COMP3721 Week Five Lab Synopsis

Signal Transformations

- In terms of data communication, we are often interested in transforming a signal from its original form to
  - a form better suited to the channel characteristics
  - better matched to channel BW
  - more noise-resilient form
  - frequency translations
  - etc.

- A simple example of this type of transformation would be a microphone that translates acoustic amplitudes (variations in air pressure) to electrical amplitudes (variations in voltage)
  - speakers do the opposite

- In general, we can consider four different categories of tranlation
  - Analog --> Analog
  - **Analog --> Digital (quantization)**
  - **Digital --> Analog (modulation)**
  - Digital --> Digital

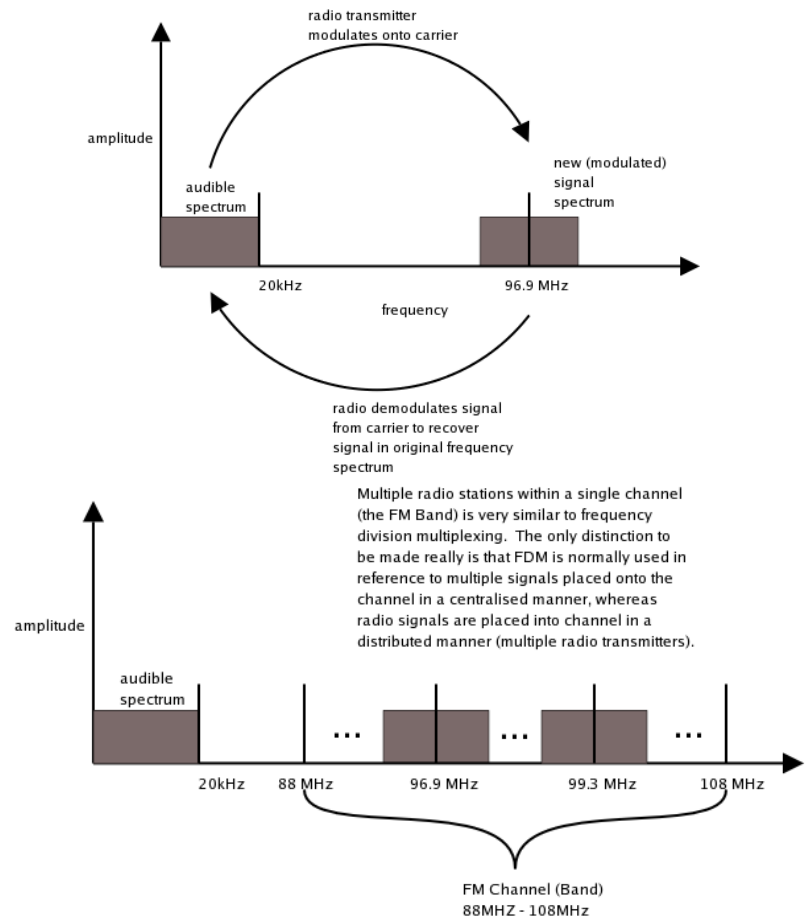
- The two middle transformations receive the greatest focus in this course, but consider examples of the outer cases:

  ### *Analog --> Analog*

  - Radio – voice and music are natural analog signals with frequencies in the 20Hz – 20 kHz audible frequency band
    - Note: the most significant frequencies in voice are between 20Hz – 3.5 kHz
    - Note: the most significant frequencies in most instruments are between 20Hz – 10 kHz
  - Considering radio as a service, there are several requirements:
    - Multiple signals (stations) must be simultaneously available
    - Signals must propagate significant distances
    - The signals should not interfere with one another or otherwise (e.g. we don't want to hear radio signals unless tuned in with a radio)
  - The above service cannot be provided with the signal in it's natural state. Solution:
    - Translate each analog signal (station) to a unique, non-overlapping frequency band outside of the audible spectrum
    - Each station is assigned a carrier frequency (i.e. 96.9 Mhz – JACK FM, 99.3 Mhz Fox) and modulates their signal onto that

carrier at the radio transmitter

- Radios translate the modulated signal (from a given carrier frequency/radio station) back down to the audible frequency spectrum and pass the signal through a speaker to produce audible output



