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: \textbf{X100 1011}
  - has an even number of 1's (4)
  - so even parity version is: \textbf{0100 1011}
  - odd parity version is: \textbf{1100 1011}
Class 3 Overview

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

Gain/Loss in dB

- **decibel**: dB
  - unit of signal strength based on *ratio of power*:
  
  \[ S_{dB} = 10 \log_{10} \left( \frac{\text{measured power}}{\text{reference power}} \right) \]

- signals strength is multiplied through cascaded blocks; using dB these strengths **add**
- note: 3dB gain means twice as ‘loud’
Using db

1. amplifier making a signal 100 times stronger has gain of:
   \[ 10 \log (100) = 20 \text{ dB} \]

2. amplifier gain of 44 dB is a \( \sim \) 25,000 X boost:
   \[ \frac{P_m}{P_r} = 10^{(44/10)} = 25,119 \]

Using db

3. a signal reduced to 5% of original strength has:
   \[ 10 \log(0.05) = -13 \text{ dB} \]

4. an amplifier produces 1.0 watt for an input of 0.5 watts has gain:
   \[ 10 \log (1.0/0.5) = 10 \log(20) = 13 \text{ dB} \]
dB and electrical strength

- If signal intensity measured in volts, then:
  \[ P = VI = V^2/Z \]
  So:
  \[ 10\log\left(\frac{V^2_{\text{meas}}}{V^2_{\text{ref}}/Z}\right) = 20\log\left(\frac{V_{\text{meas}}}{V_{\text{ref}}}ight) \]

Signal Strength in dB, example

- Overall gain is:
  \[ 15 \text{ db} + (-40 \text{ db}) + 22 \text{ db} = -3 \text{ db} \]
- So change in power over channel is:
  \[ -3 \text{ dB} = 10^{\frac{-3}{10}} = 10^{-0.3(10)} = 0.5 \]
- Input signal 0.5 watts X 0.5 gain factor = 0.25 watts
Class 3 Overview

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

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:
    - A
      - AM, FM, PM...
    - D
      - PCM, DM...
  - input is:
    - A
      - ASK, FSK, PSK...
    - D
      - 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:
  - amplitude modulation (AM)
  - frequency modulation (FM)
  - phase modulation (PM)

\[ S(t) = A \sin(2\pi ft + \phi) \]

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 \( \phi_c \) 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, \( \Delta f \) 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 90° 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)

$$C = b \log_2 n$$
QAM (with even parity)

QAM: 3 bits / baud

<table>
<thead>
<tr>
<th>Bit Combination</th>
<th>Phase Shift °</th>
<th>Amplitude</th>
</tr>
</thead>
<tbody>
<tr>
<td>000</td>
<td>0</td>
<td>low</td>
</tr>
<tr>
<td>001</td>
<td>0</td>
<td>high</td>
</tr>
<tr>
<td>010</td>
<td>90</td>
<td>low</td>
</tr>
<tr>
<td>011</td>
<td>90</td>
<td>high</td>
</tr>
<tr>
<td>100</td>
<td>180</td>
<td>low</td>
</tr>
<tr>
<td>101</td>
<td>180</td>
<td>high</td>
</tr>
<tr>
<td>110</td>
<td>270</td>
<td>low</td>
</tr>
<tr>
<td>111</td>
<td>270</td>
<td>high</td>
</tr>
</tbody>
</table>
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:

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)
  - \[ SNR_{PCM} = 20 \log_2 n + 1.76 \text{ dB} \]
    \[ = 6.02n + 1.76\text{dB} \]

- 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’, $\delta$

- signal change in step, not sample value itself
  - each bit represents a change of $+\delta$ or $-\delta$

- generally poorer SNR performance than PCM at same data rate
- attractive because easy to build

Fig 5.13

- slope overload noise
- quantization noise

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 …”
Digital Transmission of Voice

- Sample analog speech at Nyquist rate $2f_n$
  - 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

