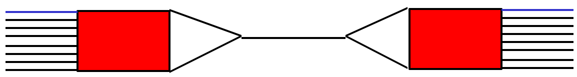
CS656: Computer Networks

Analog and Digital Transmission Interfaces & Multiplexing (Physical Layer)



Class 3
19:20 to 22:00
10 Sep 2002

Class 3 Overview

- Character transmission with parity
- Signal gain and loss: dB
- Signals and Transmission
- Switching
- ISDN
- Other Transmission Methods
- Homework & Project

Sending Character Data: Parity

- a simple error-checking code
- takes advantage of 'spare' bit in byte when using 7-bit ASCII
- **even parity**: make last bit in byte a 0 or a 1 so that the total number of 1's is *even*
- **odd parity**: make last bit in byte a 0 or a 1 so that the total number of 1's is odd

Sending Character Data: Parity

- e.g., the character '**K**' : in binary: **X100 1011**
  - has an even number of 1's (4)
  - so even parity version is: **0100 1011**
  - odd parity version is: **1100 1011**

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Gain/Loss in dB

- **decibel**: **dB**
  - unit of signal strength based on *ratio of power*:

$$\boxed{S_{dB} = 10 \log_{10}\left(\frac{\text{measured power}}{\text{reference power}}\right)}$$

- signals strength is multiplied through cascaded blocks; using dB these strengths <u>add</u>
- note: 3dB gain means twice as ‘loud’

Using db

1. amplifier making a signal 100 times stronger has gain of:

   10 log (100) = 20 dB

2. amplifier gain of 44 dB is a ~ 25,000 X boost:

   $P_m/P_r = 10^{(44/10)} = 25,119$

Using db

3. a signal reduced to 5% of original strength has:

   10 log(0.05) = -13 dB

4. an amplifier produces 1.0 watt for an input of 0.5 watts has gain:

   10 log (1.0/0.5) = 10 log(20) = 13 dB

dB and electrical strength

- If signal intensity measured in <u>volts</u>, then:

$$P = VI = V^2/Z$$

so:

$$10\log\left(\frac{V_{meas}^2/Z}{V_{ref}^2/Z}\right) = 20\log\left(\frac{V_{meas}}{V_{ref}}\right)$$

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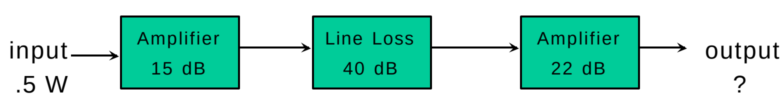
Signal Strength in dB, example



- Overall gain is:

  15 db + (-40 db) + 22 dB = -3 dB

- So change in power over channel is:

  -3 dB = $10^{\left(\frac{1}{10} P_{meas}/P_{ref}\right)} = 10^{(-3/10)} = 0.5$

- Input signal 0.5 watts X 0.5 gain factor = 0.25 watts

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Signals and Their Transmission

- what, *exactly*, do we put on 'the wire' ?
- directly:
  - what we start with (analog, digital)
    - digital pulse train
    - analog voice signal
- often use indirect **carrier:** a signal whose properties we alter in order to carry our message

Signals and Their Transmission

- Possibilities:

transmit as: / input is:	A	D
A	AM, FM, PM…	PCM, DM,…
D	ASK, FSK, PSK,…	NRZ, AMI,…

Analog to Analog

- Use **carrier** to move message
  - e.g., $S(t) = A\sin(2\pi ft + \phi)$
- signal content is in changes to the carrier

Analog to Analog

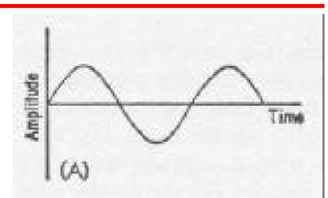
- main types of modification:

$$S(t) = A\sin(2\pi f t + \phi)$$

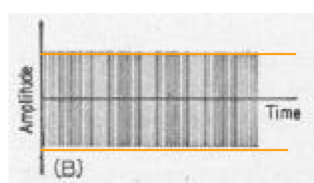
  - **amplitude modulation (AM)**
  - **frequency modulation (FM)**
  - **phase modulation (PM)**

Amplitude Modulation

- Given some signal to send:



