COMP3721 Week Six Lab Synopsis

Channel Impairments

- We've previously related the bit rate of a system to several key factors: bandwidth, modulation scheme, and S/N
- Several other practical considerations effect the achieved bit rate

Attenuation

- Any medium has a certain amount of resistance within it that reduces the amplitude of signals as they move though that channel
- In the simplest case we can use one of the following to counteract attenuation:
  - amplifier – the amplitude of the input signal is increased by some factor called the gain. Gain is normally measured in dB rather than linearly.
    - Note that the amplifier never interprets the signal in any way. It is an analog device – any noise that is added into the signal is amplified along with the original signal
  - repeater – a repeater is designed for digital signals. It must be aware of the digital encoding scheme used on the line. It then interprets each digital symbol on the input and regenerates a fresh, full-amplitude symbol on the output
- In the above we are essentially assuming that all frequencies components within a signal are attenuated equally. This is not the case.
- Consider what the following graph from the website tells us about the voice band over the local loop:
  - This graph shows *relative* attenuation.

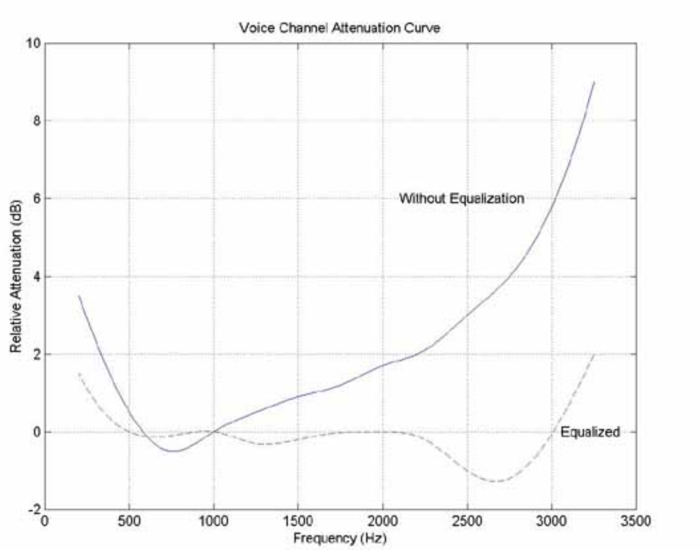
Measuring Attenuation and Gain

Attenuation or gain are measured as the power (power is a measure of amplitude over time) of a signal at one point relative to the power of that signal at another point. For transmission systems, this is normally received power versus transmitted power. For example, if the received signal is 10W but the transmitted signal was 100W, significant attenuation has occurred and the ratio is 10W/100W.

Again, engineers prefer to measure this type of relationship in decibels, so it would more commonly be expressed as:

$10\log_{10}(10W/100W) = -10$ dB

It should be clear that attenuation will be negative when expressed in decibels, gain will be positive.



- Specifically, we want to start by considering the channel's direct effect on signals – this is the line labeled “without equalization”
  - 1 kHz and ~600Hz are at 0 dB
    - *This does not mean that 1 kHz and 600 Hz components are not attenuated.*
    - Instead, all other attenuation measures are relative to the attenuation experienced by these frequencies
  - 2 kHz, as an example, is at about 2 dB
    - that is, 2 kHz it suffers 2 dB greater attenuation than 1 kHz
      - 2 dB as a linear ratio is $10^{0.2}$:1 =1.58:1; that is 2 kHz is attenuated about 1.58 times more than 1 kHz
  - 750 Hz, as an example, is at about -0.5 dB
    - *This does not mean that 750 Hz experienced gain*
    - -0.5 dB --> 0.89:1; that is, 750 Hz is only attenuated about 89% as much as 1 kHz
- The varying amounts of attenuation distort the shape of the original signal
- Equalization is an attempt to even out attenuation over a range of frequencies (see “with equalization” line)
  - this decreases the distortion suffered by signals within that frequency spectrum

Delay Distortion

- Similar to the fact that attenuation varies according to frequency, so too the propagation rate varies according to frequency
- Again, consider the diagram from the website

- In this case, ~1.7 kHz has what is considered the standard amount of delay.

