

Signal Encoding Techniques

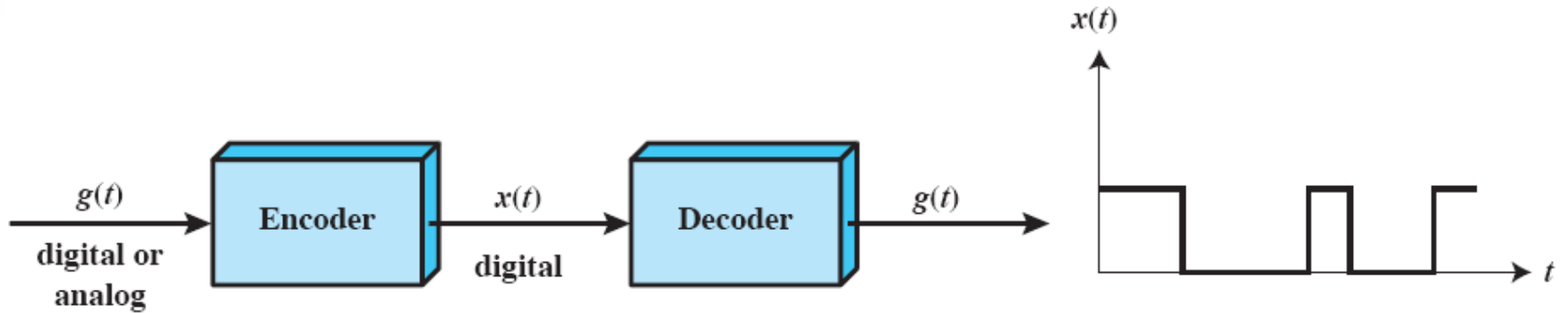
Dr Steve Gordon
ICT, SIIT

Contents

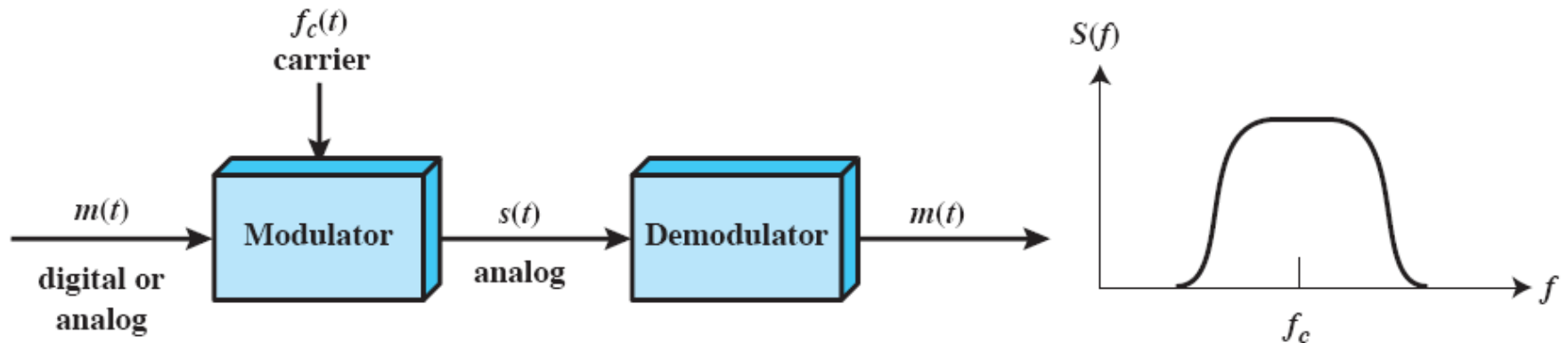
- Digital Data, Digital Signal
- Digital Data, Analog Signal
- Analog Data, Digital Signal
- Analog Data, Analog Signal



