Signal Encoding Techniques

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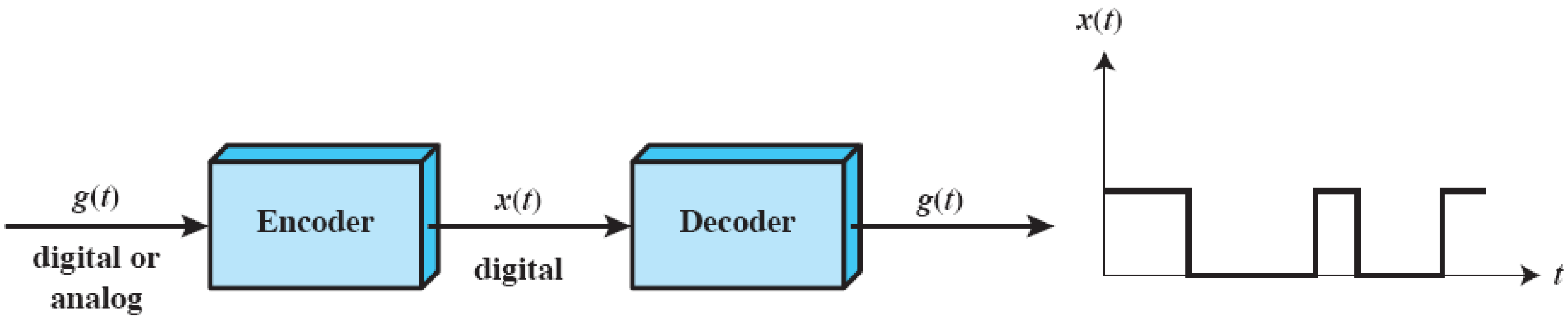
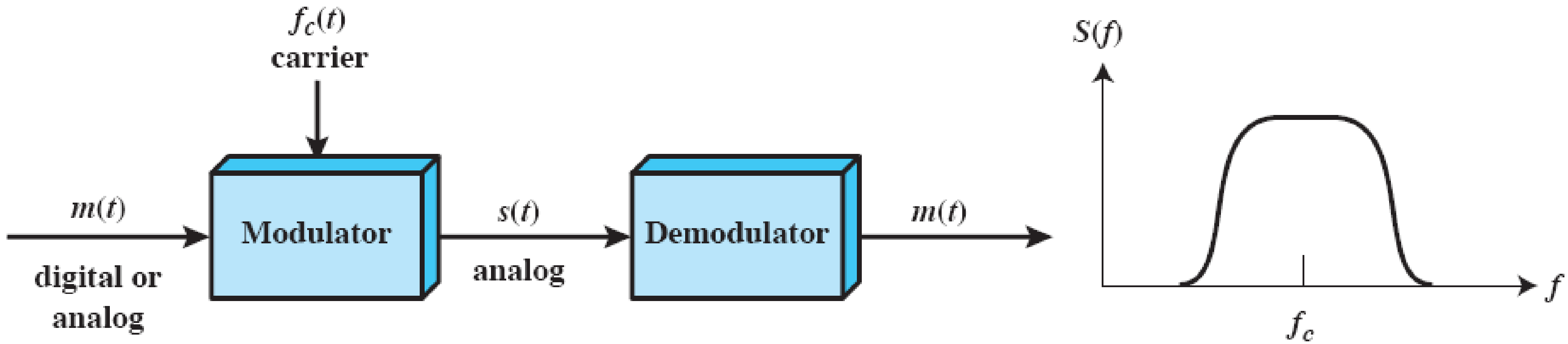


(En)coder/Decoder = codec



Modulator/demodulator = modem

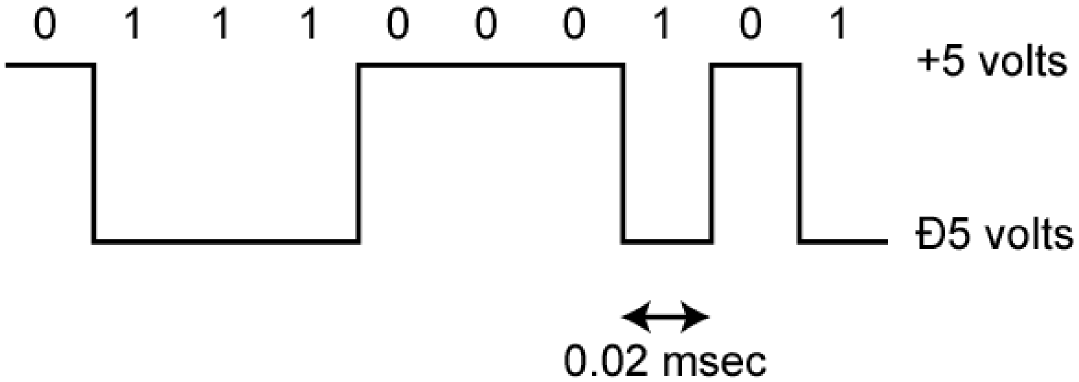
Modulation: encode source data onto a carrier signal with frequency f_c

Four Different Options

	Analog Signal	Digital Signal
Analog Data	Two alternatives: (1) signal occupies the same spectrum as the analog data; (2) analog data are encoded to occupy a different portion of spectrum.	Analog data are encoded using a codec to produce a digital bit stream.
Digital Data	Digital data are encoded using a modem to produce analog signal.	Two alternatives: (1) signal consists of two voltage levels to represent the two binary values; (2) digital data are encoded to produce a digital signal with desired properties.