- Other frequencies propagate slower and arrive later
  - For example, 1 kHz is shown to be about 800 microseconds slower(this is presumably over some specific distance, but the distance is not specified).
- As with attenuation, if left unadjusted, the signal would be distorted as the frequency components forming the signal would arrive out of phase.
  - Here too, equalization evens the delay out over a range of frequencies resulting in the components arriving in phase and being a better representation of the transmitted signal
  - Note that the equalizer cannot speed up frequency components but it can add additional delay to faster frequencies to bring them into phase with the slower frequencies

Noise

- We've already discussed noise as a general limiting factor in terms of the Shannon limit.
  - A key factor to remember – there are many sources of noise within any communication system
    - environmental noise
    - crosstalk
    - quantization noise
    - ...
  - A common mistake is to consider only quantization noise within a system. This is correct when considering the quantization process

(analog to digital conversions), but is not right when considering data transmission through an communication system (as it is only one possible source of noise).

Performance

- When considering performance of a data communications system, we generally want to measure how many bits are moved in a period of time.
  - For the physical layer, the most important performance considerations are:
    - bit rate: how quickly we can lay data onto the channel
    - propagation delay: on form of latency – the time required for the data to travel to the receiver
- The amount of user data moved over a period of time is often referred to as the *effective data rate*.
  - For example, the bit rate within the phone company system for a voice channel is 64 kbps but the effective data rate is 56 kbps since 1 bit/sample is stolen for telco signaling.
- The factors to be considered in calculating the effective data rate depend on the situation – there is no single effective data rate formula.
  - For example, when we consider data link protocols, we will assume the bit rate from the physical layer is the actual bit rate available to the user. We will then calculate how much of that potential bit rate we achieve in light of data link protocol decisions.
  - In contrast, at the physical layer we may consider a raw channel bit rate and calculate the effective data rate by factoring out synchronization bits (such as in asynchronous communications).

Multiplexing

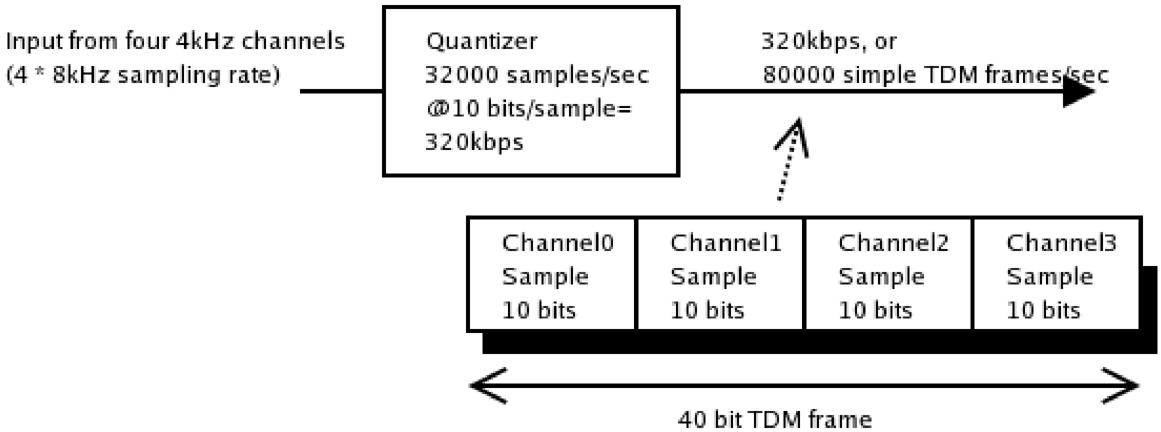
FDM

- Used to place multiple analog signals onto a single channel
- Each signal is modulated to a unique frequency band within the channel spectrum
- The sum bandwidth of the modulated signals must be less than the channel bandwidth
- Telcos used to carry voice traffic in this manner until they upgraded to digital

TDM

- See Forouzon for simple examples. We concentrate on synchronous

TDM over statistical TDM as it is the common case (especially in telcos)
- I think the newest version of Forouzan may even have removed all references to statistical TDM
- See practice question 2 for a more complex example, similar to the question from the tutorial.
- Note that when using a single sampling unit and quantizer to handle multiple signals (as in the practice problem or in tutorial), the data coming out of the quantizer is essentially a time-division multiplexed signal already (though without any synchronization or other bits).