Signal Encoding Techniques



(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

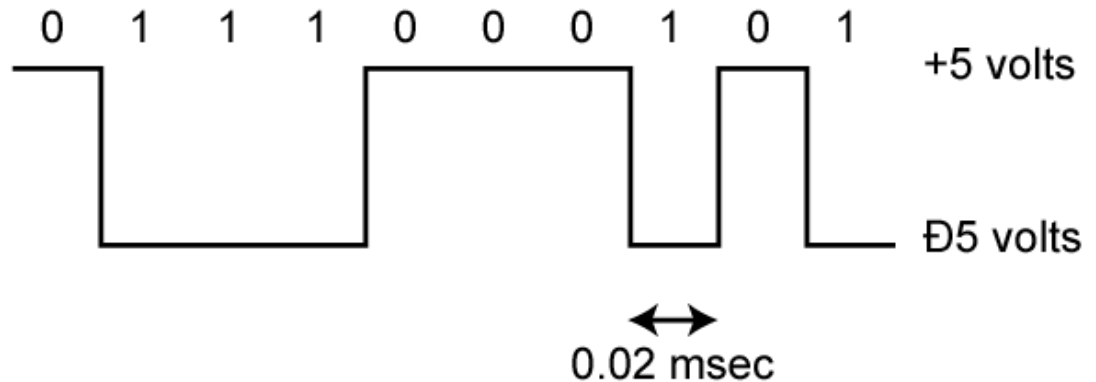
Digital data are encoded using a modem to produce analog signal.

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.



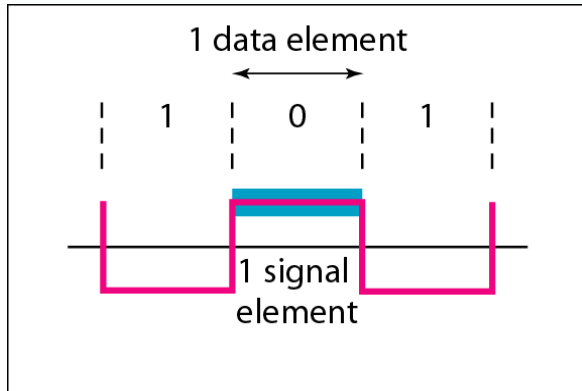
Digital Data, Digital Signal

- 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)

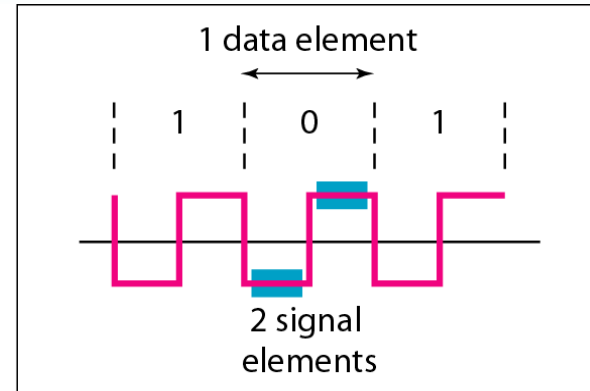


Data versus Signals

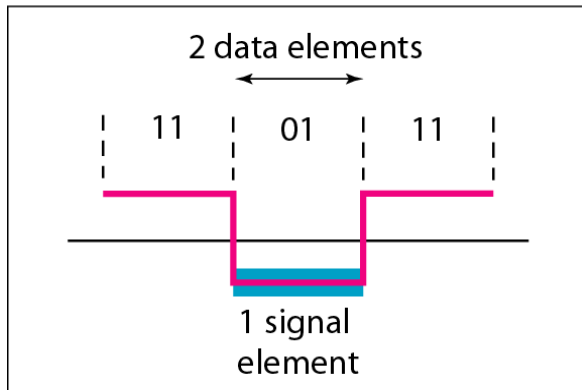
Data rate
= Bit rate



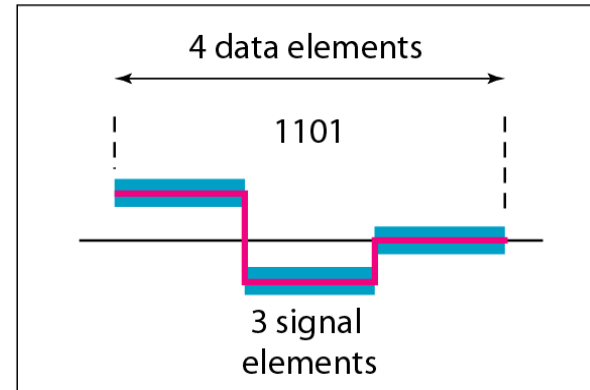
a. One data element per one signal element ($r = 1$)