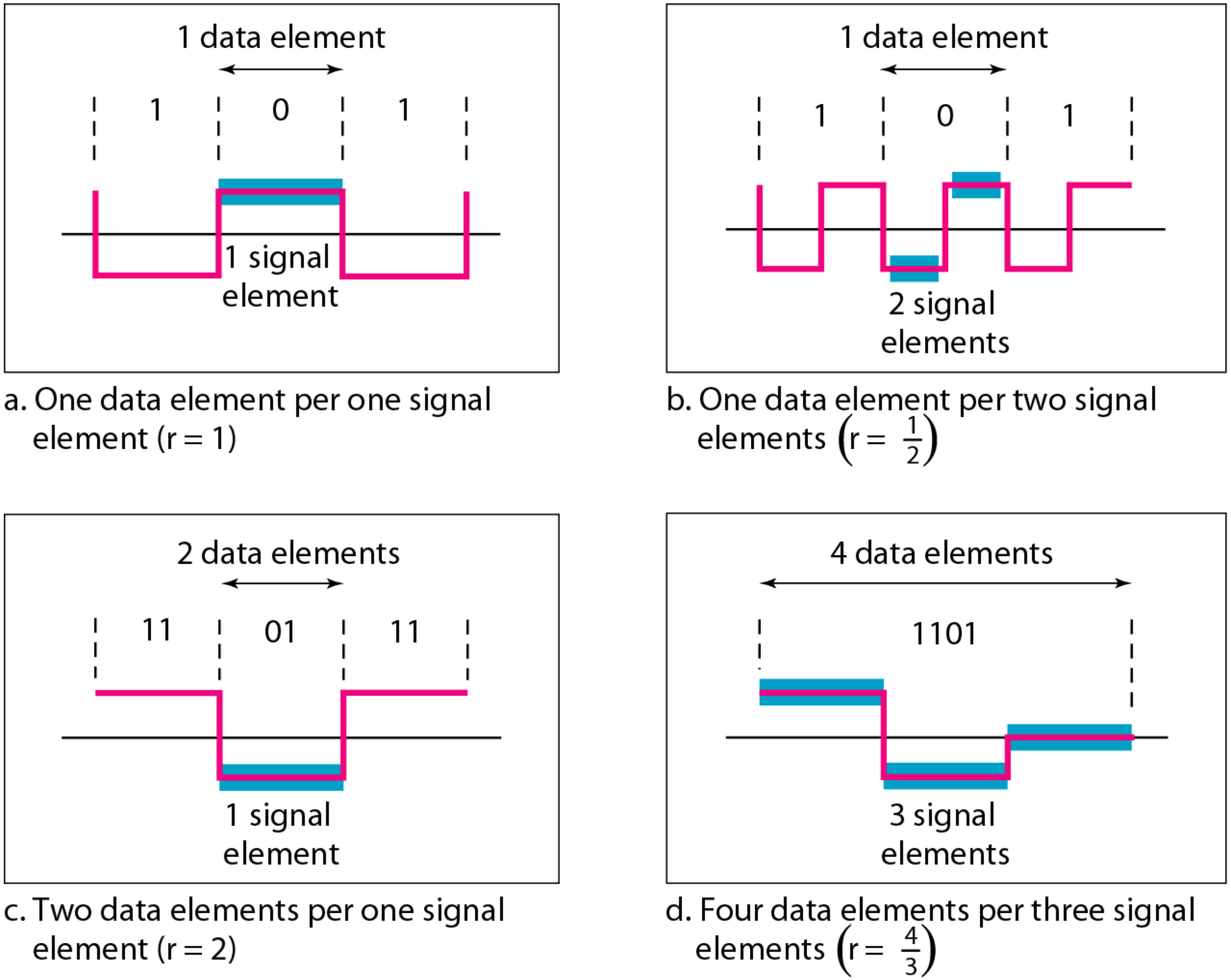
Digital Data, Digital Signal

- Digital signal
  - Discrete, discontinuous voltage pulses
  - Each pulse is a *signal element*; the rate at which pulses are sent is called the *signalling rate*
  - Binary data encoded into signal elements
    - Most obvious example: high voltage = bit 1; low voltage = bit 0
    - However, these is not used in practice!
    - Some techniques encode multiple bits into one signal element (pulse)