Digital --> Digital

- Consider the unipolar encoding scheme used internally on the motherboard of a computer for bus signals:
  - 0V --> logical 0
  - 3.3V --> logical 1
- The devices on either side of the bus must agree on the where each bit starts in time
  - Over small distances this can be achieved through shared clocking (i.e. the bus clock on a motherboard)
  - Over greater distances, synchronization of clocking is a difficult problem to solve
  - A receiver can use transitions (edge-detection – think back to computer architecture and flip-flops; low to high or high to low)

    in the signal to maintain synchronization, but unipolar encoding does not guarantee transitions in the event of a long string of 0s or 1s.
  - Solution: translate unipolar digital signal into Manchester encoded signal
    - Advantages:
      - Guarantees regular signal transitions regardless of bit pattern
      - Also, polar encoding removes DC component (0Hz signal) – DC components have poor propagation characteristics
    - Disadvantage:
      - Requires double the bandwidth

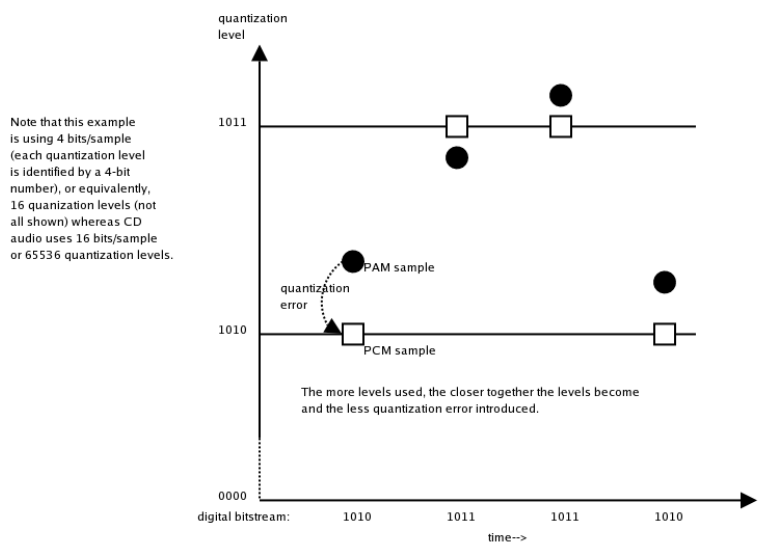
Audio CD – Analog to Digital Transformation

- Want to capture all audible information for reproduction
- Signal is naturally analog, we want to store it in a digital manner
  - Analog --> Digital
- As outlined last week – two phases to process
  - we must understand the appropriate values necessary for each phase
    - Step one: sampling – we need to know the sampling rate to use. This depends on the signal bandwidth.
    - Step two: quantization – each sample will be mapped to a quantization level. This introduces quantization error.
      - The more quantization levels, the less quantization error/noise; less levels, greater quantization noise
      - # of levels is then a matter of quality – we must have some measure of the quality expected for the digital version of the signal

- Step One:
  - We want to capture all audible information, so the signal frequencies of interest are 0 Hz – 20 kHz
  - By Nyquist, we must sample at a minimum of 2H, or 2 * 20 kHz = 40 kHz = 40 000 samples/second
  - For practical reasons (we don't need to worry about why here), signals are often *oversampled* (sampled above the Nyquist rate) – in the case of the CD standard, the actual sampling rate used is 44.1 kHz

- Step Two:
  - It should be clear that for CD audio, a reasonably high quality level would be expected.
  - Ultimately in any system, there are various sources of noise – the quantization error under consideration here is only one of these

  noise sources
  - Philips decided that a Signal to Quantization Noise ratio of 94.75 decibels would be sufficient
    - 94.75 db = 10 $\log_{10}$(S/N), then S/N = 2.95 * 10^9
      - that is, expressed in a linear manner, the signal power is about 3 billion times greater than the power of the quantization error introduced
    - This is essentially imperceptible for most systems/people. In all but the best systems, the noise from other system components (speakers, amps, etc) will be far more significant than the quantization error
    - 94.75 dB = 6n – 1.25, n = 96/6 = 16 bits/sample
      - n = 16 bits/sample, then V = 2^{16} quantization levels
      - Remember: S/N_Q(db) = 6n – α relates the number of bits/sample (and thus the number of quantization levels) to the amount of quantization error introduced