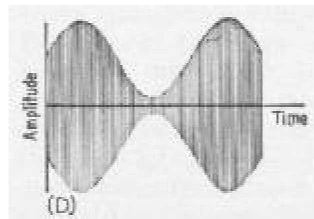
- and a carrier wave to carry it:



figures ©ARRL, 1973

Amplitude Modulation

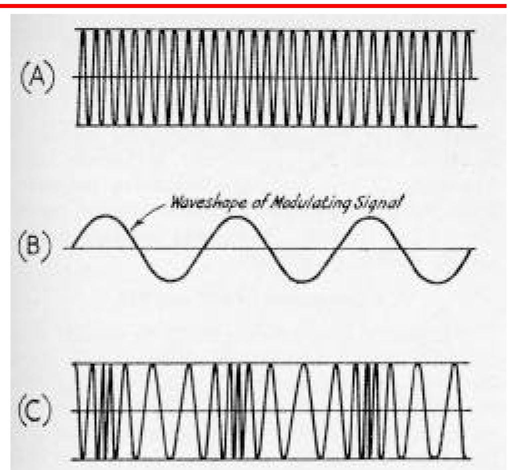
- we modulate the carrier with the amplitude of the signal we want to send:
  - envelope of carrier follows signal being sent



- dual sidebands
  - each can carry independent data
- what happens when signal amplitude is small?
- figure ©ARRL, 1973

Frequency Modulation

- start with carrier:
- signal to be transmitted:
- change f_c as $m(t)$



figures ©ARRL, 1973

Phase Modulation

- start with carrier
- change f_c as $m(t)$
  - frequency deviation in PM proportional to both frequency and amplitude of $m(t)$
  - in FM, frequency deviation proportional only to amplitude of $m(t)$
- as $|m(t)|$ increases, Δf increases, so does B_T but not overall power (both FM and PM)
  - what about AM?

Quadrature Amplitude Modulation

- use 2 copies of carrier, one is 90º out-of-phase with the other
- input bit stream split in 2: bit *i* to one carrier, bit *i* +1 to the other: sum and send
  - each sample represents 2 bits

Quadrature Amplitude Modulation

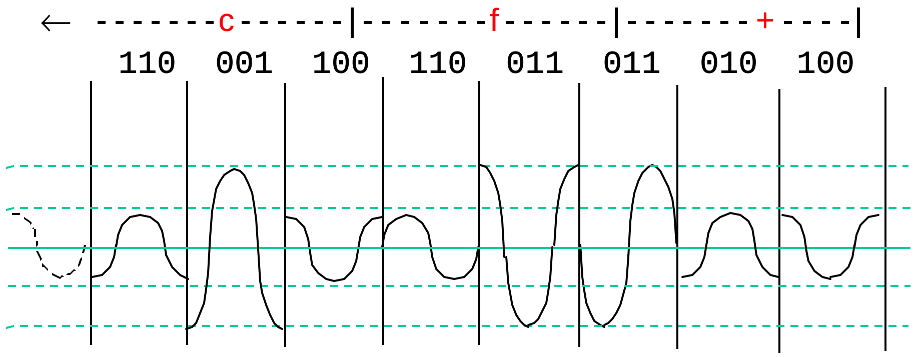
- if use multiple amplitude levels as well, can transmit more bits per signal sample
  - using 2 amplitudes, have 4 states
- combinations: more amplitudes, different phase shifts
  - can get 64, even up to 256, states

Quadrature Amplitude Modulation

- more states $\Rightarrow$ higher data rate for given B
- more states $\Rightarrow$ less discriminability between parts of signal representing each state $\Rightarrow$ higher error rates
- Given *n* levels of signal that can be discriminated in each sample based on amplitude, frequency or phase, the bit rate is: (*b* is sample rate or baud rate)

$$\boxed{C = b \log_2 n}$$

QAM (with even parity)



QAM: 3 bits / baud

Bit Combination	Phase Shift °	Amplitude
000	0	low
001	0	high
010	90	low
011	90	high
100	180	low
101	180	high
110	270	low
111	270	high

Analog to Digital

- suppose we have true digital transmission, but an analog signal
- convert analog signal via digitizing, or analog-to-digital (A to D) conversion
- performed by a circuit that codes (A to D) and can decode (D to A): a codec
- how to do the conversion?

A to D conversion

- we *sample* the analog signal at regular intervals
  - we have f_s, sampling frequency or sample rate
- if we sample analog signal S(t) with $f_s \geq$ twice the highest frequency appearing in S(t), then our samples contain all the information originally in S(t): this is the **Nyquist** rate
  - e.g., if voice signal is limited to 4,000 Hz, then sample at 8,000 samples/second

A to D conversion

- if we sample S(t) at regular intervals:

- and represent sample values as binary valued integers:

0111 1111 0011 1011

A to D: PCM & Quantization Noise

- the number of bits per sample affects the accuracy (resolution) of the digitized version
  - quantization error or quantization noise