Data versus Signals

Data rate
= Bit rate



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

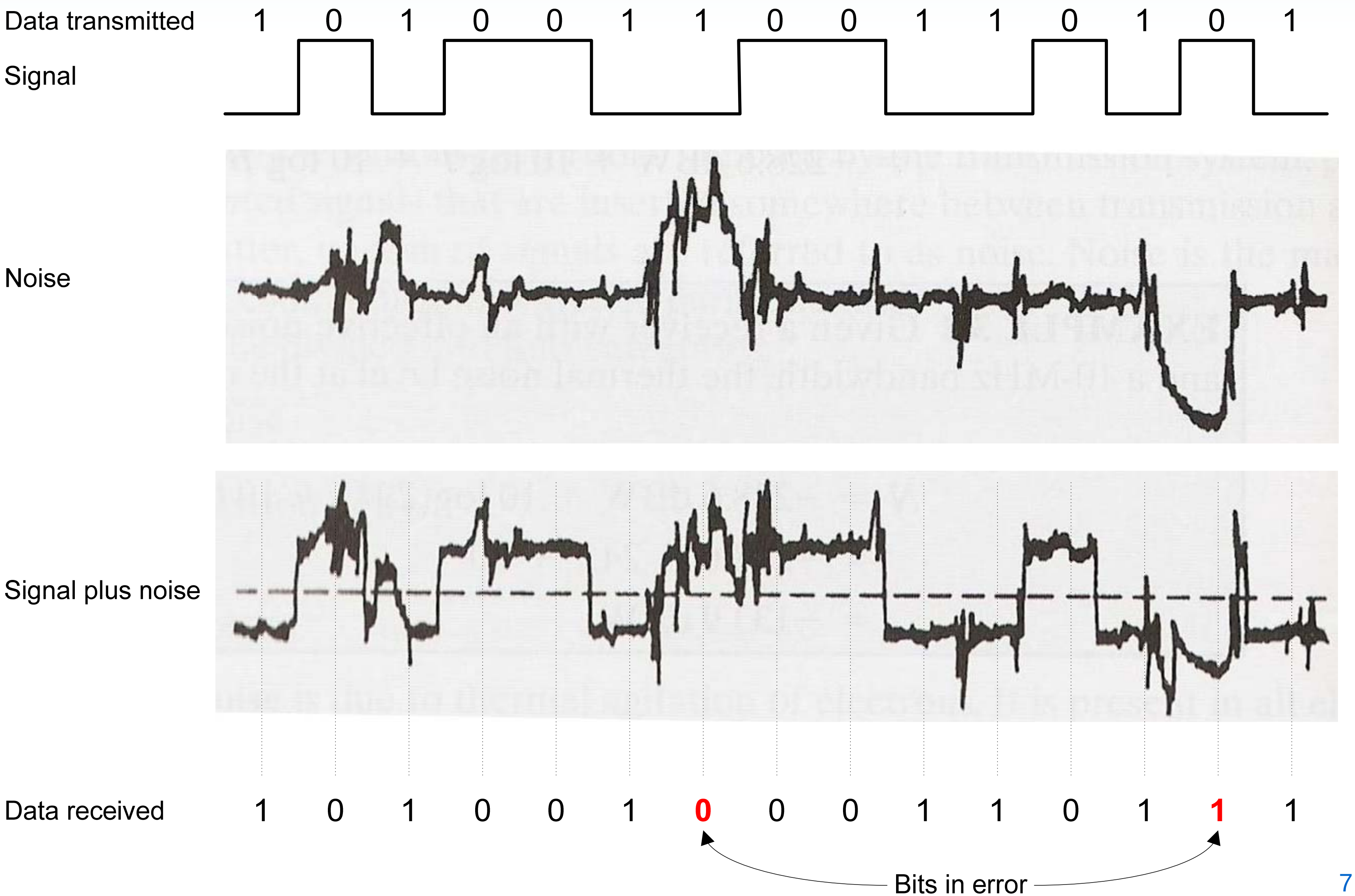
b. One data element per two signal elements $\left(r = \frac{1}{2}\right)$