Speech Coding: non-linear

- Speech is coded using non-linear scale: μ-law

- 7 bits gives effect of 13

Figure ©Texas Instruments 1986
Digital Transmission of Music

- 8000 samples/second inadequate
- 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 $\text{SNR}_{\text{PCM}}$?
  - $\text{SNR}_{\text{PCM}} = 20 \log 2^8 + 1.76 \, \text{dB}$
  - $= 20 \log 2^8 + 1.76 \, \text{dB}$
  - $= 49.9 \, \text{dB}$
  - noise can never be better than 49.9 dB below maximum signal level

- how about for a CD?
  - $20 \log 2^{16} + 1.76 \, \text{dB}$
  - $= 98.1 \, \text{dB}$
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-demodulator, 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:

\[ D = \frac{R}{b} = \frac{R}{\log_2 L} \]

- \( D \) 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: Bipolar-AMI

- bipolar with alternate mark inversion
- use 0 volts for 0 bit
- use $\pm \nu$ to signal 1 bit, alternating between $+\nu$ and $-\nu$ 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 $\pm \nu$ 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 biphase 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
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)

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

![Digital Signal Encoding Formats](Prentice_Hall_2000.png)

**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’

- $m_1(t)$ 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 advantage 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 $\lambda$ 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!
Class 3 Overview

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

Circuit Switching

- Establishes temporary connections among communicating elements
Hub Switching

Hierarchical Switching
Trunk Circuit Switching

- Concentration
- Connection
- Expansion

Inside a Circuit Switch

- Connections made at crosspoints
- Control Computer

SIGNALING
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 you choose to release it

A Familiar Circuit Switched Network
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-offices**

- **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)

<table>
<thead>
<tr>
<th>Name</th>
<th>Capacity</th>
<th>Voice Channels</th>
</tr>
</thead>
<tbody>
<tr>
<td>DS0</td>
<td>64 Kbps</td>
<td>1</td>
</tr>
<tr>
<td>DS1 (&quot;T1&quot;)*</td>
<td>1.544 Mbps</td>
<td>24</td>
</tr>
<tr>
<td>DS2</td>
<td>6.312 Mbps</td>
<td>96</td>
</tr>
<tr>
<td>DS3 (&quot;T3&quot;)*</td>
<td>44.736 Mbps</td>
<td>672</td>
</tr>
<tr>
<td>DS4</td>
<td>139.264 Mbps</td>
<td>2,016</td>
</tr>
</tbody>
</table>

* 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

Class 3 Overview

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

<table>
<thead>
<tr>
<th>No.</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Frame Ground</td>
</tr>
<tr>
<td>2</td>
<td>Transmitted Data</td>
</tr>
<tr>
<td>3</td>
<td>Received Data</td>
</tr>
<tr>
<td>4</td>
<td>Request to Send</td>
</tr>
<tr>
<td>5</td>
<td>Clear to Send</td>
</tr>
<tr>
<td>6</td>
<td>Data Set Ready</td>
</tr>
<tr>
<td>7</td>
<td>Signal Gnd/Common</td>
</tr>
<tr>
<td>8</td>
<td>Recv’d Line signal Detect</td>
</tr>
<tr>
<td>9</td>
<td>RFU</td>
</tr>
<tr>
<td>10</td>
<td>RFU</td>
</tr>
<tr>
<td>11</td>
<td>undefined</td>
</tr>
<tr>
<td>12</td>
<td>2ry received data</td>
</tr>
<tr>
<td>13</td>
<td>2ry clear to send</td>
</tr>
</tbody>
</table>

* Image from [http://www.arcelect.com/rs232.htm](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
- Switching
- ISDN
- Other Transmission Methods
- Homework & Project

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

   input .15 W  Amplifier 63 dB  Line loss -72 dB  Amplifier 12 dB  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: \( D^{16} + D^{12} + D^5 + 1 \)

```
open switch to shift out result
\( \oplus \) - XOR gate
DATA:0010 0101 1001 0100
```

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