

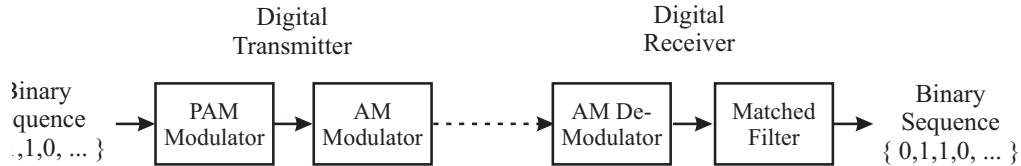
ECE 6390 Homework 5: Noise and Pseudo-Noise

- Noise in a Bent-Pipe Transponder:** A transmitting earth station sends an analog video signal of 6.0 MHz bandwidth up to a satellite at a carrier frequency of 6.2 GHz. The ES antenna is a dish with 55 dBW Effective Isotropic Radiated Power (EIRP). The target satellite is in geosynchronous orbit and employs a bent-pipe transponder to relay the signal back to the earth. The satellite has a 20-dB gain LNA with a noise temperature of 25 Kelvin followed by a mixer which downconverts to 3.7 GHz with a noise temperature of 80 Kelvin. The final stage of the satellite RF chain is a gain-controlled power amplifier capable of transmitting with a 5 dBW average output. The same horn antenna is employed upon transmission and reception; at 6.0 GHz the gain of the dish is 8 dBi and at 3.7 GHz the gain of the dish is 5 dBi. The receiving earth station uses a dish antenna with 35 dBi of gain and a 30-dB gain LNA with a noise temperature of 55 Kelvin. A directional antenna pointing towards the earth sees a physical noise temperature of 190 Kelvin in the microwave bands, while a directional antenna pointing towards space sees a physical noise temperature of 13 Kelvin. Answer the following questions based on this scenario: (10 points)
 - What is the C/N in dB at the *output* of the satellite transmitter amplifier? (2 points)
 - What is the C/N in dB at the receiving earth station? (2 points)
 - What is the weakest (noisiest) link in this satellite system? (2 points)
 - List 4 ways to improve the C/N for the video link between the two earth stations. (2 points)
 - The satellite passes in front of the sun for several minutes, so that the physical noise temperature as seen by the dish antenna is 10,000 Kelvin. What is the C/N in dB at the receiving earth station? (2 points)
- Sampling and Bit Rate:** Perform some web research to get the data specifications for a Compact Disc (CD) audio signal. Write a paragraph that answers the following questions: What is the sampling rate of the audio signal? What is the bit rate? What is the *real* data rate (discounting the redundancy due to the Reed-Solomon coding)? What is the SNR due to quantization noise? Based on this information, how many minutes of music can a 700 MB CD hold? (Note: Don't forget that most audio signals on a CD are stereo – two separate audio signals). (5 points)
- Pulse Shaping:** (10 points – *Terrestrial wireless example taken from an old undergraduate project*)

Your company has been contracted to design a wireless modem for use by laptop computers operating in office buildings or college classrooms. These wireless modems will be placed in

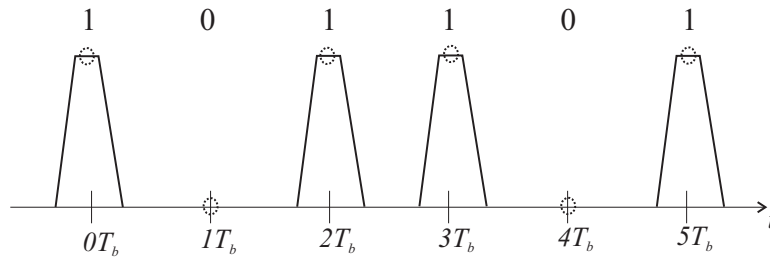
PCMCIA cards that plug into a computer and allow it to communicate with fixed transmitter-receiver access points placed at strategic areas throughout a building. The end result is a *wireless local area network* (WLAN) that allows mobile laptop terminals to access the internet and other information services anywhere within the building. You must provide a scheme for modulating and demodulating a high-data rate signal with one goal in mind: get the most bits-per-second through to the other side.

Below is a block diagram of the system that you are to design. Refer to this figure for the following technical specifications.