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

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

Signal rate
= Pulse rate = Modulation rate = Baud rate

Effect of Noise on Digital Signal



Interpreting Signals

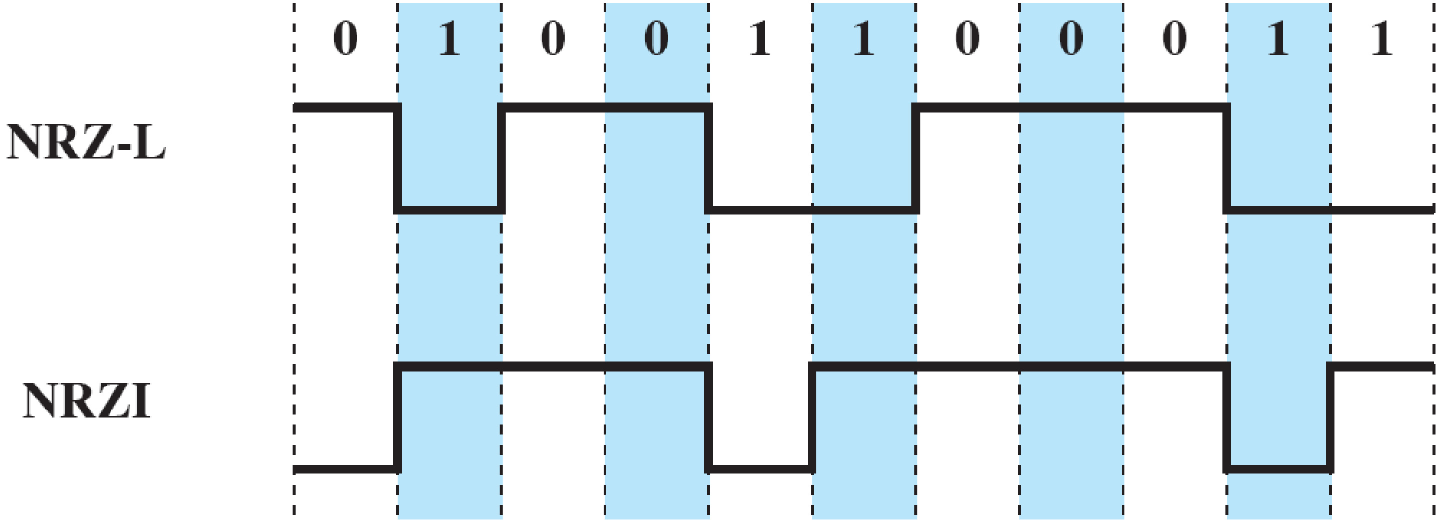
- A receiver needs to know
  - Timing of bits – when does the bit start and end?
    - Try to sample the bit in the middle of the signal element
  - Signal levels – when is it a 0 or 1?
    - Use a threshold: if the sample value is above the threshold, then value A; if below the threshold then value B
- Factors affecting the received signal, and ability to interpret correct bits:
  - Signal to noise ratio (SNR)
  - Data rate
  - Bandwidth
  - Encoding scheme

Encoding Schemes

- An encoding schemes:
  - Define the sequence of high and low signals to transmit 0's and 1's
  - The benefits of different encoding schemes can be:
    - Can shape the output spectrum to be more efficient
      - Very low frequencies (DC components) cannot pass through some systems; often need to remove these components from signal
    - Detect errors: if a sequence of signals are received that are unexpected, then an error may be detected
    - Avoid errors due to noise
    - Simplify digital circuits (hence lower cost)
      - Higher signalling rate for a given data rate leads to higher cost
  - Examples: Non-return to Zero (NRZ); Alternate Mark Inversion (AMI); Manchester; B8ZS; HDB3; …

Non-Return to Zero (NRZ)

