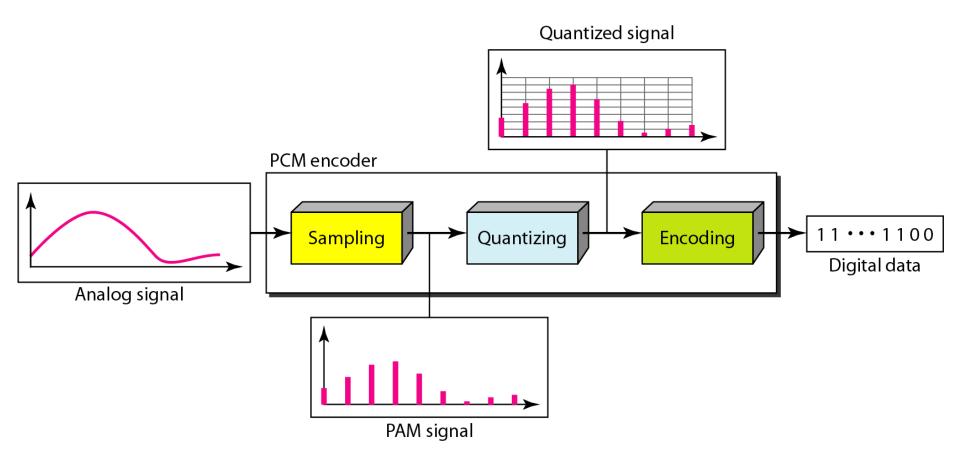
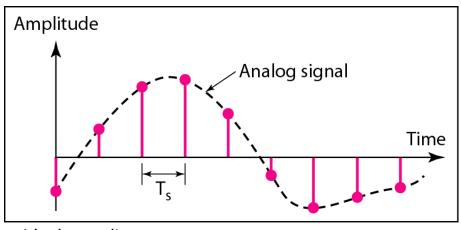
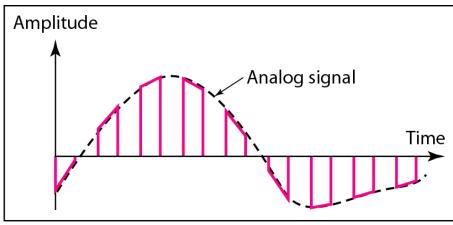


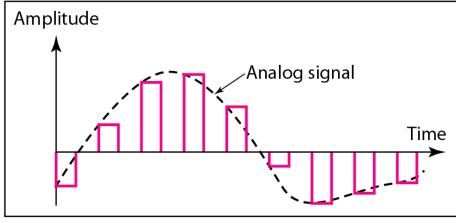
Figure 4.22 Three different sampling methods for PCM





a. Ideal sampling

b. Natural sampling

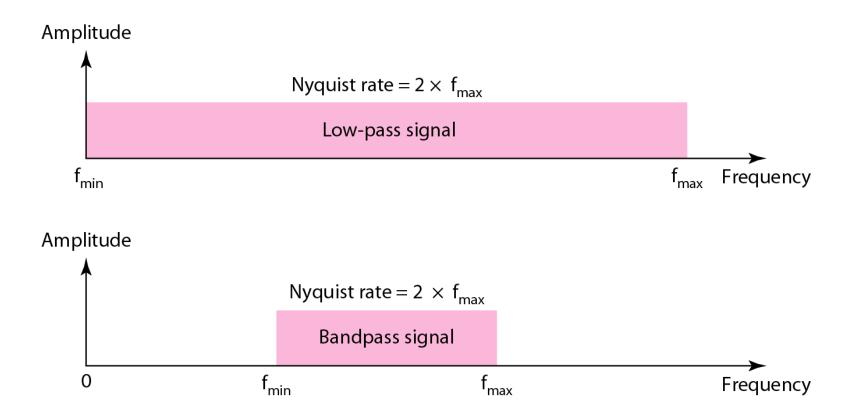


c. Flat-top sampling

Note

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

Figure 4.23 Nyquist sampling rate for low-pass and bandpass signals

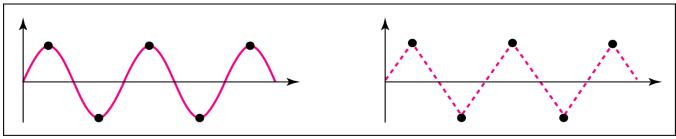


4.46

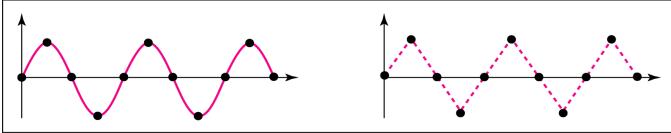
For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: $f_s = 4f$ (2 times the Nyquist rate), $f_s = 2f$ (Nyquist rate), and $f_s = f$ (one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.

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.