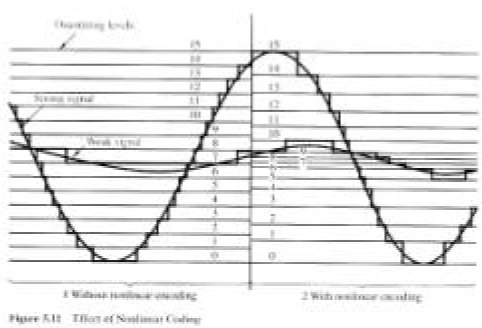
A to D: PCM & Quantization Noise

- how improve?
  - more bits (~ 6dB improvement to SNR [quantization noise] per added bit)
  - $\boxed{SNR_{PCM} = 20 \log 2^{n} + 1.76 \text{ dB}}$

    $= 6.02n + 1.76\text{dB}$

A to D: PCM & Quantization Noise

- how improve?
  - using non-linear coding: range of amplitudes is not even-stepped (e.g., Stallings Fig 5.11)
    - used in voice telephony



- figure ©Prentice-Hall 1996, 2000

A to D: Delta Modulation

- approximate analog signal with ‘staircase’
  - have sample time: width of a ‘stair’, T_s
  - have step size: height of a ‘stair’, d
- signal change in step, not sample value itself
  - each bit represents a change of $+d$ or $-d$
- generally poorer SNR performance than PCM at same data rate
- attractive because easy to build

A to D: Delta Modulation

- Fig 5.13
  - slope overload noise
  - quantization noise

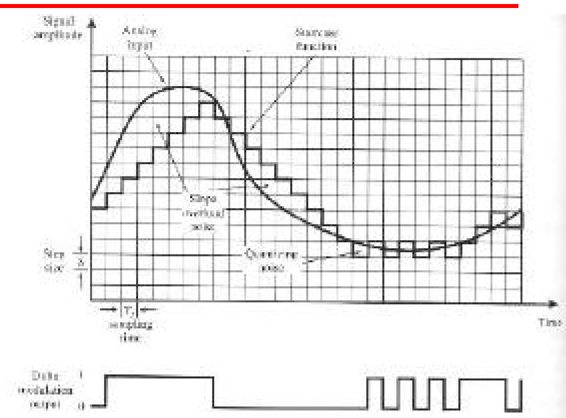


Figure 5.13 Example of Delta Modulation

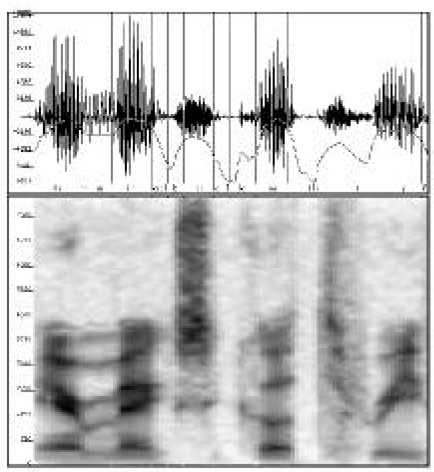
figure © Prentice Hall, 2000

A to D: Linear Predictive Coding

- used for digitized voice communication
- represent speech as progression of component speech sounds
- can achieve VDR (voice data rate) as low as 2.4 kbps

Speech

- "air mexico …"



- figure © C. Snow 1998

Digital Transmission of Voice

- sample analog speech at **Nyquist rate** $2f_h$
  - for phone, ~ 4 kHz, so sample at 8000 sps
- convert each sample to an 8 bit value (**PCM**)
- what bit rate do we need?
  - 8000 samples/second X 8 bits/sample = 64,000 bps
- a group of 24 such voice channels needs:
  - 24 X 64,000 bps = 1,536,000 bps
  - this fits on a **T1** carrier channel

C S 4 5 5 C O M P U T E R 10 SEP 2 0 0 2 © J MARK PULLEN 2001 03.35

Speech Coding: non-linear

- speech is coded using non-linear scale: <u>µ-law</u>
- 7 bits gives effect of 13

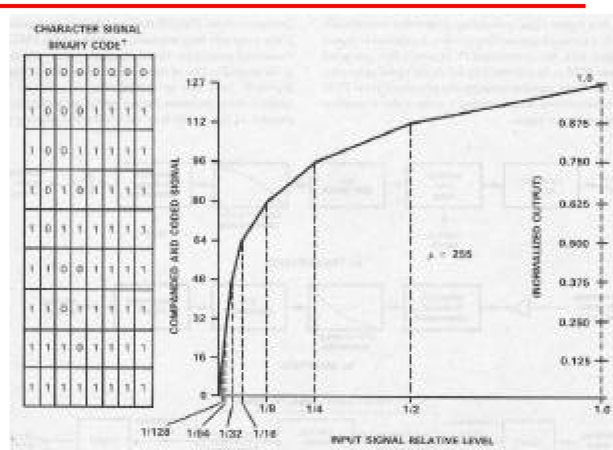


figure ©Texas Instruments 1986

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Digital Transmission of Music

- 8000 samples/second inadquate
- 8 bits per sample inadequate

- CD uses:
  - 44,100 samples/sec
  - 16 bit samples
  - 2 channels for stereo (interleaved channels)
  - what is bit rate of a CD player?