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



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

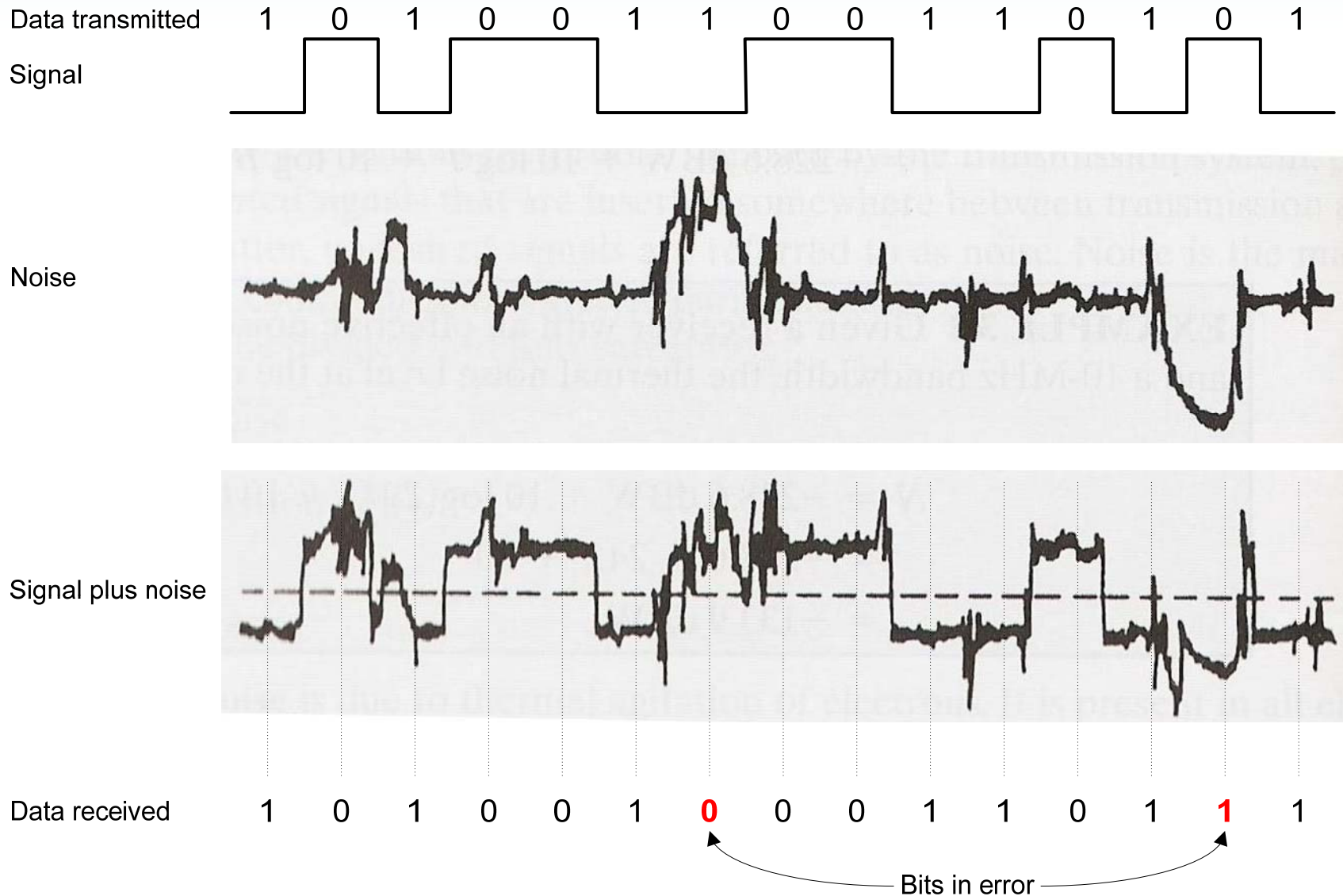


d. Four data elements per three signal elements ($r = \frac{4}{3}$)