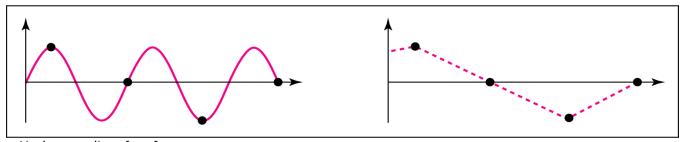
Figure 4.24 Recovery of a sampled sine wave for different sampling rates



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



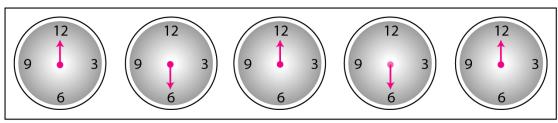
b. Oversampling: $f_s = 4 f$



c. Undersampling: $f_s = f$

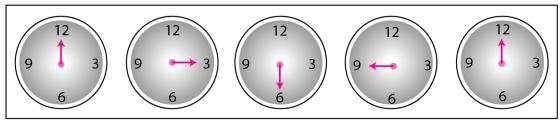
Consider the revolution of a hand of a clock. The second hand of a clock has a period of 60 s. According to the Nyquist theorem, we need to sample the hand every 30 s $(T_s = T \text{ or } f_s = 2f)$. In Figure 4.25a, the sample points, in order, are 12, 6, 12, 6, 12, and 6. The receiver of the samples cannot tell if the clock is moving forward or backward. In part 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 part c, we sample below the Nyquist rate $(T_s = T \text{ or } f_s = f)$. 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.

Figure 4.25 Sampling of a clock with only one hand



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): $T_s = T\frac{3}{4}$

An example related to Example 4.7 is the seemingly backward rotation of the wheels of a forward-moving car in a movie. This can be explained by under-sampling. A movie is filmed at 24 frames per second. If a wheel is rotating more than 12 times per second, the under-sampling creates the impression of a backward rotation.

Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz. The sampling rate therefore is 8000 samples per second.

A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

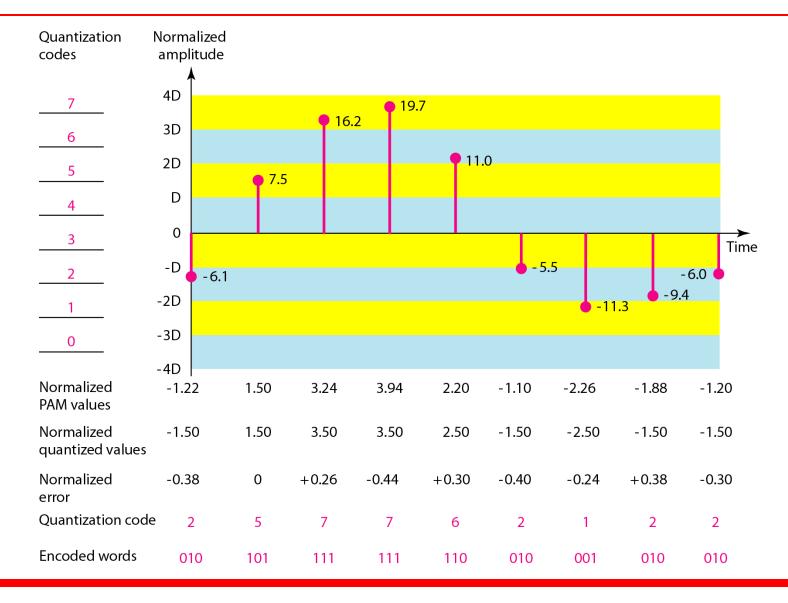
The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

A complex bandpass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

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.

Figure 4.26 Quantization and encoding of a sampled signal



What is the SNR_{dB} in the example of Figure 4.26?

Solution

We can use the formula to find the quantization. We have eight levels and 3 bits per sample, so

$$SNR_{dB} = 6.02(3) + 1.76 = 19.82 dB$$

Increasing the number of levels increases the SNR.

A telephone subscriber line must have an SNR_{dB} above 40. What is the minimum number of bits per sample?

Solution

We can calculate the number of bits as

$$SNR_{dB} = 6.02n_b + 1.76 = 40 \implies n = 6.35$$

Telephone companies usually assign 7 or 8 bits per sample.

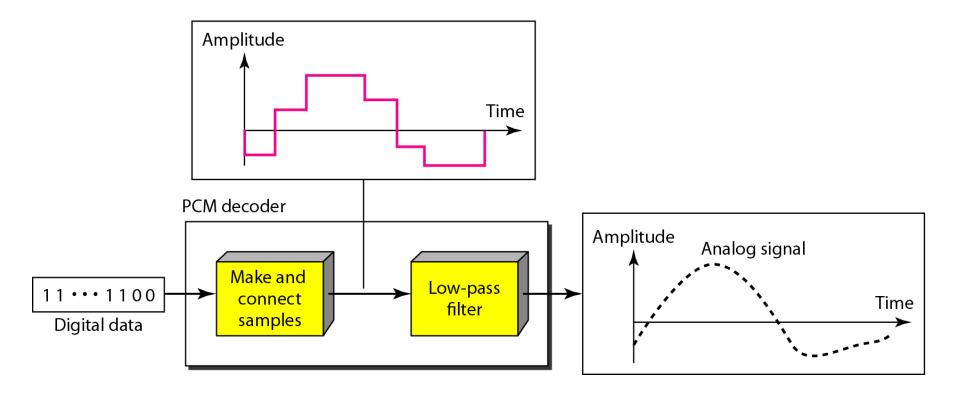
We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