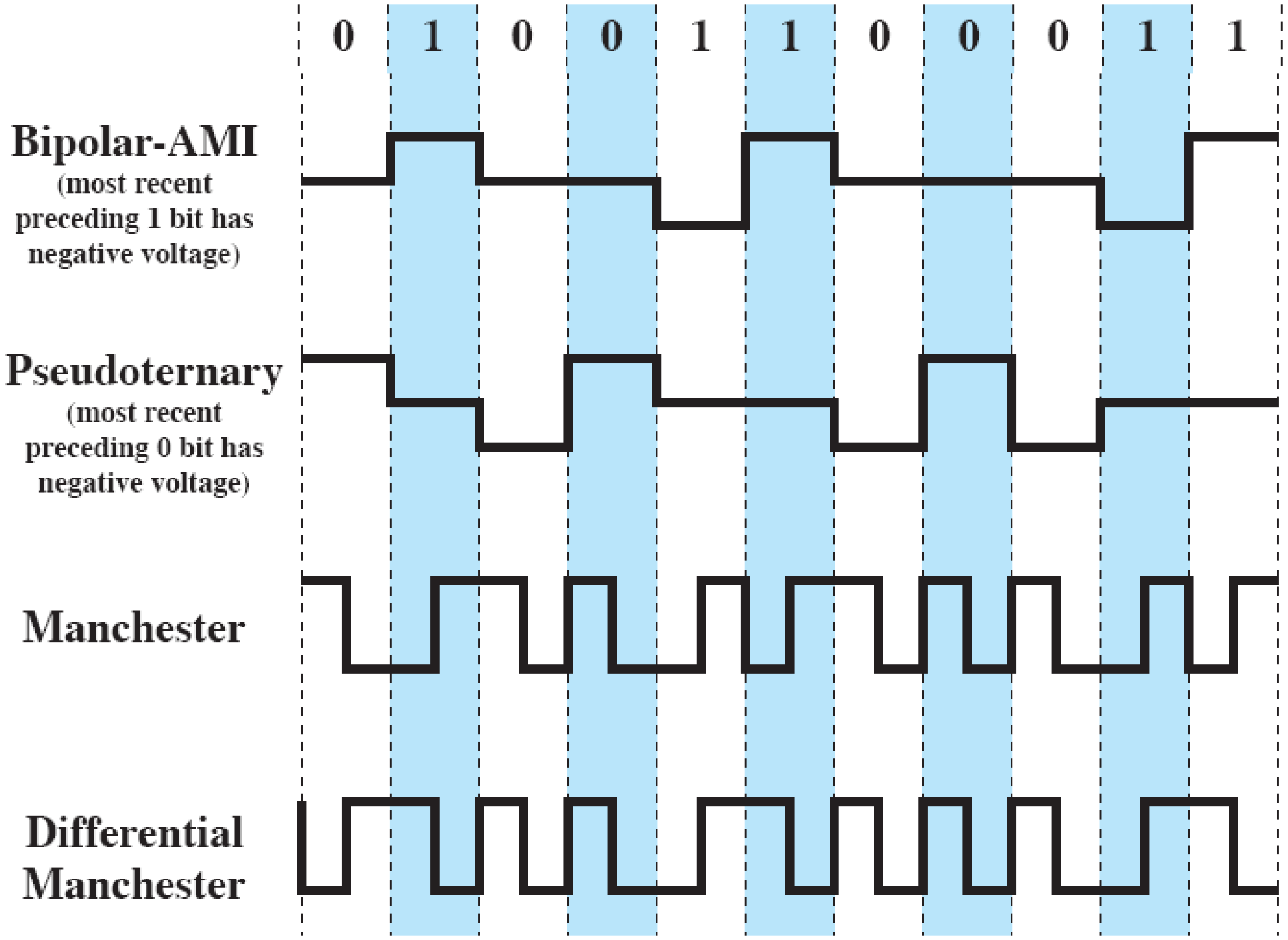
- Non-Return to Zero Level (NRZ-L)
  - Low voltage represents binary 1
  - High voltage represents binary 0
    - (or other way around)
- A variant: NRZ Invert on ones (NRZI)
  - A transition (from high to low or low to high) represents binary 1
  - No transition represents binary 0



Digital Signal Encoding Formats

- Nonreturn to Zero-Level (NRZ-L)
  - 0 = high level
  - 1 = low level
- Nonreturn to Zero Inverted (NRZI)
  - 0 = no transition at beginning of interval (one bit time)
  - 1 = transition at beginning of interval
- Bipolar-AMI
  - 0 = no line signal
  - 1 = positive or negative level, alternating for successive ones
- Pseudoternary
  - 0 = positive or negative level, alternating for successive zeros
  - 1 = no line signal
- Manchester
  - 0 = transition from high to low in middle of interval
  - 1 = transition from low to high in middle of interval
- Differential Manchester
  - Always a transition in middle of interval
  - 0 = transition at beginning of interval
  - 1 = no transition at beginning of interval
- B8ZS
  - Same as bipolar AMI, except that any string of 8 zeros is replaced by a string with two code violations
- HDB3
  - Same as bipolar AMI, except that any string of 4 zeros is replaced by a string with one code violation