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



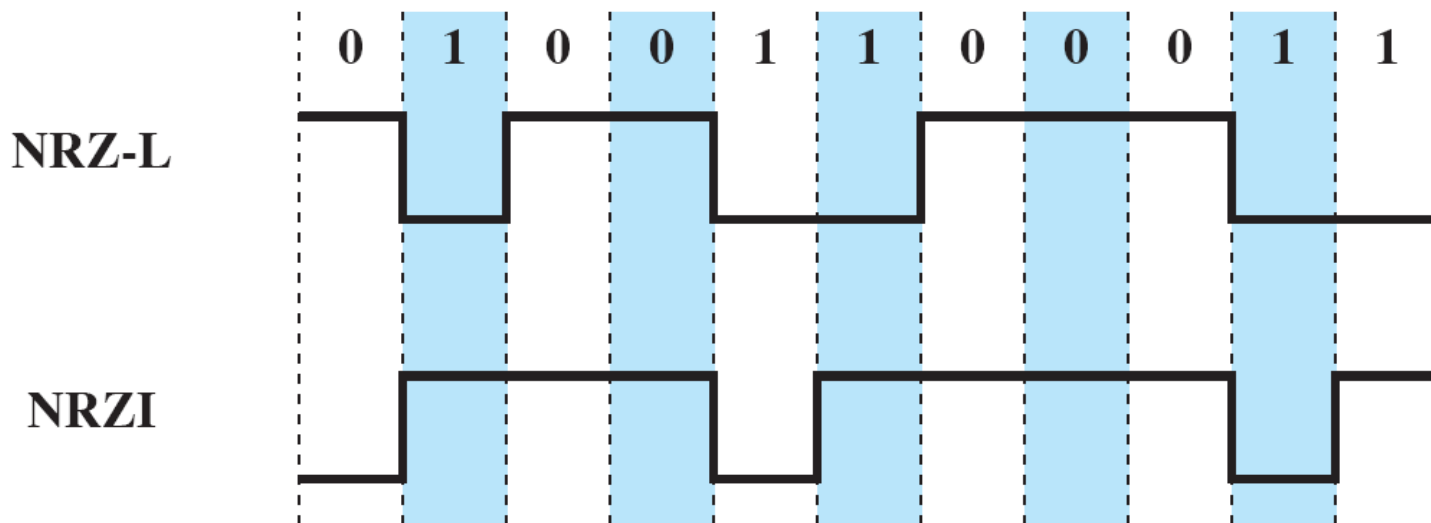
Encoding Schemes

- 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)
 - Negative voltage represents binary 1
 - Positive 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