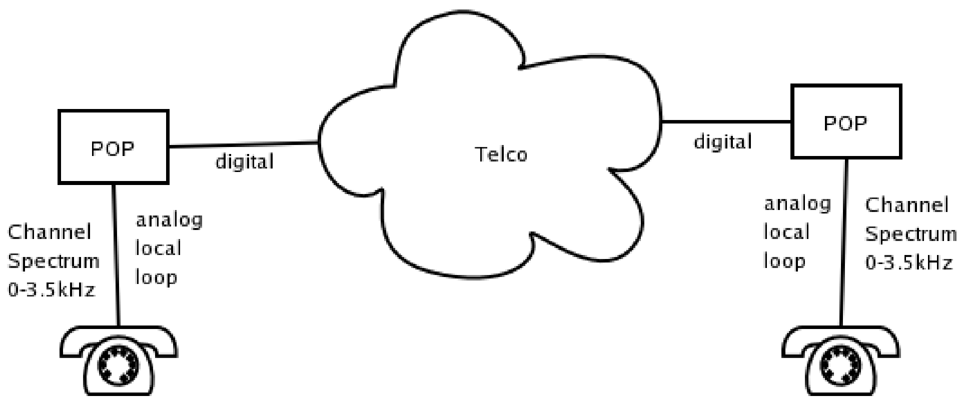
- Putting the two steps together, we get a digital bitrate:
  - CD Audio Data Rate (one channel) = 44.1 kHz * 16 bits/sample = 705.6 kbps => 88.2 kBps
  - The above would be for a single channel, or mono signal. CD Audio stores a left and right signal (stereo) in an interlaced manner (left channel sample_0, right channel sample_0, left channel sample_1, right channel sample_1, ...).
  - CD Audio Data Rate (stereo) = 2 * 88.2 kBps = 176.4 kBps
    - This may look familiar – it is the 1X CD rate; it is the rate the CD player must spin the disk and pick up data for normal CD audio playback
    - For data transfer, CDs can spin faster (and therefore read/write

faster) – which is why you have 16X, 32X and higher CD spinrates.

- An Audio CD is then just a stream of interleaved PCM samples. This is also the format of a WAV file. This is the reason your operating system (generally) presents the tracks of an Audio CD as a collection of WAV files

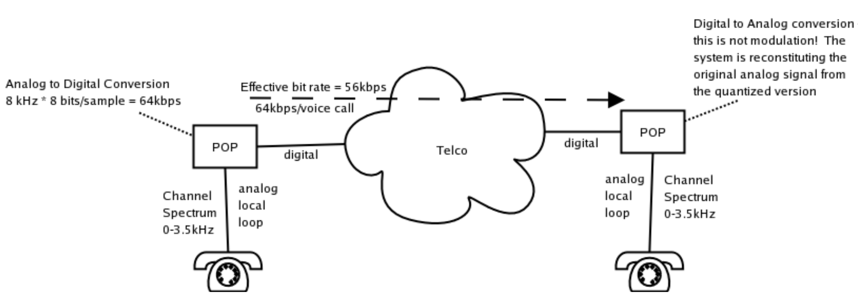
The Phone System

- Consider the following **block diagram** of the phone system:



- The POP is the Point-of-Presence, the first step into the phone company's network.
  - There are numerous POPs throughout a geographic area, each serving a different neighbourhood.
  - All of the residential phone lines (local loops) within a given geographic area go to a single POP
- The local loop is generally referred to as an analog channel. Why?
  - *First and foremost, remember that the phone system exists to carry voice traffic. Voice traffic is analog in nature and the phone company assumes the phone line is being used to transfer this form of analog signal.*
  - A piece of cable itself is not analog or digital in nature.
  - The channel bandwidth provided by the cable (and its associated equipement) tends to dictate whether it is used to carry analog signals (generally lower channel BW) or digital signals (greater BW requirements)
  - Local loop lines were originally installed to carry voice traffic in the range of 20Hz to 3.5 kHz.
    - Noise characteristics within this band are good
    - So too, attenuation and delay distortion
    - A filter is actually installed on the line to restrict bandwidth to these 'expected' frequencies (see section on quantization of voice traffic below for the reasoning behind this).

- Because of the very limited channel BW, we are effectively restricted to analog transmission, even when we want to move digital data (think computers interconnected with modems)
  - See practice problem 3 for more on this
- Pretty much everywhere throughout the world, telephone companies run digital networks – all signals, both voice and data, are carried as digital signals.
- The POP must then perform an analog to digital transformation (quantization again)
  - Step One: by Nyquist the sampling rate should be 2 * 3.5 kHz = 7 kHz. This requires that the signal be bandlimited to 3.5 kHz, thus the line filter mentioned earlier.
    - Again, the standard actually calls for oversampling at **8 kHz**
  - Step Two: quality. Because the equipment (i.e. phones) is low quality (cheap microphone, cheap speaker) and the line already is somewhat noisy, a much lower standard is necessary than with CD audio.
    - Telcos have standardized on 8 bits/sample or equivalently 256 quantization levels.
  - Digital data rate for a phone line = 8 kHz * 8 bits/sample = 64kbps
  - The 64 kbps rate is the fundamental rate underlying all data rates within the phone system. Every other data rate related to phone company services is essentially a multiple of 64 kbps.
    - While physical constraints require a local loop cable from every customer to a Telco POP, the phone company does not run individual cables for each customer throughout their networks. This implies that they carry more than one signal per cable elsewhere in their network. For digital signals this is accomplished through time-division multiplexing (TDM). This is one of next week's topics.
  - While the Telco propagates 64 kbps / voice call, they also steal some of this for internal signaling (i.e. whether the call has been disconnected, quality of service, routing).
    - the rate at which Telcos steal from the 64 kbps channel rate varies, but in the worst case it is 1 bit/sample reducing the customer digital data rate (effective data rate) to 8 kHz * 7 bits/sample = 56 kbps.