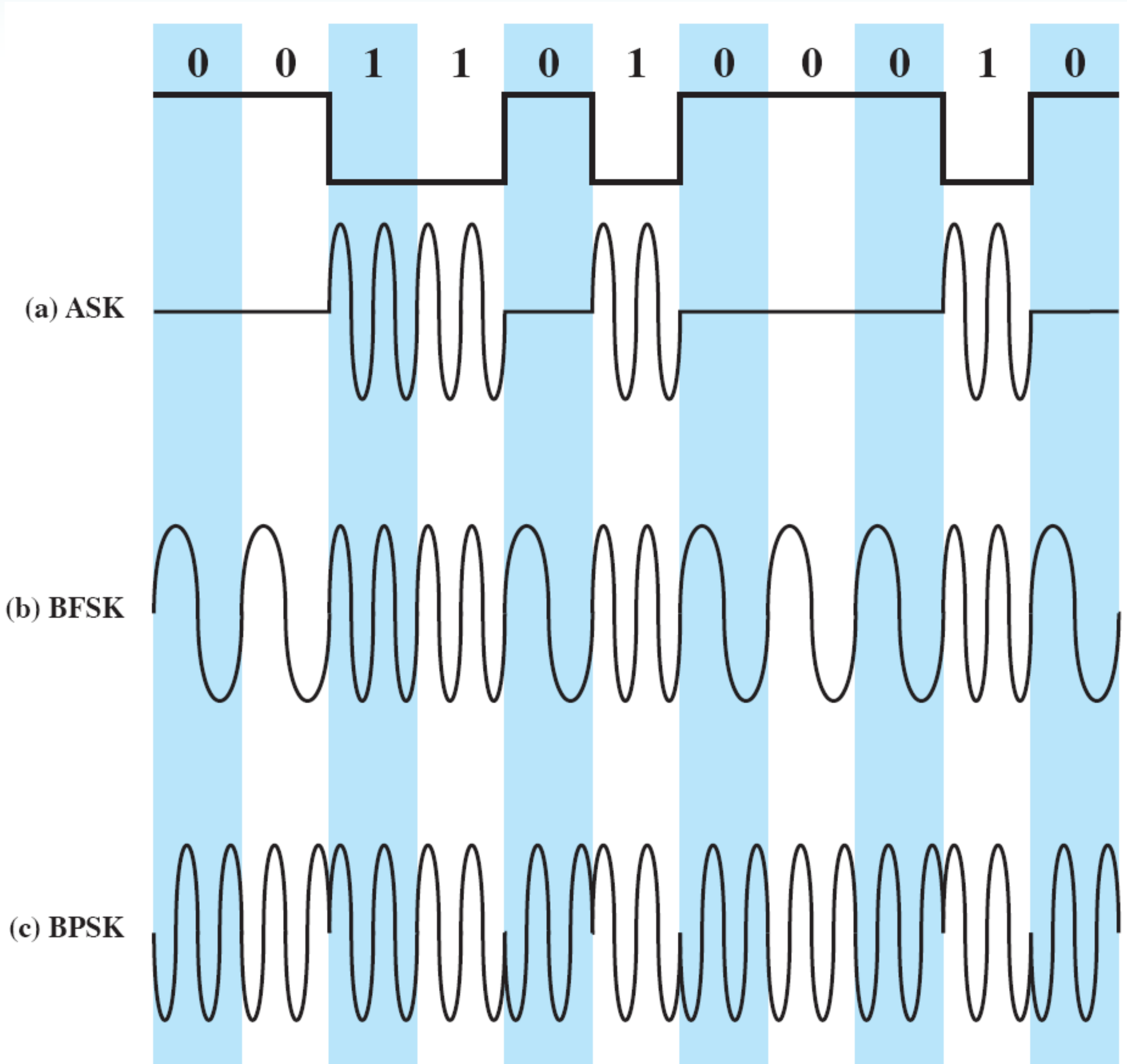
Digital Signal Encoding Formats



Digital Data, Analog Signal

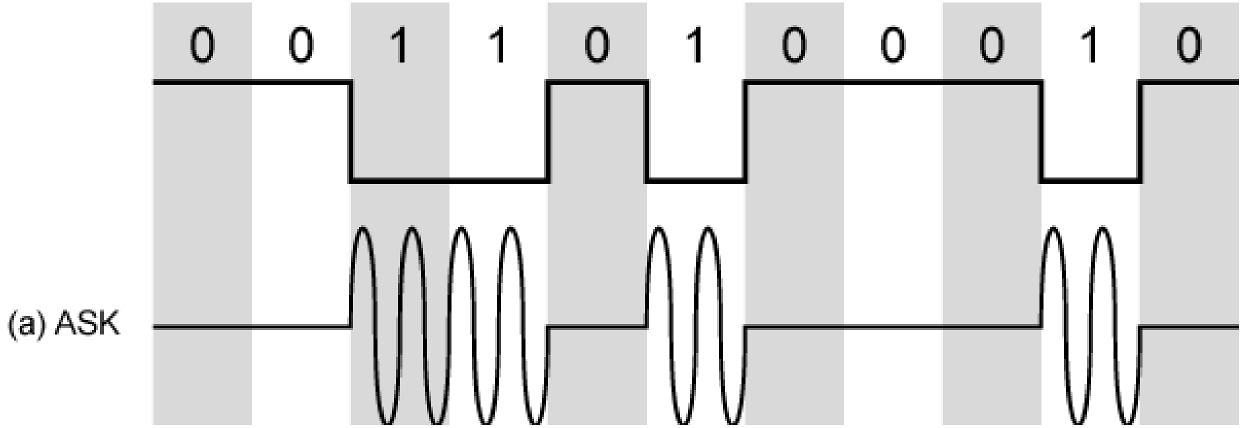
- The public (fixed, wired) telephone system is best example of analog transmission system
  - Analog signals are sent in the voice frequency range of 300Hz to 3400Hz
  - (Voice over telephone system is example of analog data, analog signal)
  - To make use of this analog transmission system to send digital data, need to modulate data onto a analog signal (carrier)
  - Device that modulates and demodulates: modem
- Encoding techniques
  - Modulation is performed by varying analog signal amplitude, frequency or phase
    - Amplitude Shift Keying (ASK)
    - Frequency Shift Keying (FSK)
    - Phase Shift Keying (PSK)
    - And combinations …