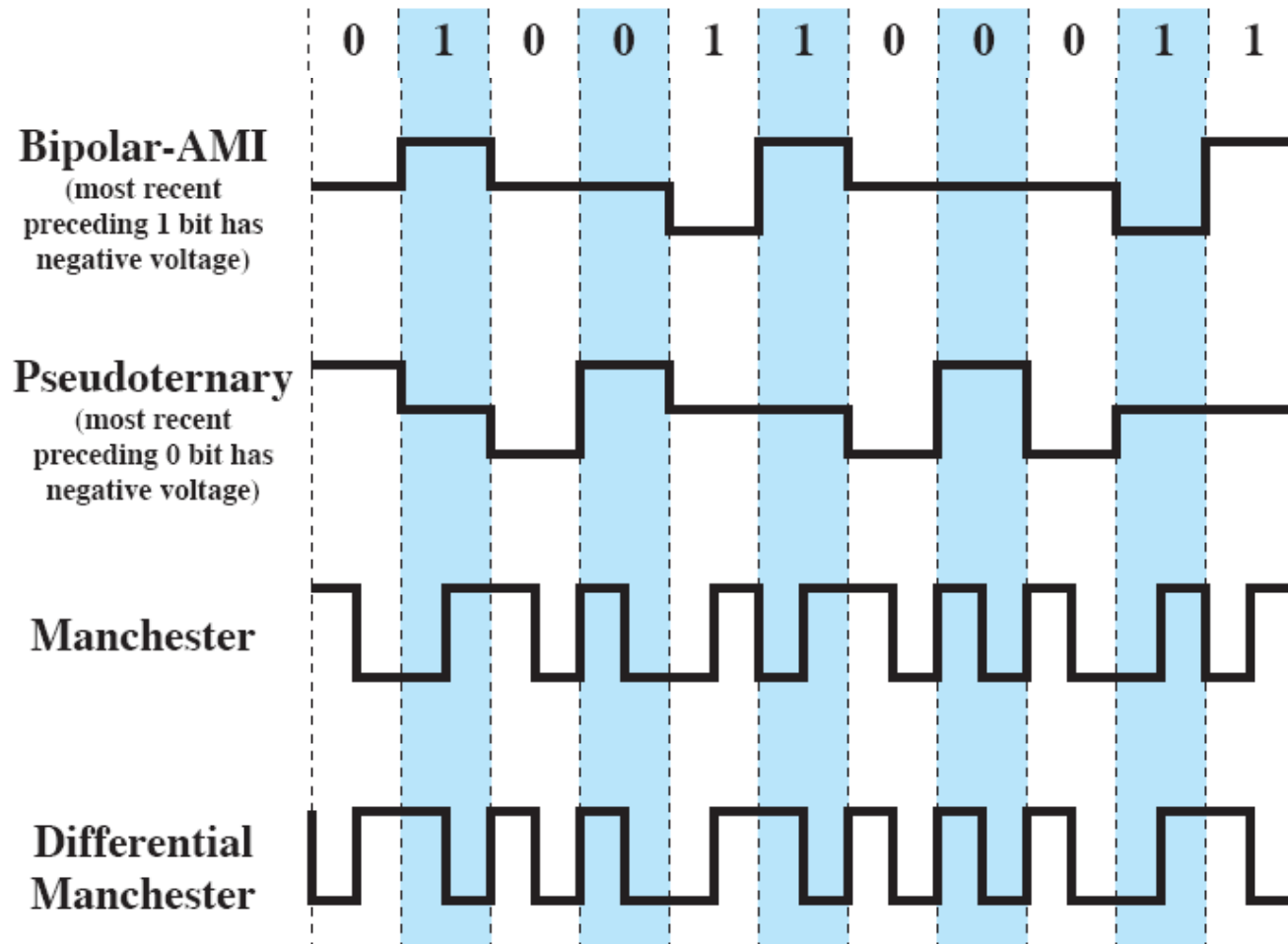


Digital Signal Encoding Formats

- 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