Solution

The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

Sampling rate = $4000 \times 2 = 8000$ samples/s Bit rate = $8000 \times 8 = 64,000$ bps = 64 kbps

Figure 4.27 Components of a PCM decoder



We have a low-pass analog signal of $4 \, kHz$. If we send the analog signal, we need a channel with a minimum bandwidth of $4 \, kHz$. If we digitize the signal and send $8 \, kHz$ bits per sample, we need a channel with a minimum bandwidth of $8 \times 4 \, kHz = 32 \, kHz$.

Figure 4.28 The process of delta modulation

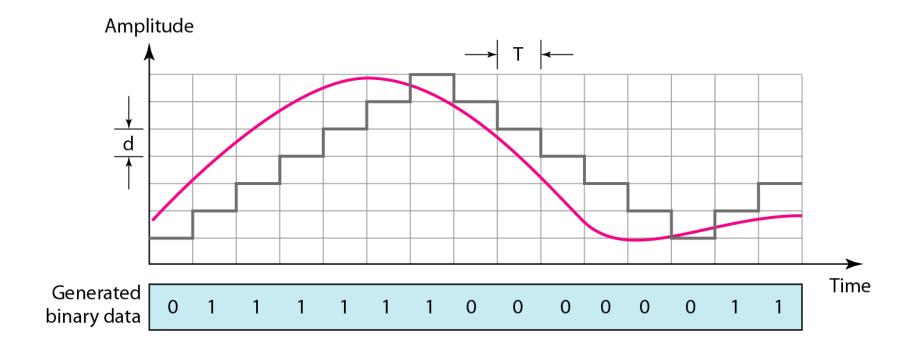


Figure 4.29 Delta modulation components

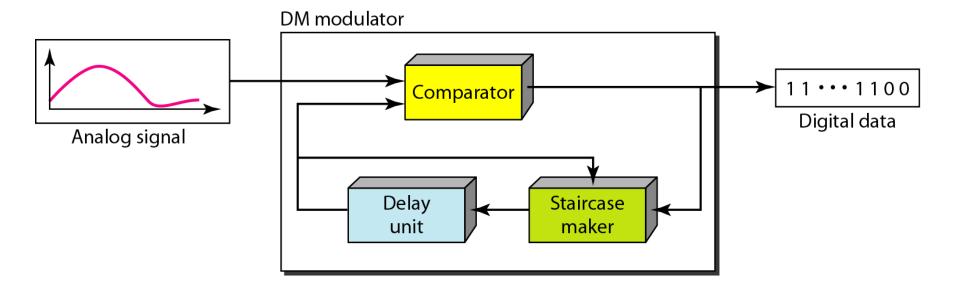
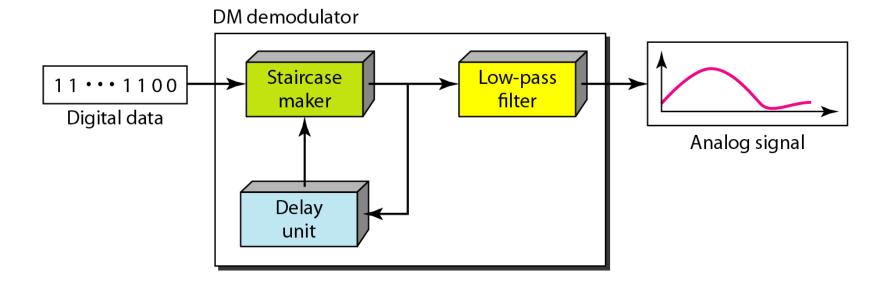


Figure 4.30 Delta demodulation components



4-3 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.

Topics discussed in this section:

Parallel Transmission Serial Transmission

Figure 4.31 Data transmission and modes

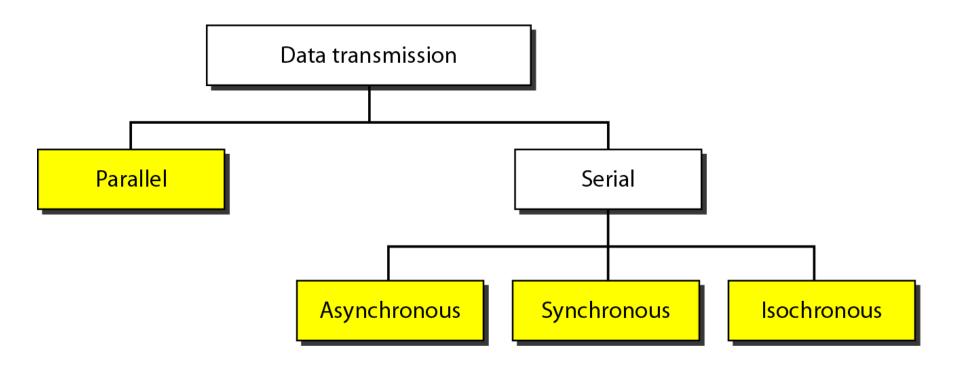


Figure 4.32 Parallel transmission

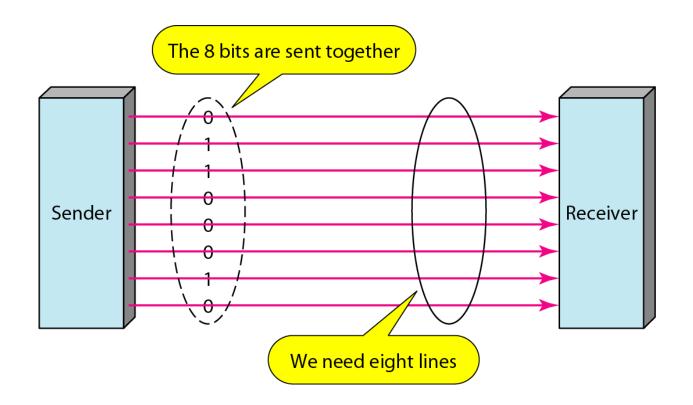
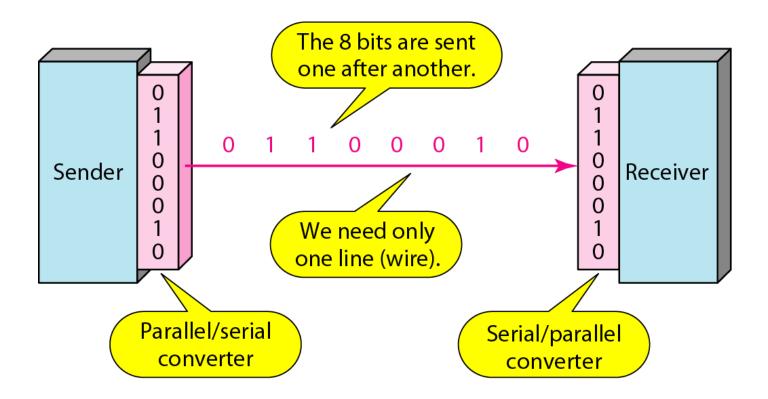


Figure 4.33 Serial transmission



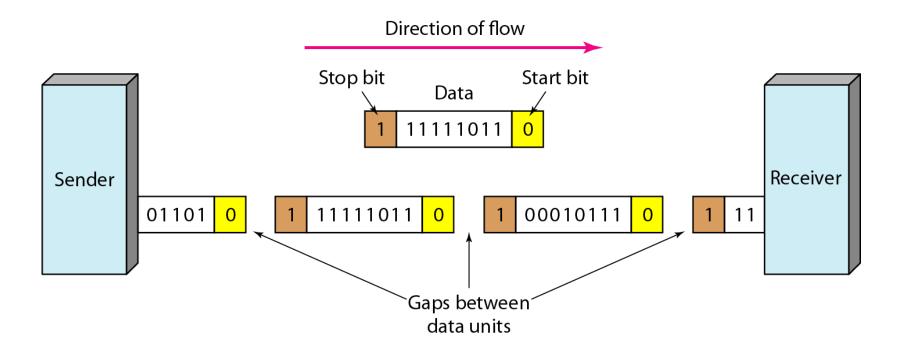
Note

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.

Note

Asynchronous here means "asynchronous at the byte level," but the bits are still synchronized; their durations are the same.

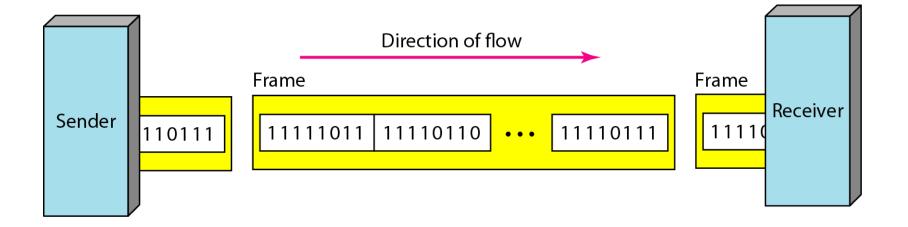
Figure 4.34 Asynchronous transmission



Note

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.

Figure 4.35 Synchronous transmission







Chapter 4 Digital Transmission

4-2 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 describe two techniques, pulse code modulation and delta modulation.