Modulation Techniques



Amplitude Shift Keying

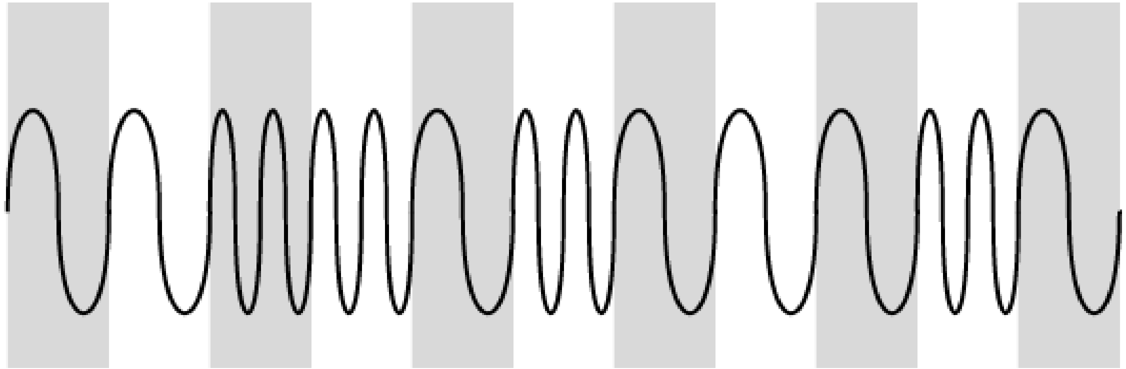
- Encode 0/1 by different amplitudes of the carrier signal
  - Usually have one amplitude zero, and the other non-zero
- Simple, but has problems for telephone lines:
  - Susceptible to sudden gain changes and distortion
  - Inefficient use of bandwidth
- Used for
  - Up to 1200bps on voice copper lines
  - Very high speeds over optical fiber



Frequency Shift Keying

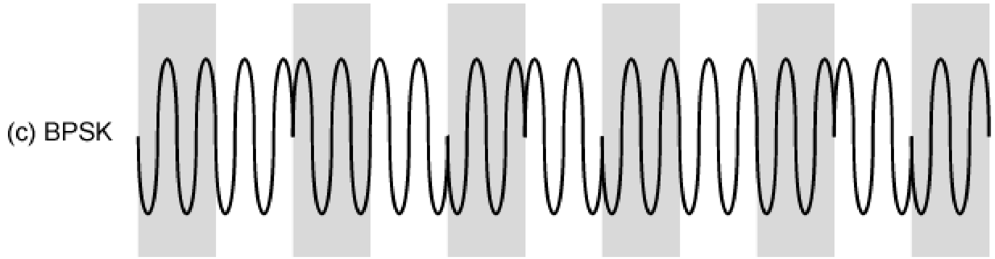
- Use different frequencies to represent digital data
  - Most common is Binary FSK
    - Two binary values represented by two different frequencies (near carrier)
- Less susceptible to error than ASK
- Used for
  - Up to 1200bps on voice grade lines
  - High frequency radio
  - even higher frequency on LANs using co-axial cable

(b) BFSK



Phase Shift Keying

- Phase of carrier signal is shifted to represent data
- Binary PSK
  - Two phases represent two binary digits



- Differential PSK
  - Phase shifted relative to previous transmission rather than some reference signal
- Quadrature PSK
  - Shift phase by 90 degrees; a signal represents two bits (00, 01, 10, 11) depending on phase shift (45°, 135°, 225°, 315°)
  - More efficient: carry more information (bits) per signal
- Quadrature Amplitude Modulation
  - Combines ASK and PSK
  - Used, for example, in ADSL modems