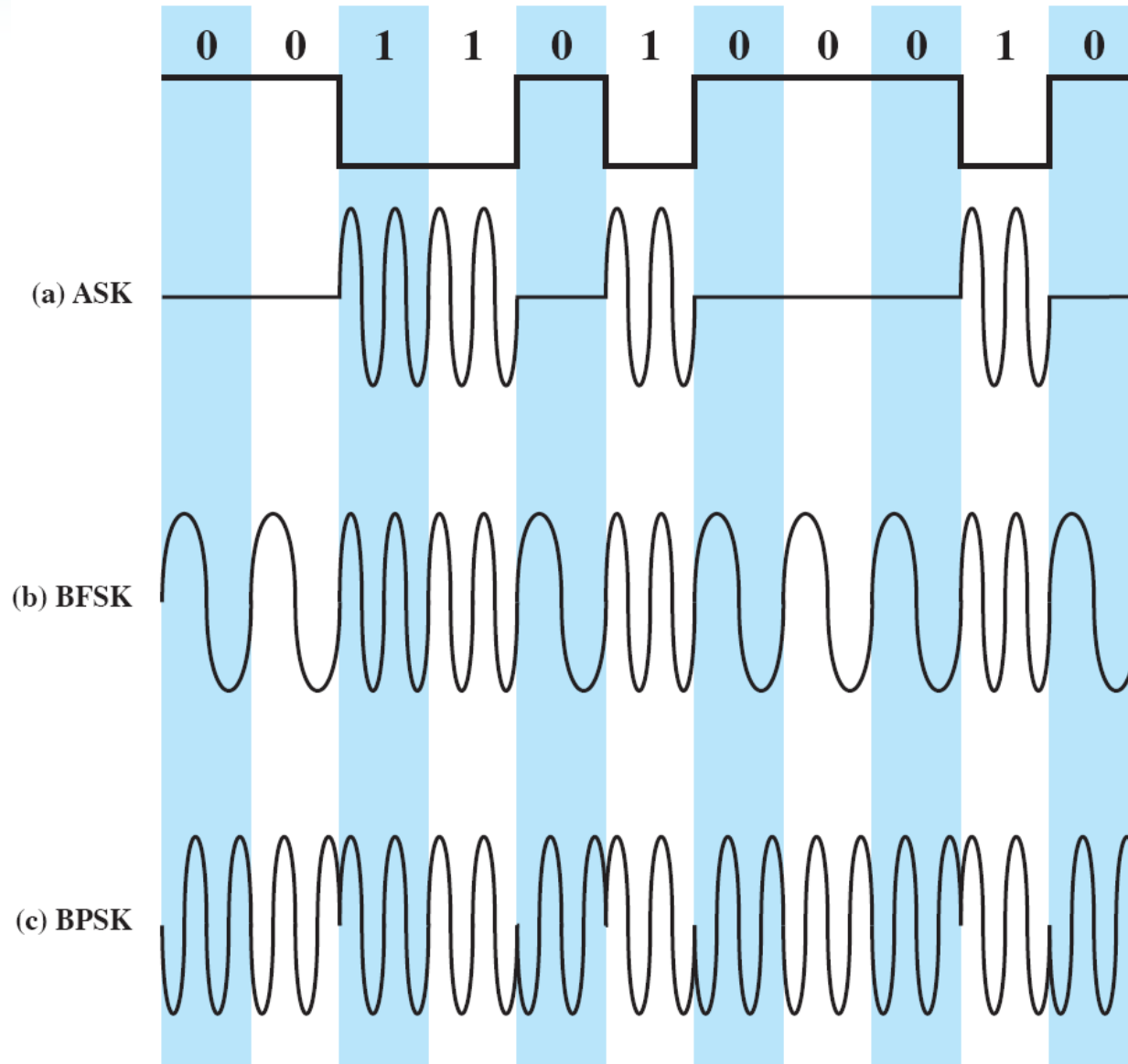


Digital Data, Analog Signal

- 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 ...

