ECE4634 –Digital Communications

Design Project 2
Modulation
Oct. 15th, 2020
Due Date: Nov. 3rd, 2020

Objective

The previous project dealt with the concepts of sampling and quantization. This project deals with the transmission of voice signals using carrier modulation. Note that I am not going to give you as much instruction this time. The objective of this project is to again transmit a voice signal in the smallest possible bandwidth with the best possible quality. In this project you will not need to experiment with different sampling rates or quantization schemes. You should use a sampling rate of f_s = 8192Hz with 6 bits (64 levels). Further, you should utilize μ-law companding with μ=255.

Approach

In class we have discussed the complex envelope (or complex baseband) representation for different modulation formats. Specifically, a bandpass signal v(t) can be represented as $v(t)=Re(g(t)e^{j2\pi f_c t})$ where g(t) is the complex baseband signal. It can be shown that we can analyze the performance of a modulation scheme by analyzing the performance in complex baseband. In this project we will take advantage of that fact by using complex baseband version of different modulation schemes to examine the trade-off between energy efficiency and bandwidth efficiency for different modulation schemes.

Specifics

Once again you can find the programs you need on the canvas page in the file `Project2.zip`. The voice signal required for this project is located in the binary file DesignProject1.mat. You should sample the signal at 8192Hz. The functions that I will provide are as follows:

`Analog2Digital(x, f_s, bits, companding, mu, f_s_original)`
– This function converts an 'analog' (i.e., highly oversampled continuous amplitude signal) to a digital signal (i.e., a vector of ones and zeros). *x* is the input vector, *f_s* is the desired sampling rate, *bits* is the number of bits per sample (i.e., quantization levels = 2^{bits}), *companding* is a binary value which specifies whether or not companding is to be used,[1] the value of *mu* must be specified for mu-law companding,[2] *f_s_original* is the original sampling rate of the 'analog' signal[3]

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[1] If companding is not 0 (i.e., any value other than zero) it is interpreted as true.
[2] If companding == 0, mu need not be specified.

[3] It is assumed to be 65536 unless specified. Thus, *f_s_original* is also an optional input variable.

`PhaseMod(x, k, GrayCoding)` – This function uses an input vector of data bits (x) of length N to create an output vector of symbols of length N/k where $k=\log_2(M)$ is the number of bits per symbol and M is the number of possible symbols. The symbols are complex baseband representations of M-ary phase modulated signals. Note that GrayCoding is an optional binary input. If it is not used, the function does not use Gray Coding. If it is specified and is not equal to 0, Gray Coding is used.

`QAM16_Mod(x)` - This function uses an input vector of data bits (x) of length N to create an output vector of symbols of length $N/4$. The symbols are complex baseband representations of *16QAM* modulated signals.

`AddNoise(x, SNRperBit, BitsPerSymbol)` – This function adds noise to the vector x. The output SNR per bit of this vector is determined by the input variable *SNRperBit*. Note that SNRperBit is *linear* not in dB.

`PhaseDemod(x, k, GrayCoding)` – This function creates a vector of bits of length $N*k$ (where $k=\log_2(M)$ is the number of bits per symbol) from a length N vector of phase modulated symbols. The assumed modulation is M-PSK where $M=2^k$. Note that GrayCoding is an optional binary input. If it is not used, the function does not use Gray Coding. If it is specified and is not equal to 0, Gray Coding is used.

`QAM16_Demod(x)` – This function creates a vector of bits of length $4N$ from a length N vector of QAM modulated symbols.

`Digital2Analog( x, NumBits, companding, mu)`- This function creates a numerical valued signal from a string of bits assuming that *NumBits* per quantization level are used. Further, it is assumed that the levels are evenly distributed about zero. *Companding* and *mu* are the same as in `analog2digital`.

In this project you are to analyze 5 modulation schemes (BPSK, QPSK, 8-PSK, 16-PSK, 16-QAM) for bandwidth efficiency and energy efficiency. This analysis will be based on the use of the modulation schemes to transmit a voice PCM signal over an AWGN channel with an SNR per bit of 7.5dB. You will accomplish this by first converting the given voice signal to a digital bit stream using the function given. (Note that the given function simply encapsulates what you did in the last project with the added step of converting the quantized signal to bits. Second you must modulate the signal using the two modulation functions provided (note that PhaseMod is to be used for all PSK techniques). After modulation you will pass the signal through an AWGN channel by using the AddNoise function. After adding noise to the signal, you will demodulate the noisy signals using the demodulate functions. As mentioned above, your goal is to minimize the bandwidth of the transmitted signal while obtaining acceptable voice quality. Assume that the cost in this case will be related to the bandwidth. Specifically, the cost is $0.01/minute/kHz and that raised cosine pulses are used with a roll-off factor of r = 0.25. For the PSK schemes, examine the performance with and without Gray Coding. How do they compare? Why?

Report

Your project report is the means for communicating to me what you know about the principles that you have learned from the project. Note that unlike the first project, I expect more independent thinking on your part in this project. Things for you to include:

Voice quality of each modulation scheme
Bit error rate for each of the modulation schemes
Signal constellation plots for the modulation schemes (with and without noise)

Hint: to plot the constellation diagram from the complex baseband signal g(t), use plot(real(g), imag(g)

Bandwidth of each of the modulation schemes (theoretical values are sufficient)

Note the amount of noise added by quantization versus the amount of noise added by the channel. (By noise I mean the error/distortion in the *final signal after D-to-A conversion).*

Time plots of the voice signal

One last note: Communication skills are important in any career. Engineering is no different. Use this project to refine your report writing skills. In my experience engineers who cannot communicate their ideas are less likely to get promoted and are not taken as seriously. Part of our job as professors is to help prepare you for either engineering careers or graduate school. In either case you will benefit from an ability to communicate.