More Quantization Noise

- for speech encoded using 8-bit samples, what is SNR_{PCM}?
  - $SNR_{PCM} = 20 \log 2^n + 1.76$ dB
    $= 20 \log 2^8 + 1.76$ dB
    $= 49.9$ dB
  - noise can never be better than 49.9 dB below maximum signal level
- how about for a CD?
  $20 \log 2^{16} + 1.76$ db
  $= 98.1$ dB

Digital to Analog

- most familiar use: digital data xfer through voice-grade analog telephone lines
  - what bandwidth? spectrum?
- many signals (e.g., voice) that may have been digitized for transmission must be converted back to analog at receiver
- use device to receive digital and generate modulated analog (and *vv*): modulator-demodultor, or, **modem**

D to A

- analog signal to carry digital info: properties to use?
  - amplitude: ASK amplitude shift keying
  - frequency: FSK frequency shift keying
  - phase: PSK phase shift keying
- resulting signal occupies B centered on f_c

D to A: ASK

- use 2 amplitude levels, typically:
  - for bit value 0: 0
  - for bit value 1: $A\ \cos(2\pi f_c t)$
- good to ~ 1200 bps on voice grade lines
- used for driving LED transmitters on fibre optic
  - also for lasers, though these usually have low-level analogous of DC offset (‘bias’)

D to A: FSK

- use 2 frequencies typically:
  - for bit value 0: $A\ \cos(2\pi f_1 t)$
  - for bit value 1: $A\ \cos(2\pi f_2 t)$

  where f_1 and f_2 are offset from f_c by fixed amount in opposite directions
- less susceptible to error than ASK
- on voice-grade lines, used up to 1200 bps
- also used for radio transmission (3 to 30 MHz)
- use at higher frequencies in LANs using coax

D to A: PSK

- use 2 phases typically:
  - for bit value 0: $A\ \cos(2\pi f_c t)$
  - for bit value 1: $A\ \cos(2\pi f_c t + \pi)$

  to signal new bit value relative to previous one:

  this is differential PSK

- if bit is 0, send burst in same phase as previous
- if bit is 1, send burst 180 out of phase as previous

D to A: QPSK

- can also use quadrature to increase bit rate:
  - multiple phase angles
  - multiple amplitudes
- e.g., V .32 modem standard does 9600 bits per second at 2400 baud:

D to A: QPSK

- in general, quadrature allows increased bit rate per sample:

$$\boxed{D = \frac{R}{b} = \frac{R}{\log_2 L}}$$

  - D is modulation rate in baud
  - R is data rate in bps
  - b is # bits per signal element
  - L number of different signal elements

Digital to Digital

- how represent, as EM signals, digital quantities?
- need clocks to agree at sender and receiver

- simplest:
  - use one fixed voltage level for a 0,
  - a different fixed level for a 1
  - hold those fixed voltages for one ‘pulse time’
    - short: higher bit rates
    - long: lower error rates
  - called NRZ: non-return to 0

D to D: NRZ, NRZ-L, NRZI

- in practice, implement NRZ as:
  - negative voltage for 1 bit
  - positive voltage for 0 bit
  - this is called NRZ-L (non-return-to-zero-level)
- another variant: NRZI (NRZ, invert on 1s)
  - is a differential coding
  - if current bit is 0, use same level as preceding bit
  - if current bit is 1, use different level from previous

D to D: NRZ family

- intolerant to synchronization drift
  - what happens with long string of 1s or 0s?
- usually used for digital magnetic recording
  - not so well suited to transmission

D to D : Biploar-AMI

- bipolar with alternate mark inversion
- use 0 volts for 0 bit
- use $\pm v$ to signal 1 bit, alternating between $+v$ and $-v$ on successive 1s
  - avoid sync problems on long strings of 1s
  - what about long strings of 0s?
- allows for simple error detection
  - any erroneous insertion or deletion of a pulse violates alternating $\pm v$ property

D to D: NRZ vs. Bipolar-AMI

- how do these compare?
  - bipolar-AMI less sync error prone, provides simple error detection, has no net DC component
  - but uses 3 levels instead of NRZ's 2:
    - $\log_2(2) = 1$
    - $\log_2(3) = 1.58$
  - bipolar-AMI receiver needs 3dB stronger signal for same error rate as NRZ
    - or, for same SNR, NRZ has lower error rate

D to D: Manchester

- a <u>biphase</u> technique: do transition at mid-point of each bit period
  - acts as clocking mechanism
  - signals data:
    - low to high for 1 bit
    - high to low for 0 bit
- requires 2X bandwidth in medium