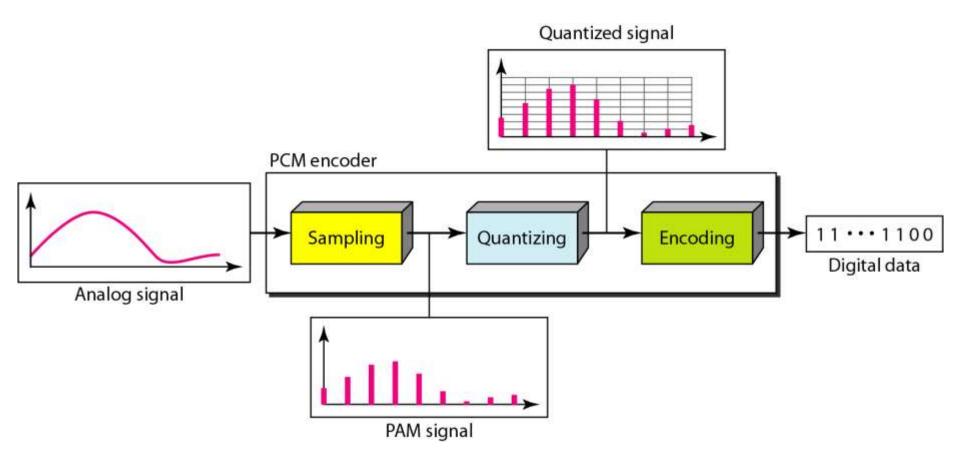
Topics discussed in this section:

- **Pulse Code Modulation (PCM)**
- Delta Modulation (DM)

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, ie remove high frequency components that affect the signal shape.

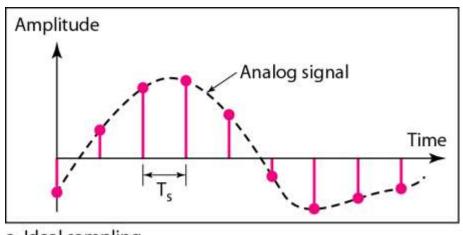
Figure 4.21 Components of PCM encoder

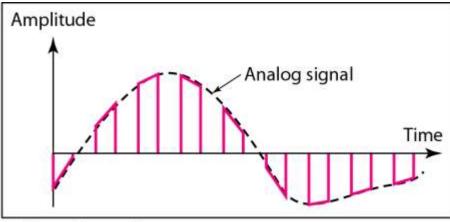


Sampling

- Analog signal is sampled every T_S secs.
- \bullet 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
- The process is referred to as pulse amplitude modulation PAM and the outcome is a signal with analog (non integer) values

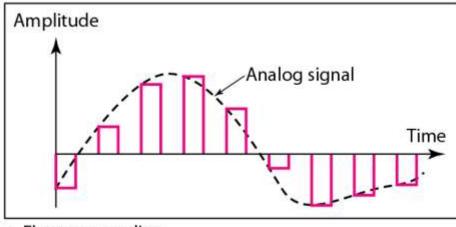
Figure 4.22 Three different sampling methods for PCM





a. Ideal sampling

b. Natural sampling



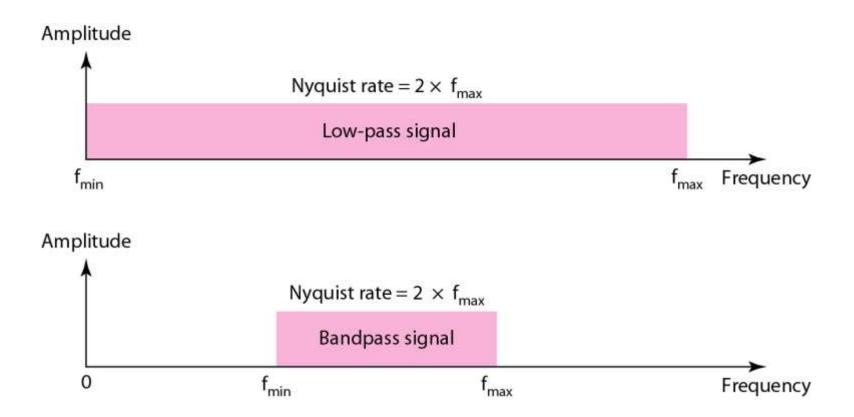
c. Flat-top sampling

-

Note

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

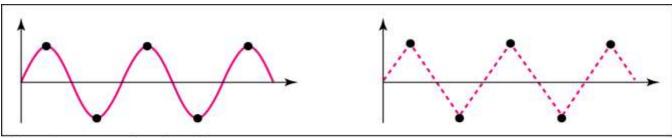
Figure 4.23 Nyquist sampling rate for low-pass and bandpass signals



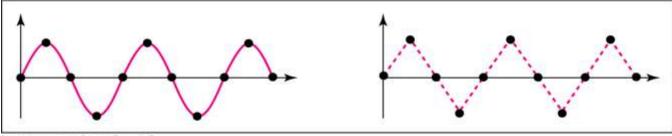
For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: $f_s = 4f$ (2 times the Nyquist rate), $f_s = 2f$ (Nyquist rate), and $f_s = f$ (one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.

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.