- Now imagine trying we want to move digital data through the plain old telephone system (POTS).
  - We cannot never 56 kbps – even if we managed to modulate more than this over the local loop, the extra data would be lost within the digital portion of the network.
  - Because the local loop is an analog channel (as established earlier), we must convert the digital data to analog (modulation)
  - From earlier, the digital data rate possible through an analog channel depends on bandwidth, total channel S/N (this includes all forms of noise within the channel – quantization error is one source, but definitely not the only source of noise), and the encoding scheme used.
  - Typical phone lines provide total S/N of 30dB.
  - the channel spectrum is between 300 Hz and 3.5 kHz – then the channel BW = 3.5 - 0.3 kHz = 3.2 kHz
  - Then by Shannon, the maximum possible DTR = B * $\log_2(1+S/N)$ = 3.2 kHz * $\log_2(1+1000)$ = 3.2 * 10^3 * 9.97 = 31.9 kbps
  - With a more precise S/N value we would obtain the actual upstream limit of 33.6 kbps.
  - How then do 56 kbps achieve the higher data rate and why is it only in one direction?
    - The bandwidth is essentially fixed, so the channel S/N must be better in one direction
    - 56kbps ISPs are connected to the Telco via a digital connection – they have upgraded their local loop connections.
    - The phone company does not quantize the digital signal coming from the ISP, they instead pass it directly into their digital network. Because there is no A-->D conversion, there is no quantization error and a better

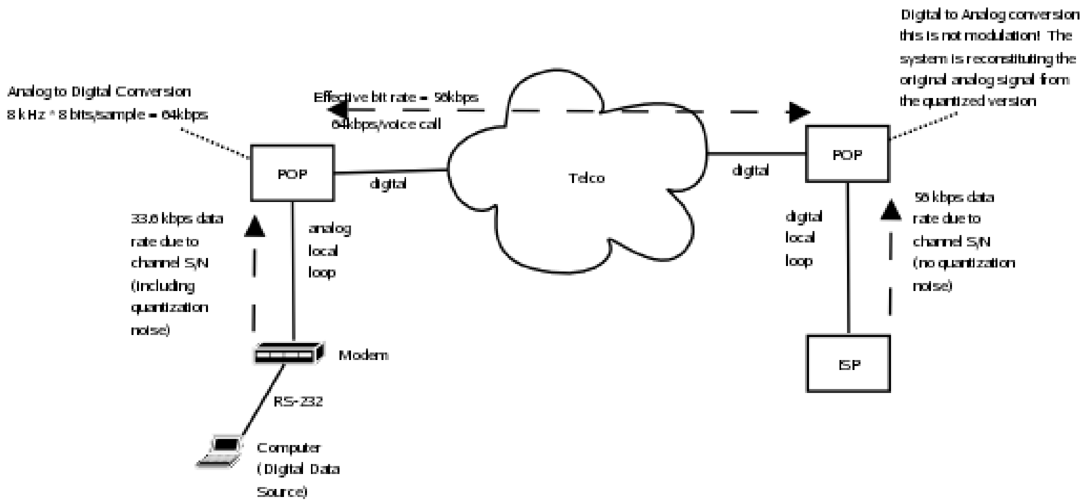
A Digital Local Loop?

If the rest of the phone system is digital, why not change the local loop to be digital as well?

Two key factors are involved:

1. cost – there is a lot of copper wire out there. It was engineered to carry frequencies between 0-3.5 kHz and does so very well, but would have to be replaced for effective digital transmission.
2. Voice is analog in nature. You can buy a phone for $10 because the analog components are cheap. In contrast, if the phone line is digital, then the phone becomes much more complex – it must perform the quantization (and signal reconstitution)that is currently done at the POP.

In fact, there was a standard developed for digital phone traffic – ISDN. And how many of you have ISDN lines?