D to D: Differential Manchester

- do transition at mid-point of each bit period
  - midbit transition acts as clocking mechanism only
  - signals data:
    - if transition at start of bit period: 0 bit
    - if no transition at start of bit period: 1 bit

D to D: Biphase

- Advantages of biphase techniques:
  - self-clocking: mid-bit transition assures sync
  - no DC component in signal
  - error detection: missing transitions indicate errors
    - how could an error be missed?
  - good speed locally (10 Mbps), but inefficient for transmission over long distance (high D to R)

D to D: Biphase

- Manchester used in IEEE 802.3
  - baseband coax and twisted-pair CSMA/CD bus LANS
- Differential Manchester used in IEEE 802.5

D to D Summary

- Stallings fig 5.2

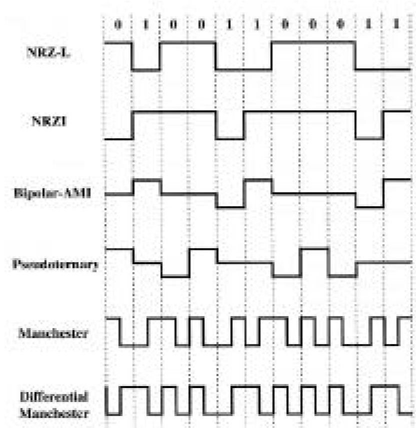


Figure 5.2 Digital Signal Encoding Formats

figure © Prentice Hall 2000

Data Compression

- if compress data to be sent, then re-expand on receipt, can get higher effective data rate for fixed signal rate (companding)
- introduces processing overhead at sender and receiver

Data Compression

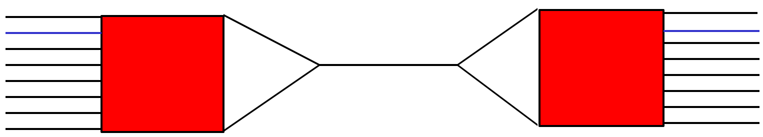
- **RLE: run length encoding**
  - replace long sequence of 1s or 0s with something like: <tag><count>
    - tag indicates what follows is not plain data but count of repeating 1s or 0s
    - count is number of 1s or 0s in a sequence
  - what benefit to NRZ? bipolar-AMI?
- **Ziv-Lempel compression** used in V .42 bis modems

Concentrators

- used to obtain high-utilization of links
  - let multiple stations share links
  - all senders served in turn (older technology)

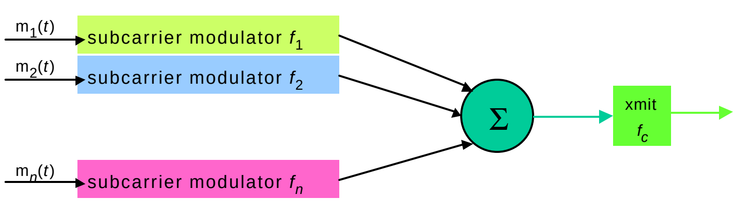
Multiplexing

- Multiplexing (muxing) allows multiple flows to share a channel within limits of overall capacity



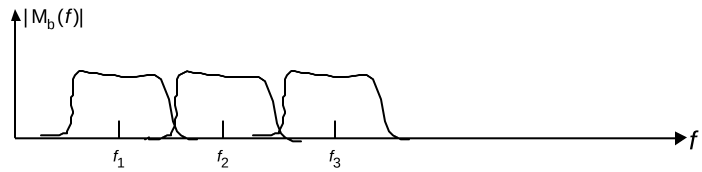
Muxing

- **FDM**: **frequency division** multiplexing
  - analogous to radio spectrum within a cable
  - not good for data due to noise from ‘baseband loading’



- $m_i(t)$ already band-limited
  - e.g., voice telephony: 3 kHz

Muxing: FDM



- separate bands may slightly overlap
  - hence need for ‘guardbands’ at sides

Muxing

- **TDM**: **time division** multiplexing
  - interleave bits from different slower streams into one faster stream
- STDM: statistical time division multiplexing
  - take avdvantage of idle time on link to run more TDM streams
  - not good for data, good for voice
- TDMA: time division multiple access
  - used with radio and satellite
  - transmitters take turns sending in closely spaced slots
  - wasteful of spectrum

Muxing

- **WDM**: **wavelength division** multiplexing
  - send multiple λ through fiber concurrently
  - up to 96 commonly used today

An application: FAX machines

- scanner + printer + modem in-a-box
- scanner digitizes page image
- digitized page image converted (back to) analog (but different kind) in modem for transmission over voice-grade telephone
  - why does computer-generated fax look better?
- extensive use of compression (e.g., RLE)
- can use protocols that take advantage of document characteristics (e.g, “group 3”)

Modern Transmission Systems

- most commercial systems are digital end-to-end
  - analog data converted to digital at or near sender
  - every amplifier along path restores digital signal to clean bits
  - digital data converted to analog at or near receiver
- what advantages from this?