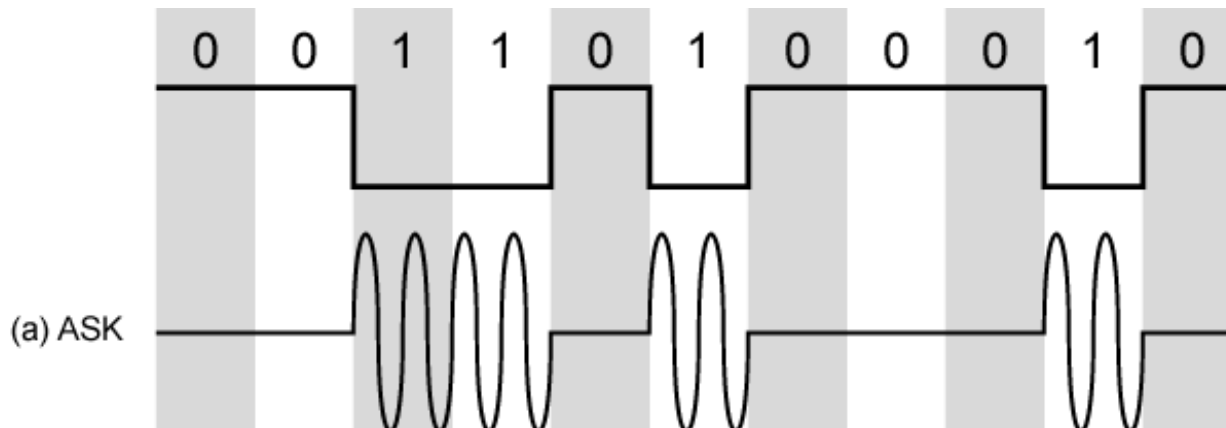


Modulation Techniques



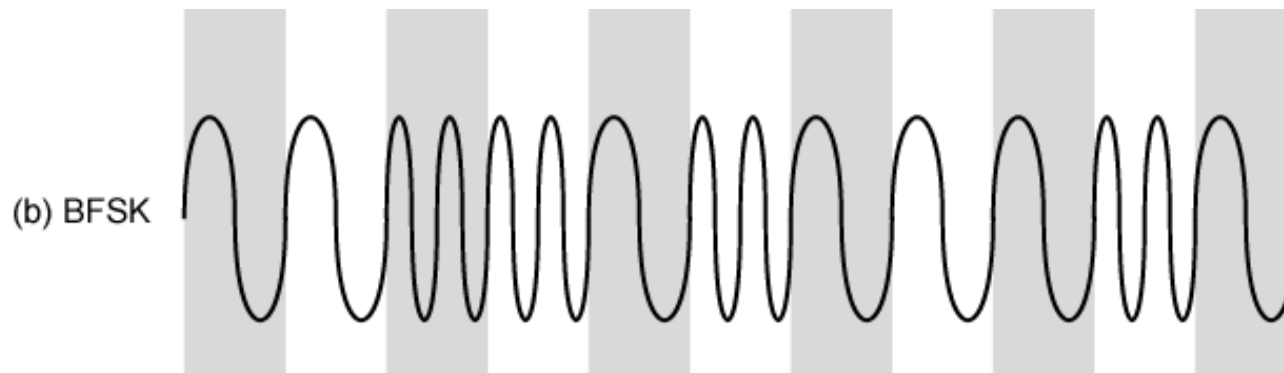
Amplitude Shift Keying

- 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



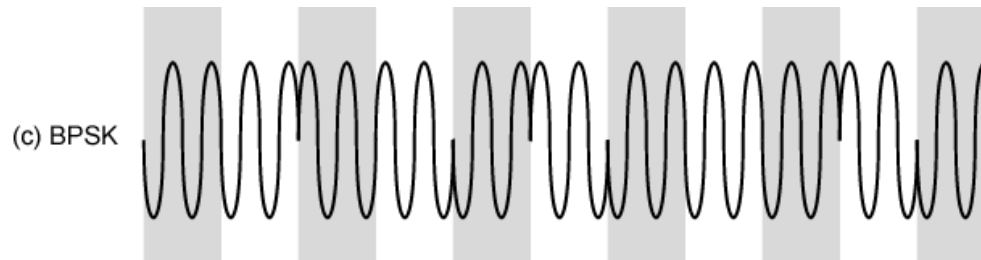
Frequency Shift Keying

- 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



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

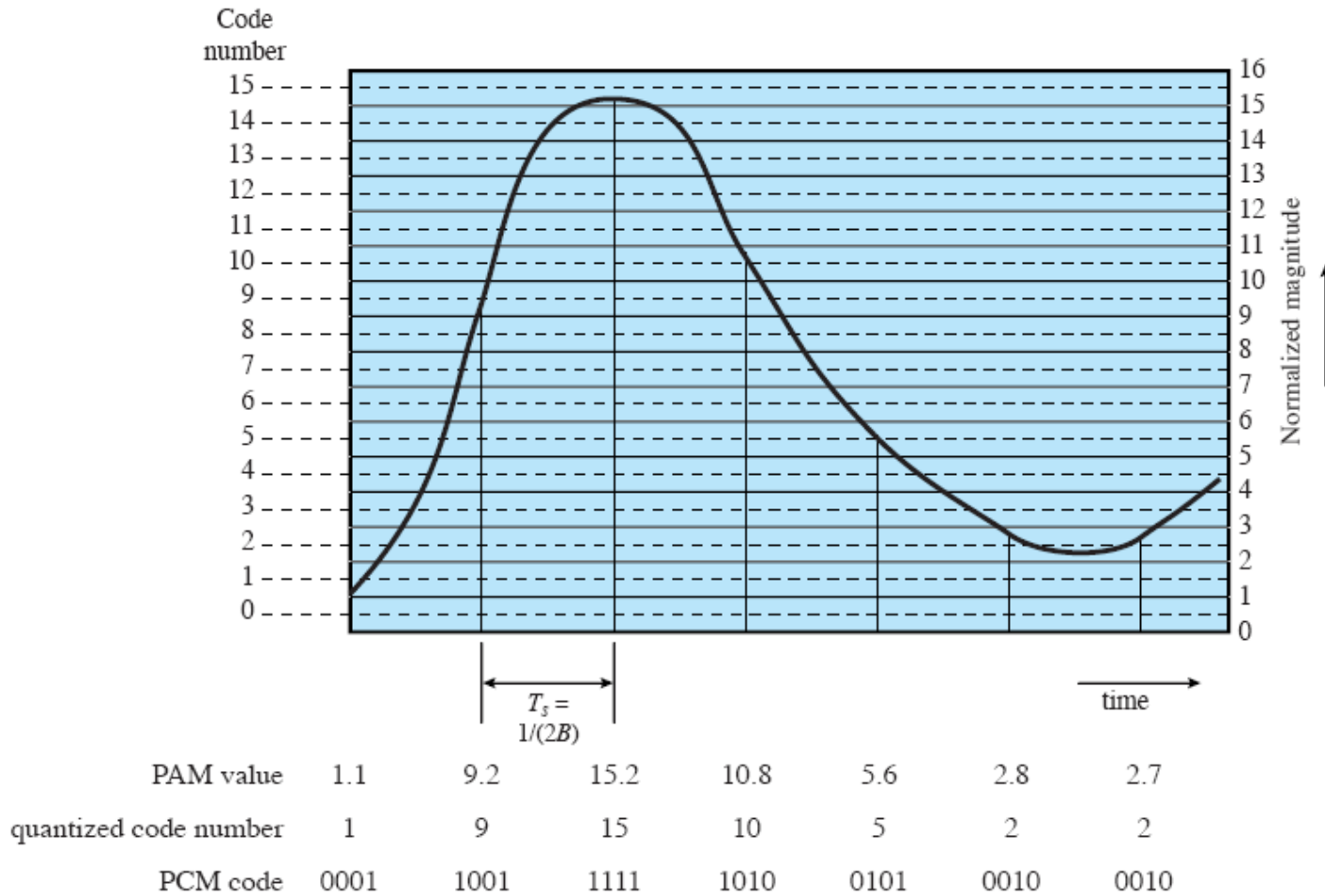


Pulse Code Modulation (PCM)

- 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 a 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



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

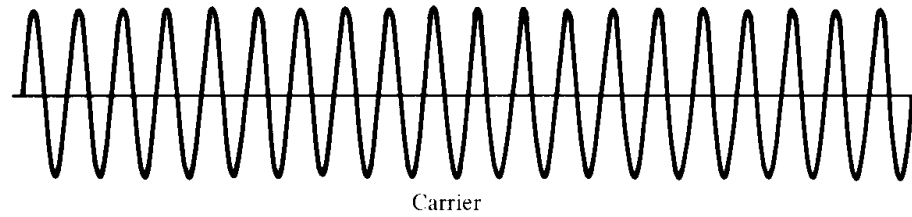