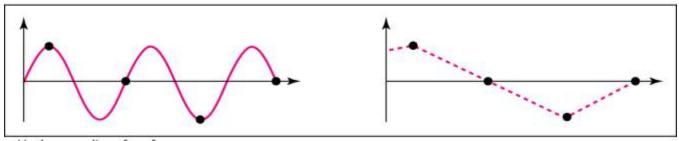
Figure 4.24 Recovery of a sampled sine wave for different sampling rates



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



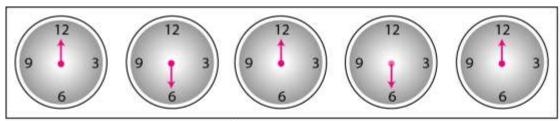
b. Oversampling: $f_s = 4 f$



c. Undersampling: $f_s = f$

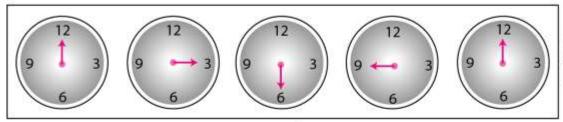
Consider the revolution of a hand of a clock. The second hand of a clock has a period of 60 s. According to the Nyquist theorem, we need to sample the hand every 30 s $(T_s = T \text{ or } f_s = 2f)$. In Figure 4.25a, the sample points, in order, are 12, 6, 12, 6, 12, and 6. The receiver of the samples cannot tell if the clock is moving forward or backward. In part 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 part c, we sample below the Nyquist rate $(T_s = T \text{ or } f_s = f)$. 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.

Figure 4.25 Sampling of a clock with only one hand



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): $T_s = T\frac{3}{4}$

An example related to Example 4.7 is the seemingly backward rotation of the wheels of a forward-moving car in a movie. This can be explained by under-sampling. A movie is filmed at 24 frames per second. If a wheel is rotating more than 12 times per second, the under-sampling creates the impression of a backward rotation.

Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz. The sampling rate therefore is 8000 samples per second.

A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

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.

A complex bandpass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

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.

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 △.

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

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 amplitutes V_{min} =-20V and V_{max} =+20V.
- We want to use L=8 quantization levels.
- Zone width $\Delta = (20 -20)/8 = 5$
- 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

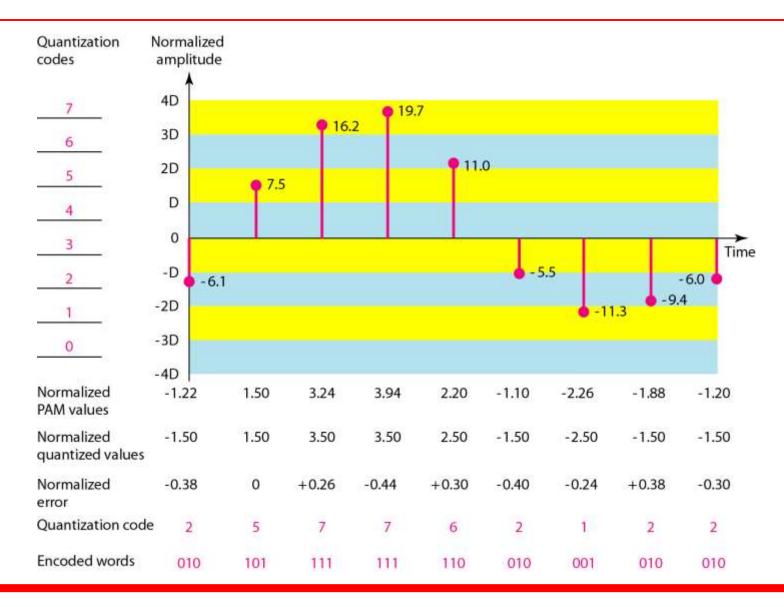
Assigning Codes to Zones

- 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.

Figure 4.26 Quantization and encoding of a sampled signal



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 ∆ which results in smaller errors.
- BUT, the more zones the more bits required to encode the samples -> higher bit rate

Quantization Error and SN_QR

- Signals with lower amplitude values will suffer more from quantization error as the error range: $\Delta/2$, is fixed for all signal levels.
- Non linear quantization is used to alleviate this problem. Goal is to keep SN_QR fixed for all sample values.
- Two approaches:
 - The quantization levels follow a logarithmic curve. Smaller Δ 's at lower amplitudes and larger Δ 's at higher amplitudes.
 - Companding: 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

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.