Advantages of all-digital transmission

- result:
  - immunity to noise
  - lower cost
  - uniform data format
  - better security
  - better reliability
  - better control
- application:
  - 56 kbps modem: is digital from provider to user, all digital (hdx); is 33.6 kbps analog from user to provider… doesn't work everywhere!

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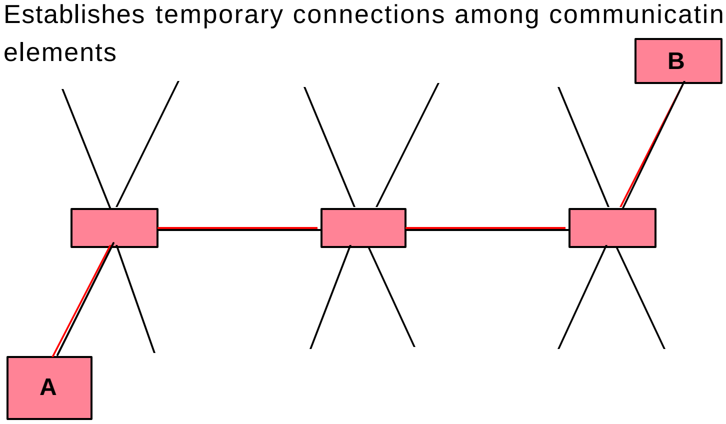
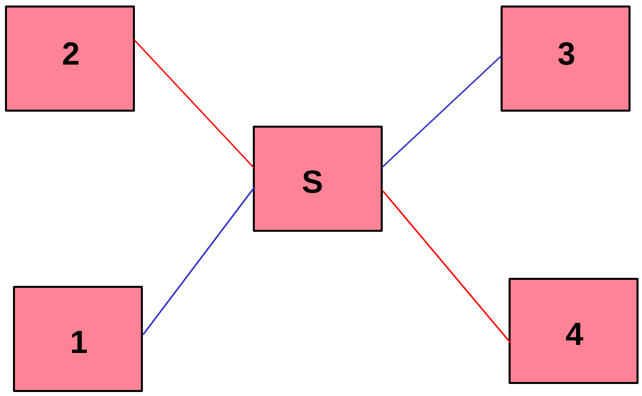
Circuit Switching

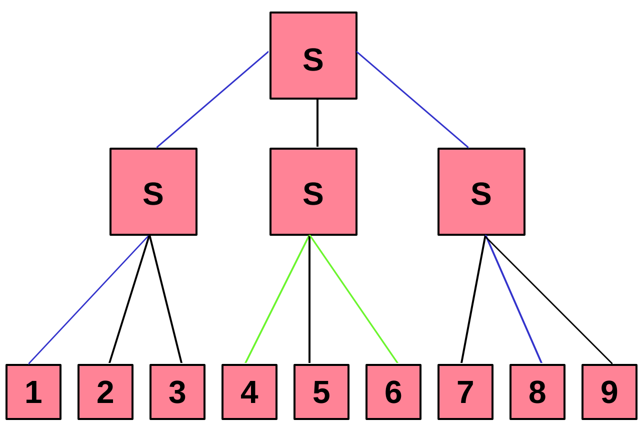
- Establishes temporary connections among communicating elements



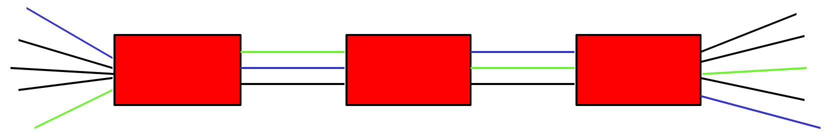
Hub Switching



Hierarchical Switching

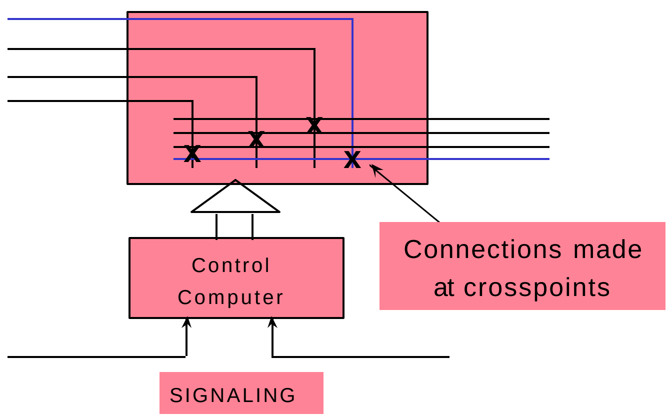


Trunk Circuit Switching



Concentration Connection Expansion

Inside a Circuit Switch



Circuit Switching for Data

- Real-time capability
- Call setup delay
- End system must place call
- Blocking (e.g., busy signal) possible
- Once you have a circuit you can use it until to choose to release it