Analog Data, Analog Signals

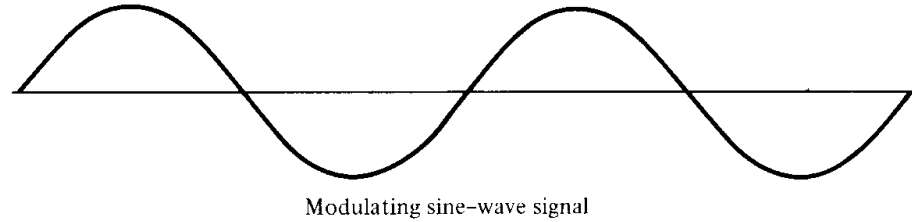
- Nothing needed – just send the analog data as a signal?
 - But often want to send the analog data at a different frequency, that is, modulate the analog data onto a carrier signal
- Why modulate analog signals?
 - Higher frequency can give more efficient transmission
 - Example: for wireless transmission 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

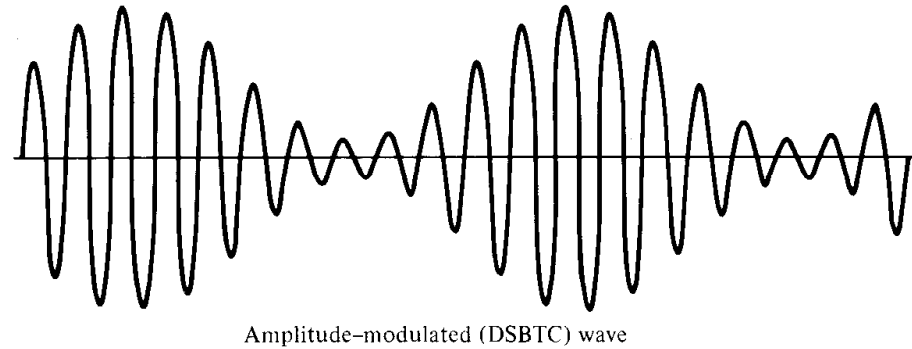


We want to
send this data

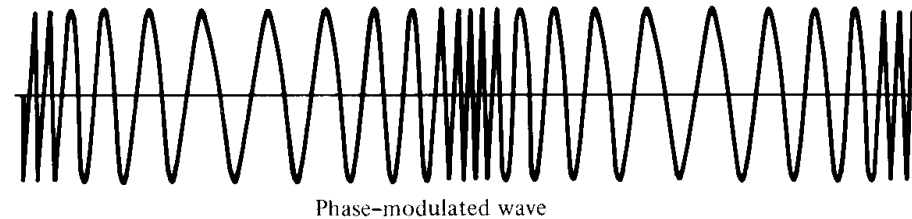


The resulting signal
if we use:

Amplitude
Modulation (AM)



Phase Modulation
(PM)



Frequency
Modulation (FM)