We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

Solution

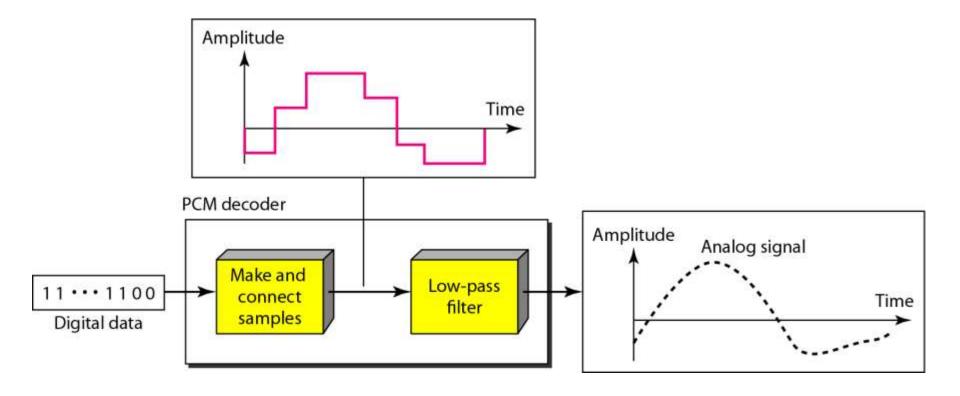
The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

Sampling rate = $4000 \times 2 = 8000$ samples/s Bit rate = $8000 \times 8 = 64,000$ bps = 64 kbps

PCM Decoder

- 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.

Figure 4.27 Components of a PCM decoder



We have a low-pass analog signal of $4 \, kHz$. If we send the analog signal, we need a channel with a minimum bandwidth of $4 \, kHz$. If we digitize the signal and send $8 \, kHz$ bits per sample, we need a channel with a minimum bandwidth of $8 \times 4 \, kHz = 32 \, kHz$.

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.

Figure 4.28 The process of delta modulation

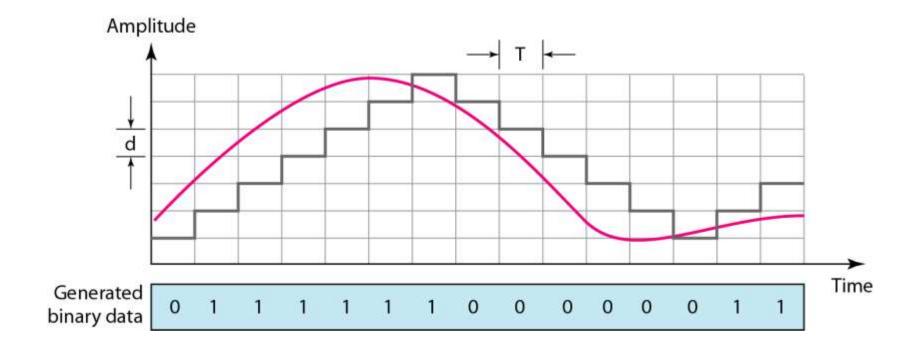


Figure 4.29 Delta modulation components

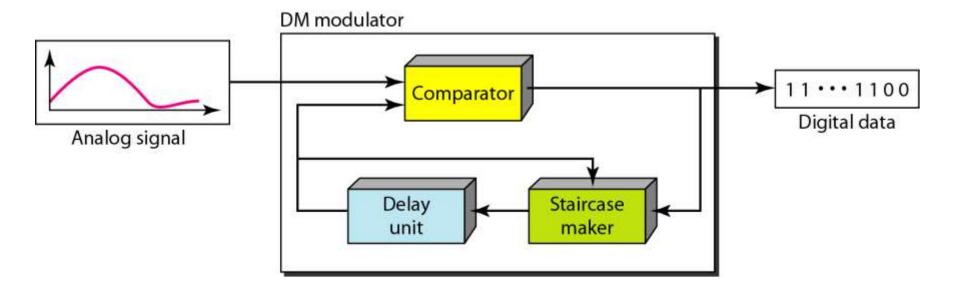
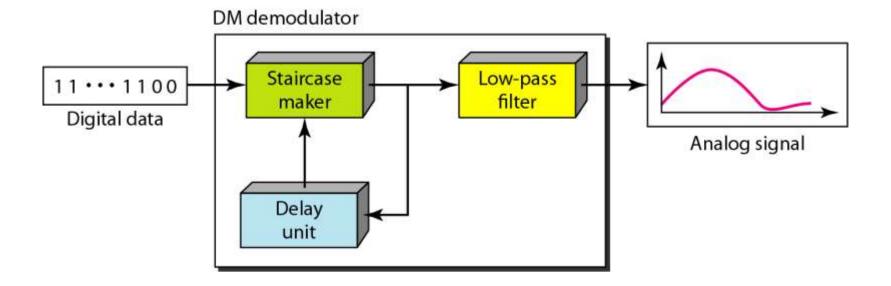


Figure 4.30 Delta demodulation components



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.

4-3 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.