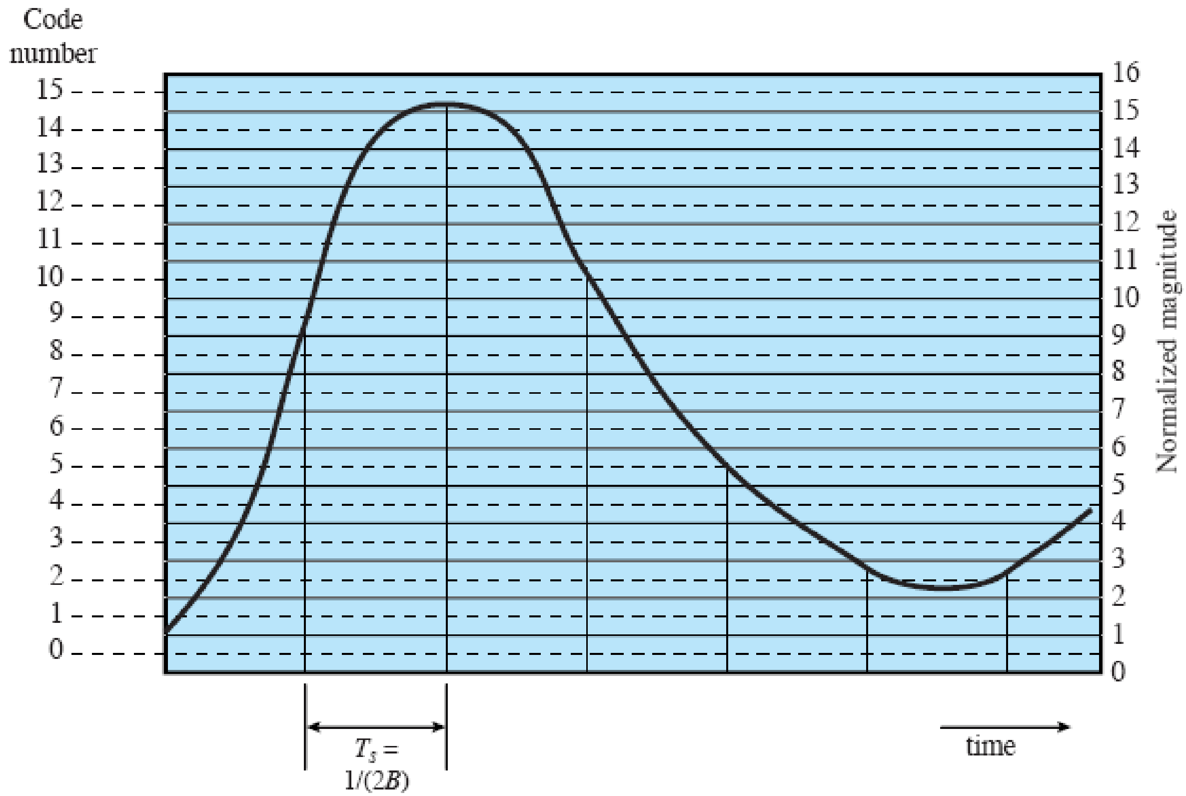
Analog Data, Digital Signal

- Involves *digitization* of analog signal
  - Convert analog data to digital data
  - Then send digital data using, for example:
    - NRZ-L or similar code
    - Or convert to analog signal and send
- Analog to digital conversion done using a codec
  - Pulse Code Modulation
  - Delta Modulation
    - Simpler than PCM, but not as good performance

Pulse Code Modulation (PCM)

- PCM is used to convert analog data to digital data
- Sampling theorem:
  - “If a signal is sampled at regular intervals at a rate higher than twice the highest signal frequency, the samples contain all information in original signal”
  - E.g. 4000Hz voice data, requires 8000 samples per second
- Sampling an analog signal (data) creates Pulse Amplitude Modulation (PAM) samples
  - E.g. Assume the analog data has an normalized amplitude between 0 and 16. Then one sample gives a PAM value of 1.1, the next sample a PAM value of 9.2, and so on…
- Each PAM sample is assigned a code (Pulse Code Modulation)
  - E.g. Assume using 4-bit samples, then code 0000 (decimal 0) would be assigned to the samples with PAM between 0 and 1; code 0001 (decimal 1) assigned to PAM between 1 and 2; and so on…
- The PCM codes represent the digital data to send

PCM Example



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

PCM Performance

- Analog (continuous) data is converted to digital (discrete) data
- When the digital data arrives, the receiver must be able to reconstruct the original analog data
  - The Sampling Theorem defines how many samples are needed to reconstruct the information
  - How big should each sample be?
    - The larger the sample, the better quality of the received data, but the more bandwidth required
- Performance Example:
  - Voice (analog) data has a maximum frequency component of 4KHz
  - Need to sample at 8000 samples per second
  - Good voice reproduction typically requires 128 levels, that is 7 bit PAM codes
  - Therefore, voice data can be sent at 56Kbit/s
- Sending voice as analog signal requires 4KHz of bandwidth
- Sending voice as digital signal requires 28KHz of bandwidth (according to Nyquist)
- But advantages of digital signals (such as combining multiple signals into one, repeaters not accumulating noise, …) mean they are desired in telecommunications systems

Analog Data, Analog Signals

- Nothing needed – just send the analog data as a signal?
  - But often want to send the analog data at a different frequency, that is, modulate the analog data onto a carrier signal
- Why modulate analog signals?
  - Higher frequency can give more efficient transmission
    - Example: for wireless transmission need a larger antenna for lower frequencies. The low frequencies music and speech would need antennas many kilometres in diameter if not modulated!
  - Makes Frequency Division Multiplexing easier (combined multiple signals into one signal - covered in later topic)
- Types of modulation
  - Amplitude (AM)
  - Frequency (FM)
  - Phase (PM)
- Trade-offs between techniques: bandwidth used, power required, resilience to errors, …

We want to send at this carrier frequency

We want to send this data

The resulting signal if we use:

Amplitude Modulation (AM)

Phase Modulation (PM)

Frequency Modulation (FM)