A Familiar Circuit Switched Network

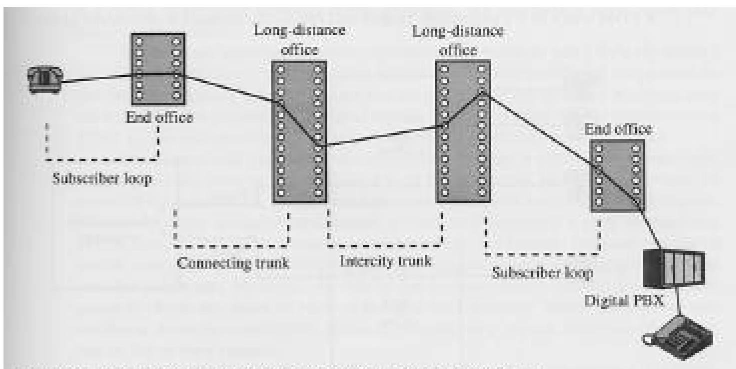


Figure 9.2 Example Connection Over a Public Circuit-Switching Network

- figure © Prentice Hall 2000

Circuit Switched Network Terminology

- **subscriber**: device attached to network (at endpoint), e.g., a telephone
- **subscriber loop**: link between subscriber and network
  - most are twisted-pair
  - typical range: few km to few 10s of km
- **exchange:** switching center on the network
  - exchanges directly supporting subscribers are **end-office**s
- **trunk:** links between exchanges
  - multiple voice frequency circuits
  - using FDM or synchronous TDM

Private Branch Exchange (PBX)

- a PBX is a small circuit switch providing:
  - local dial-up service
  - access to large system, like public switched system
- new PBXs are fully digital
  - interfaces for (analog) plain old telephone system (POTS) available
  - what would such an interface have to do?

Digital Line Hierarchy
(North America & Japan)

Name	Capacity	Voice Channels
DS0	64 Kbps	1
DS1 ("T1")*	1.544 Mbps	24
DS2	6.312 Mbps	96
DS3 ("T3")*	44.736 Mbps	672
DS4	139.264 Mbps	2,016

* normally available as leased service

- Europe uses different digital hierarchy, also based on 64 Kbps voice channels
  - e.g., E1 is 2.048 Mbps, 32 channels

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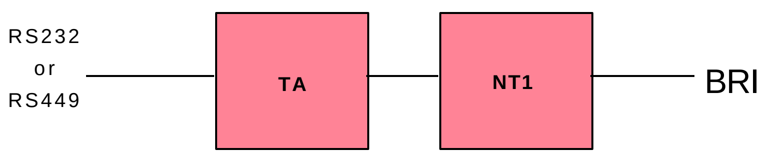
Integrated Services Digital Network (ISDN)

- common standard for “dial-up digital” circuits:

B channel	64 kbps
D channel	16 kbps (used for signaling)

- packaged as:
  - basic rate: 2B + D
  - primary rate: 23B + D (30B + D in Europe)
- “broadband ISDN”: 155 Mbps and up
  - using ATM
  - slowly becoming available

ISDN Components



- TA: Terminal Adapter
  - for non-ISDN directly compatible equipment
- NT1: Network Terminator
- Stallings Appendix A has details on ISDN

Synchronous Optical Network (SONET)

- a network of optical carriers installed by the common carriers for most long-distance trunks
- data rates occur at multiples of 51.84 Mbps, called Optical Carrier 1 (OC-1)
- commonly available data rates include:
  - OC-3 ~155 Mbps
  - OC-12 ~622 Mbps
  - OC-24 ~1.2 Gbps
  - OC-48 ~2.4 Gbps

Broadband ISDN

- operates over SONET
- uses cell-switching technique: **asynchronous transfer mode (ATM)**
- sends 53-byte cells (fixed-sized packets) across SONET links between cell switches
- cell paths requested using ISDN call setup
- cells sent into network, switched at cell switches, then brought brought out of network at dest.

Class 3 Overview

- Character transmission with parity
- Signal gain and loss: dB
- Signals and Transmission
- Switching
- ISDN
- Other Transmission Methods
- Homework & Project

ATM and AAL

- **ATM Adaptation Layers (AAL)** integrate application data with cell structure
  - AAL1: constant bit rate
  - AAL2: variable bit rate
  - AAL5: available bit rate
- more on ATM later in course…(stay tuned)

Asymmetric Digital Subscriber Line (ADSL, DSL)

- approach new in 1995
- basic idea: asymmetric speeds reflect usage:
  - high capacity to subscriber (≤ 9 Mbps)
  - low capacity from subscriber (≤ 1 Mbps)
- may also provide voice telephony by muxing
- runs on std copper wire up to ~ 3 miles/5 km from telephone office
  - longer distance, lower data rate
  - 3 miles/5km: 1.5 Mbps
  - 1.5 miles/2.5 km: 9 Mbps