Topics discussed in this section:

- Parallel Transmission
- Serial Transmission

Figure 4.31 Data transmission and modes

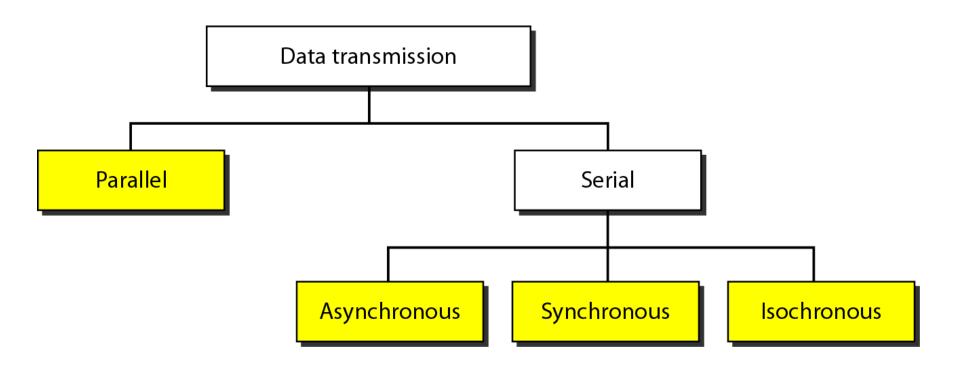


Figure 4.32 Parallel transmission

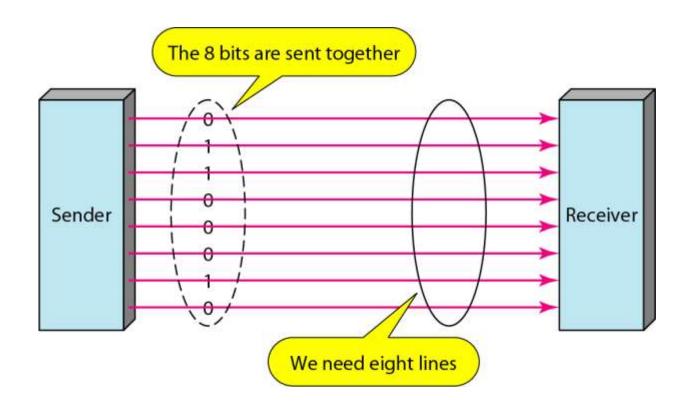
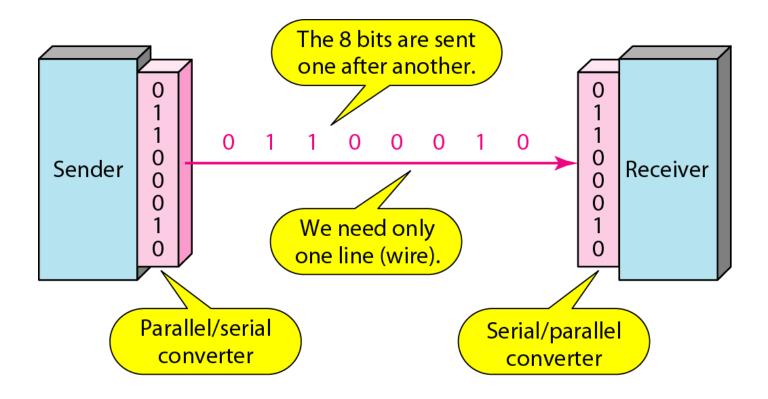


Figure 4.33 Serial transmission



4

Note

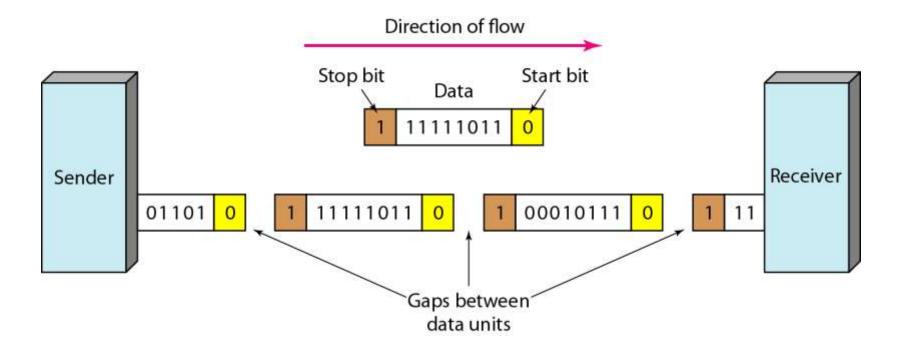
In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. There may be a gap between each byte.

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Note

Asynchronous here means "asynchronous at the byte level," but the bits are still synchronized; their durations are the same.

Figure 4.34 Asynchronous transmission

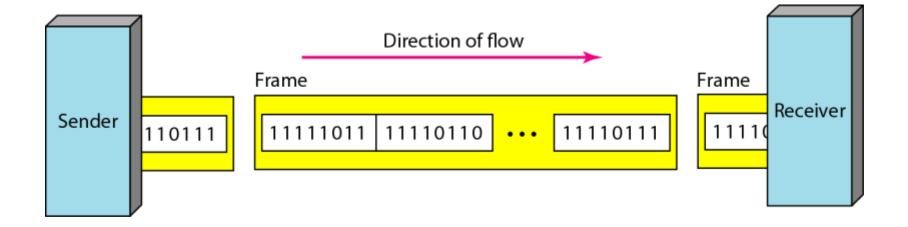




Note

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.

Figure 4.35 Synchronous transmission



Isochronous

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