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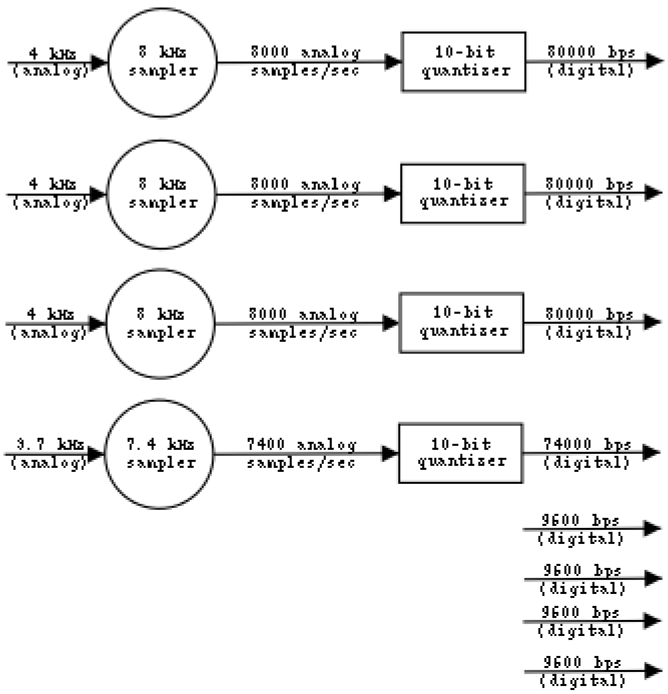
Practice Questions

1. Attenuation tends to be greater at higher frequencies. Accepting this, explain why channel bandwidth appears to decrease over distance.
2. A system has 4 digital sources @ 9.6 kbps, 3 analog sources @ 4 kHz, 1 analog source @ 3.7 kHz are to be TDM multiplexed - show a block diagram for the system specifying data format and bit rates at each point in the system. Note that analog samples are quantized at 10 bits/sample and the Nyquist sampling theorem obviously must be obeyed when sampling.

Answers to Practice Questions

1. Attenuation increases with distance – in fact it is often specified in dB/km. Because higher frequencies suffer greater attenuation, they will a greater amount of amplitude over a fixed distance (d) than a lower frequency. As the distance, d, becomes larger, these higher frequencies are attenuated so much as to disappear from the channel, while the lower frequency still remain.

2. Start by considering how you would build the system if each of the pieces was handled completely independently.



As it stands, we have eight separate signals to be multiplexed. Keep in mind the following when doing TDM:

1. TDM is digital, so all of the signals will need to be digital. We have made a first pass at this in the diagram above – all signals are digital (though the process is not yet optimal).
2. TDM channels must have a fixed data rate - the data rate for all channels is the same.
3. A single signal that exceeds the rate of a single channel may be spread across multiple TDM channels.

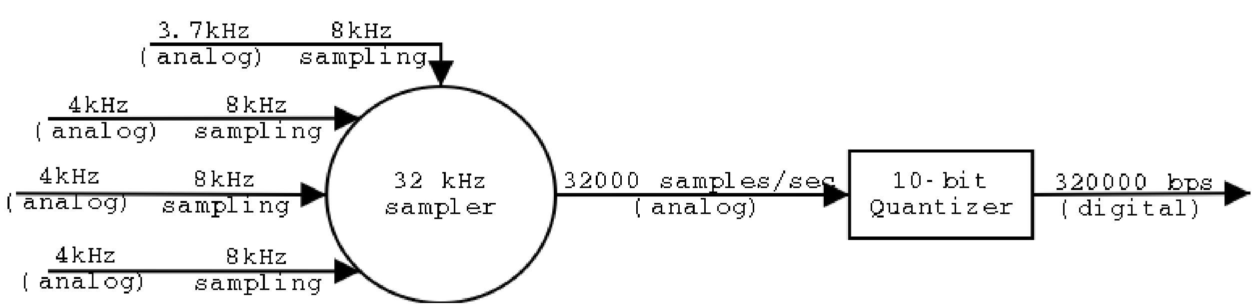
If we were to try and multiplex the above, we would need to first address 2; that is we would need to decide on a single channel data rate that supports all eight signals with minimal waste. A good guess might be 9600 bps (our lowest data rate) - then each of the 9600 bps signals occupies a single channel. The higher data rate signals would require how many channels?

