Lecture 4 - Ch2. BG

• Topic: point-to-point communication (WAN's). (multiaccess later in Ch. 4 -> LAN's)

• Covers layers 1-4

• We won't discuss physical transmission. (Important, so please look at/skim sometime; we have no time in course)

• We will talk about physical channel terminology

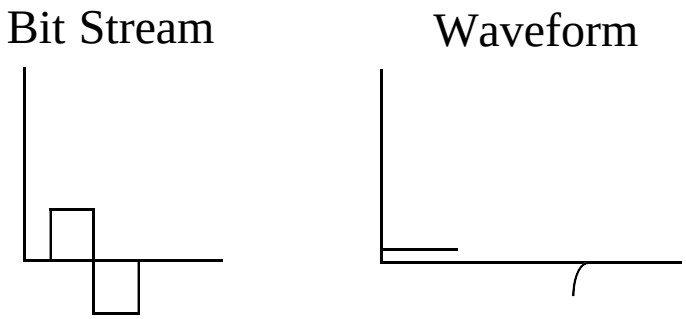
• Most important topic: ARQ (Automatic Repeat Request) -> How DLC and Transport layers recover from errors.

- 2.2 Physical Layer Terminology

1) Analog vs. Digital Channels (p. 40, BG)

-- Digital Channels = bit pipe
--- accepts a bit stream as input
--- produces a bit stream as output

-- Anaolog Channel:
--- accepts waveform as input
--- produces waveform as output
--- waveform = any function of time



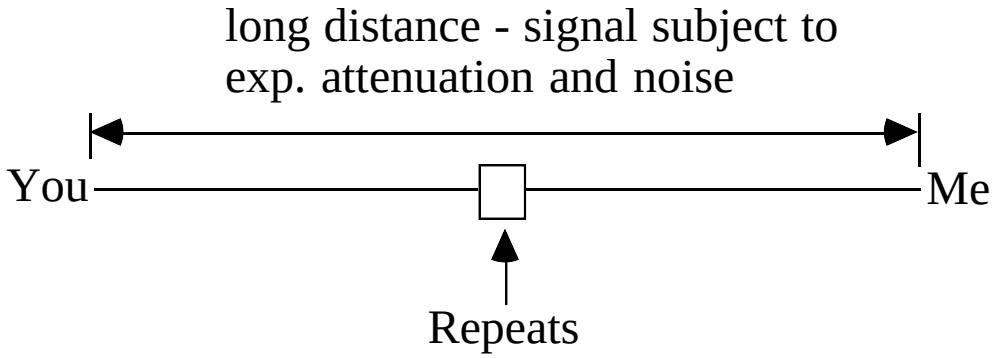
-- Modem converts bit stream to waveform, waveform to bit stream.

- Examples: telephone system

-- Your home has an analog channel - 3kHz voice to telephone local loop
-- Backbone for most (all) long distance lines is digital channel (voice converted to bit stream, then to analog again)
-- Campus phones are digital (cost of stolen phone)

- Analog vs. Digital Channel (pp. 53-54)
    - Q: Because digital channel runs on analog medium, why distinguish digital vs. analog channel?

- A:



-- Analog channel: Repeater amplifies signal + noise. I receive accumulation of noise over entire path.

-- Digital channel: Repeater recovers digital signal, regenerates analog -> noise supressed at each stage.

-- Problem: cost of digital (modems)

-- Voice now uses digital channel just for noise suppression. This is important to stereo systems too!
Music company->Master DAT->CD->
->Copy your DAT->your amplifier

• (BG) page 51 [Held & Sarch pp. 116-117 has a good explanation, too]

- A bit stream is transmitted as an AC signal at some frequency.

- What is the highest data rate one could ever achieve given a certain maximum frequency that a medium could propagate?

  Let C = data rate (bits/sec), W = max frequency (Hz), V= # voltage levels used in coding

  -- Answer 1: (1924) Nyquist theorem:
  --- Ignoring noise, using binary encoding (i.e.,2 voltage levels)

  $$C = 2W\log_2 V$$

  --- So, for a W=3000 Hz voice line, with V=2 levels, C = 6 kbps
  --- But if V=8, then C=18 kbps.
  --- Why can't we just use arbitrarily large V to get build modem with super-high C?
  Answer: As V increases, differences between voltage levels shrinks, so signals are distorted in shape and receiver cannot recover signal.

  -- Answer 2: (1948) Shannon then related encoding (e.g., value of V) to C:

  $$C = W \log_2 (1 + (S/N))$$

  where S/N is signal to noise ratio

Note: S/N is in db:
10db=10, 20db = 10^2, 30db=10^3

--- For a 3000 Hz voice line, and S/N = 30db,

$$C = (3x10^3)(\log_2(1+10^3))$$
$$= (3x10^2)(10)$$
$$= 30kbps$$

--- Today's modems are close to theoretical limit. They also use compression to achieve higher data rates (e.g., a 24kbps modem with 4:1 compression can achieve 96 kbps).