The transmitter uses pulse-amplitude modulation (PAM) to represent a *binary signal* of 0's and 1's. This procedure is illustrated below using trapezoidal pulses. You have the freedom to chose the bit rate, $R_b = \frac{1}{T_b}$, and the shape of the pulse, $p(t)$, used in the baseband modulator. There is only one restriction on your pulse – digital PAM chip sets are incapable of generating pulses with more than $4T_b$ of time support.

Baseband PAM of a Digital Signal



Although the binary signal you will modulate is *random*, the resulting signal spectrum will resemble the pulse spectrum $P(f)$. In fact, the *power spectral density* of this baseband signal is given mathematically as

$$G_Y(f) = |P(f)|^2 + \frac{1}{4}|P(0)|^2\delta(f)$$

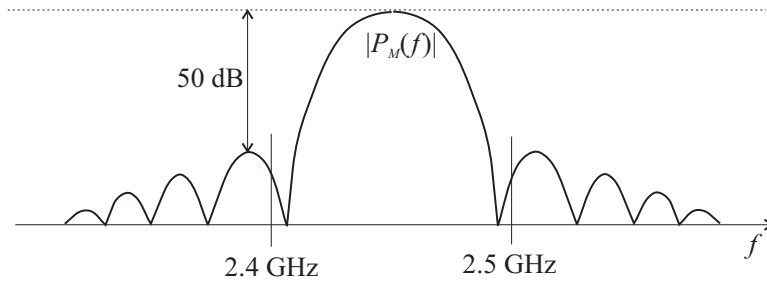
The baseband signal at the transmitter is then *amplitude modulated* to a carrier frequency. The resulting modulated signal must fit within the *Industrial-Scientific-Medical* (ISM) band, an unlicensed wireless frequency band in the range of 2.4-2.5 GHz.

The Federal Communications Commission (FCC) regulates wireless transmissions and requires that transmitters confine most of their radiated energy within the specified band. The specification for your device requires that the power spectrum outside your frequency band be 50 dB *less* than your peak value in band. For a modulated time-domain pulse, $p_M(t)$, this criterion may be written mathematically as

$$20 \log_{10} \frac{|P_M(f)|}{|P_M(f_{\text{peak}})|} < 50 \text{ dB} \quad \text{for all out-of-band } f$$

This is illustrated below:

Design Criterion for Modulated Pulse Spectrum



Find the best possible pulse shape for this system and attach plots of the pulse spectrum that demonstrate the validity of your pulse design. See below for help in taking Fourier Transforms in Matlab.

Notes on Using the FFT Command in Matlab

You will need to calculate and plot the spectrum of pulses in Matlab for this project. Thus, you will need to know how to use the `fft` command. Here is a generic piece of Matlab code you can use:

```
t1 = -10;          % time of first sample
t2 = 10;          % time of last sample
N = 512;          % number of time-domain samples

t = linspace(t1,t2,N);          % create a time-domain axis
f = (-N/2:N/2-1)/(t2-t1);      % create a frequency-domain axis

x=ustep(.5-abs(t)); % create a time-domain sinc function
X=fft(x);           % take the Fast Fourier Transform

subplot(2,1,1); % make a figure with two plots
plot(t,x);      % plot the time domain signal
xlabel('time (s)'); title('Example of an FFT');

XX = abs(fftshift(X))*(t2-t1)/N; % shifts/scales FFT output for plotting
subplot(2,1,2); % switch to second plot
plot(f,XX,'b-'); % use semilogy for logarithmic plot
xlabel('Frequency (Hz)');
```

A few helpful notes:

- The number of samples N in the FFT should be a power of 2. ($N = 2^n$)
- When taking the FFT of a time-domain function of finite-support, be sure to include a large portion of 0-valued samples on either side of the function. This is called *zero-padding* and increases the frequency domain resolution.
- Note that the output of the FFT must be *shifted* using the `fftshift` command. The `abs` function is used to plot just the *magnitude* of the FFT. The output must also be multiplied by $(t_2-t_1)/N$ to get the units right. These operations are only used for *plotting*.

- It is best to choose $t_1 = -t_2$ to keep a symmetric time axis.
- Most properties that work on Fourier transforms still work on FFT's (i.e. time shift, convolution, etc.)
- In the sample code, the `ustep` function is the unit step function – you will need to code your own function or replace it with equivalent code.