Cable Modem

- cable TV originally unidirectional
  - all signals flow from "head end" through tree of wire, fibre, and distribution amplifiers
  - practically no capacity for flow back to head
- contemporary cable TV bidirectional
  - competing for Internet service to home
- subscriber (at home) connects via **cable modem**

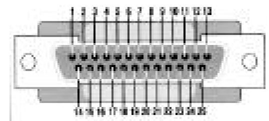
Cable Modem

- the cable modem:
  - bypasses (is independent of) home cable converter
  - provides bit rates of hundreds of kbps to/from Internet
  - upstream transmissions contend for shared channel
  - mechanism similar to Ethernet (we see later)

Physical Interfaces

- EIA-232-D (**RS-232**)
  - most common **serial** interface
  - asynchronously: as few as 5 wires
  - normal limit 20 kbps (though some over 100 kbps)
  - see Stallings Fig 6.5, table 6.1
  - note variety of standards involved:
    - mechanical (ISO 2210)
    - electrical (V.28)
    - functional (V.24)
    - procedural (V.24)

RS232 Interface Standard



- 25 wire standard:

1	Frame Ground	14	2ry transmitted data
2	Transmitted Data	15	xmit sig element timing
3	Received Data	16	2ry received data
4	Request to Send	17	rcvr sig element timing
5	Clear to Send	18	undefined
6	Data Set Ready	19	2ry request to send
7	Signal GndCommon	20	data terminal ready
8	Recv'd Line signal Detect	21	signal quality detector
9	RFU	22	ring detector
10	RFU	23	data sig rate select
11	undefined	24	xmit sig element timing
12	2ry rcv'd line signal dtx	25	undefined
13	2ry clear to send		

- image from `http://www.arcelect.com/rs232.htm`

Physical Interfaces

- EIA-449 (**RS-449**)
  - higher data rates (up to 2 Mbps)
  - balanced line capable
  - common on 56/64 kbps and T1/E1 links
  - built-in loopback capability
  - variations include:
    - RS-422
    - V.32

Class 3 Overview

- Character transmission with parity
- Signal gain and loss: dB
- Signals and Transmission
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- ISDN
- Other Transmission Methods
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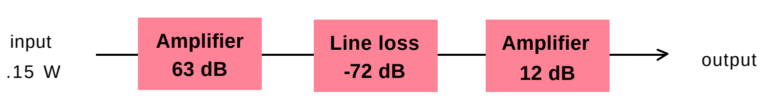
Homework Problems

1. Express in dB the gain of an amplifier with output of 75W, when the input is 150 mW
2. If attenuation results in an output = .013 X input (measured in Watts), express this loss in dB.
3. Sketch the QAM signal for “nEt” in 8-bit ASCII with even parity. Do also for Manchester.
4. Calculate the ratio of signal to quantization noise for 24-bit PCM encoding. How does this compare to ordinary audio CD?
5. For the figure below, (a) calculate the overall dB and (b) find the output

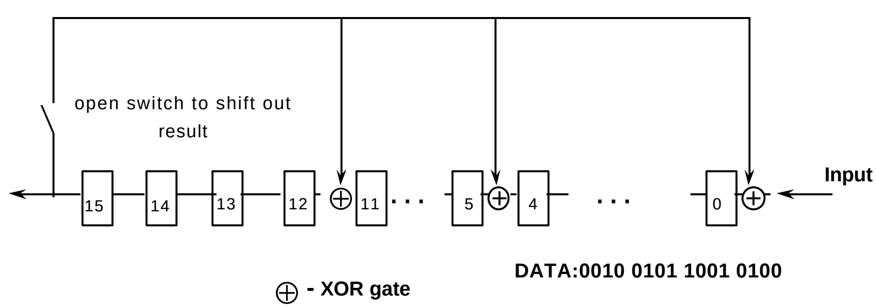


Project DLC2

- `FCS stack::generate_FCS (bit_frame* FCS_frame)`
  - Given a bit frame delimited by two flags and with a 16-bit CRC placeholder (immediately preceding the closing flag), compute a 16-bit Cyclic Redundancy Check Frame Check Sequence using the CCITT 0-5-12-16 polynomial. Return the 16-bit FCS.
  - Do not include the flags in the CRC computation.
- `code/crc.cpp` contains function stub and algorithm
- See *UIP* Chapter 4 for details.

FLAG	Address	Control	Data	CRC-FCS	FLAG

Hardware CRC Generation Circuit
Polynomial: $D^{16} + D^{12} + D^5 + 1$



uncomment `CRC_example();` in `dlc2.cpp` to see how this works