- (BG) pp. 38 and 52

  - 3 types of physical channels
    1) synchronous: transmit idle fill between frames
    2) intermittent synchronous
    3) character asynchronous: bits within a character are synchronous

  - Note that sender and receiver must be synchronized during transmission of data. Hence, 1 works at highest speed and 3 at lowest speeds

  - So:
    -- for a 2400 Hz voice line (over unshielded twisted pair) 2400bps is cut-off between 1 and 2
    -- for a coaxial cable with ethernet, 100Mbps is cut-off

Analog Channel - Synchronous vs. Asynchronous

- High speed modems require continuous transmission even if DLC has no frames to send

- Low speed modems (<2400 bps on a voice grade 3kHz telephone line) can use intermittent synchronous or character asynchronous.

- Coaxial cable has higher bandwidth -> crossover point is higher. 10 Mbps ethernet is intermittent sychronous (problem: cannot have > 100Mbps ethernet because it fundamentally is intermittent synchronous mac)

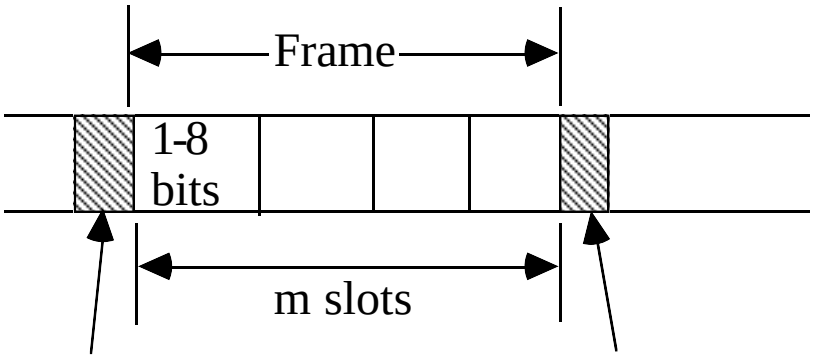
- (BG) 2.2.6 FDM vs. TDM

Goal: multiple channels over one physical medium

- Examples:
  - -- many voice calls over one transatlantic optical fiber
  - -- many TV stations over cable
  - -- ISDN: voice + data over one wire to your home

- FDM:
  - -- Shift each channel in frequency by adding a different carrier frequency
  - -- Split *W* Hz medium into *m* channels, each with approximately W/m bandwidth. (< W/m in practice - guard bands to reduce cross talk (surely you've heard this on your telephone!))

- TDM:
  - -- Bandwidth = W has greater data rate than BW <W/m
  - -- Multiplex m bit streams into one bit stream using bandwidth W
  - -- Divide bit stream into frames:



Extra bits to help receive and maintain synchronization

- In U.S.A.
  - -- T1:
    - --- 125 µsec for a 24x8+1 = 193 bit frame
    - --- m = 24
    - --- 8 bit slots -> 64kbps
    - --- 1 bit extra for synchronization -> 1/8 x 64 = 8 kbps (contains 10101010 = sine wave @ 4000Hz)
    - --- data rate = 1.544 Mbps
      = 24 x 64 kbps + 1x8 kbps
  - -- T3:
    - --- data rate = 44.736 Mbps (internet backbone rate in U.S.A)
    - --- multiplexes 28 T1 signals
- In Europe
  - -- m = 32
  - -- data rate = 2.048 Mbps

- Digital Channels (BG 2.2.8)

  - Should phone company/data carrier sell you a digital vs. analog phone? (hot debate in communications industry)

    -- One answer: simply a question of who buys modem

    -- Better answer: digital channels have higher data rates with lower error rates

    -- Current answer:
    --- local loop: analog
    --- backbone net: digital - TDM + T1

    -- Future answer from phone companies:
    --- digital local loop (plus expensive phone for customers, w/ modem)
    --- ISDN: Integrated Services Digital Network
    --- Sonet: (Synchronous Optical Network)
    -> STS-1 = 51.84 Mbps
    -> STS-n = n (51.84 Mbps)
    -> TDM, 125µs frame with large number of 64 kbps voice circuits with 1 byte per frame

- ISDN

  - Re-use existing telephone wiring for data + voice

  - "Integrated" means backbone + local loops are digital, and service can be voice, data, video

  - Today's local loop cannot carry voice + data together/simultaneously to your home

  - Common offering:
    -- Basic service:
    --- "B" - 2 channels of 64 kbps (two voice or data channels)
    --- "D" - 1 channel of 16 kbps (network management, disconnecting calls, and low data-rate services)

    -- "2B + D" service
    --- You could use computer at home without losing phone calls
    --- Maybe "1B + D" is adequate
    --- Could use "B" channel as:
    -> Virtual circuit to one destination
    -> Access line to packet switched network

    -- Primary service: for businesses with PBX's:
    --- A T1 link:
    -> 24 channels of 64 kbps (US, Japan)
    -> 31 channels of 64 kbps (Europe)
    -> one channel is D, rest are B

- ISDN limitations
  - -- You need a digital phone for for voice traffic
  - -- What about
    - --- TV service to home?
    - --- Data transfer of images (10^9 bits)?