80000 bps / 9600 bps/channel = 8.3 channels which rounds up to 9 channels
74000 bps / 9600 bps/channel = 7.7 channels which rounds up to 8 channels.

Now consider the 'rounding up' part. The signals do not provide enough bits to completely fill the channels – in the first case the 9th channel is 2/3 empty, in the second case 3/10 of the 8th channel is unused. Since TDM requires a fixed data rate this shortage would have to be made up to provide the fixed data rate. This is accomplished by adding extra bits, a process called bit stuffing. The sender stuffs extra bits in to fill the channel and the receiver pulls out the extra bits – obviously both sides must be aware of the bit stuffing.

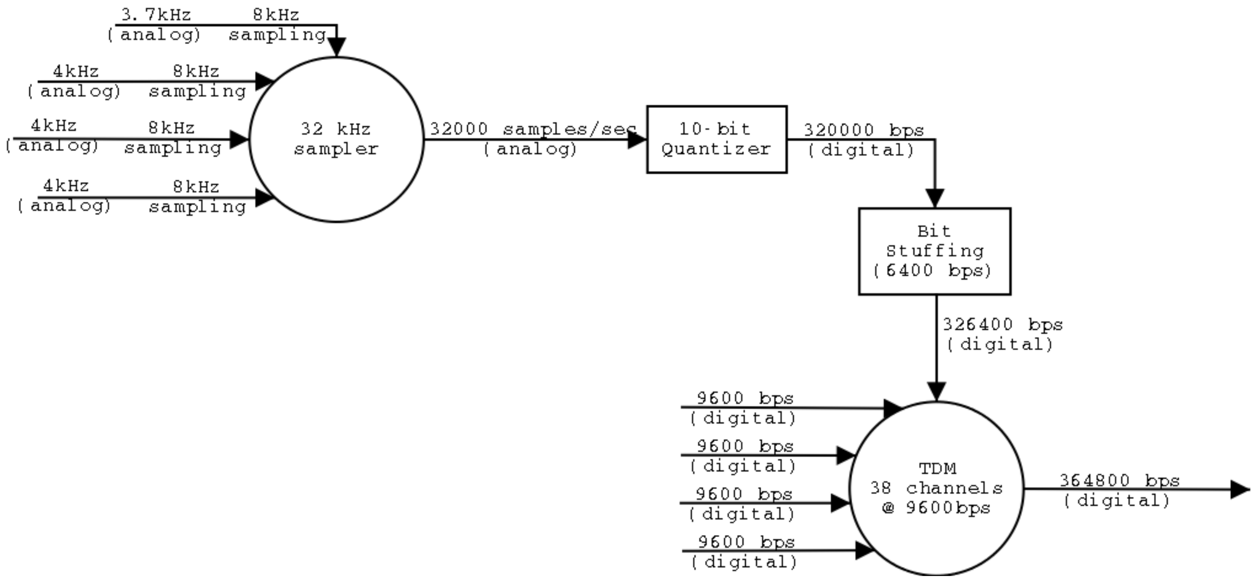
With this in mind, we could complete the above diagram, adding bit stuffing and TDM block(s) but another issue still exists. A quick look at the diagram shows a significant duplication of hardware – there are four sampling units and 4 quantizing units. A better solution will use a single sampling unit and single quantizing unit.

So how do we put together the sampling units? We know the single sampling unit must output *at least the rate of the independent sampling units:* 3 * 8000 + 7400 = 31600 samples/sec. In fact we need to sample slightly faster. Samples must be taken at regular intervals – for this to occur we need to 'normalize' the sampling rate for all signals. In this case, the easiest way is to over-sample the 3.7 kHz signal at 8000 samples per second. Then all four signals are sampled at 8000Hz and the sampler rate is 4 * 8000 samples/second = 32000 samples/second. This single stream of samples can be fed into a single quantizing unit.



If we keep our 9600 bps TDM channel rate, we need 320000bps /

9600 bps/channel = 33.33 channels which rounds up to 34 channels. The rounding up implies we need bit-stuffing. How much? 34 channels * 9600 bps = 326400 bps – 320000 bps = 6400 bps. The resulting system is shown below:



An alternate solution could have the four 9600 bps digital signals multiplexed prior to multiplexing with the (digitized) analog signals (reducing the number of channels). A solution of this nature would is shown below.