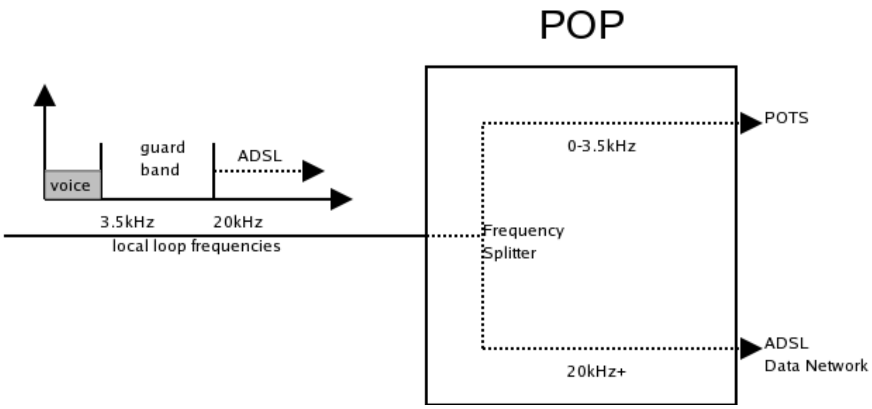


channel S/N.

ADSL

- How then does ADSL break these limits?
  - The signal power used on the channel cannot be increased for a number of reasons
  - Noise has effectively been minimized
  - The only remaining factor is bandwidth
    - ADSL achieves higher data rates by removing the 3.5 kHz channel bandwidth limit

- The 3.5 kHz filter is effectively removed
- Voice traffic is carried as usual (0-3.5 kHz)
- Modulated digital data is carried in a number of channels starting at 20kHz.
  - This is not a trivial undertaking; as mentioned before, the channel characteristics (noise, attenuation and delay distortion) are very poor outside of the voice band.
  - Frequencies between 3.5 – 20 kHz are unused to avoid overlap (guard band) and because digital data modulated in the audible spectrum (0 – 20 kHz) would be heard on the transmitter side.
- The phone company upgrades the POP as seen below. This is the reason ADSL takes time to become available in different areas – the Telcos must upgrade their equipment (at the POP) to support the ADSL traffic.



Practice Questions

1. What is the bit rate for a 40X CD reader?
2. The disc format used for writing audio CDs provides approximately 800 MB of storage (the audio format uses less redundancy bits for error correction than the ISO9660 data format). Estimate the amount of stereo music that can be stored on a CD.
3. As a rough measure, to send square-pulse polar signals through a channel (such that the receiver can reliably decode them) requires bandwidth approximately equal to the bit rate. For example, to send at 300 bps requires approximately 300 Hz of bandwidth. What bit rate would then be possible using digital signaling over the local loop to the POP?
4. The phone company effectively uses 7 bits/sample (once the rob bit is accounted for) for voice signals. What is the S/N_Q in decibels?
5. Is modulation the inverse function of a quantization?
6. Assume the maximum possible baud rate is used to achieve the 33.6 kbps bit rate on over the local loop. How many levels are being used in the modulation scheme?

Answers to Practice Questions

1. Bit rate = 40 * 176.4 kBps = 7.056 Mbps
2. 800 MB / 0.1764 Mbps = 4535 seconds => 75.6 minutes
3. 3.5 kHz of channel BW => 3.5 kbps. Note that this is only about 1/10th of the Shannon capacity of the channel.
4. S/N_Q = 6 (7 bits/sample) – 1.25 = 40.75 dB'
5. No. Modulation is the process of encoding digital data onto an analog carrier. It is not designed to generate an audible signal. Even when the modulated signal is within the audible spectrum, it would not 'make sense' to someone listening.

   Quantization is (generally) used in the process of encoding a signal from the audible spectrum as digital data. In transforming the digital data back into an analog signal, it must be reconstituted specifically in its audible form. Essentially this is a specific form of digital data (amplitudes) that has a specific form as an analog signal.
6. By Nyquist:
   33.6 kbps = 2 * 3.5 kHz * $\log_2 V$
   $\log_2 V$ = 4.8 bits/baud

   Now we cannot modulate a fractional number of bits onto each signal element (baud). To round up (to 5 bits/baud) would push us up over the Shannon limit. In this case, errors would occur and the actual rate of data transfer would fall back below the Shannon limit.

   To round down (to 4 bits/baud) would result in a data rate within the Shannon limit, but also under the expected 33.6 kbps data rate.

   One solution here is to use 5 bits/baud, or **2^5 = 32 modulation levels** and a baud rate slightly lower than the maximum (33.6 kbps / 5 bits/baud = 6720 baud per second)