Introduction to Modern Communications

361-1-3221

Computer Assignment 1

Spring 2022

General Instructions

- Electronic submission via the course website by **10.05.2022, 23:59**
  - All figures must be labeled correctly on both the x and y axes.
  - In case a figure includes more than one plot, a legend should be added. Use markers to facilitate readability in the case of black and white printing.
- The assignment has three questions: In Question 1 you will simulate an analog communication system. This question is intended to be carried out using MATLAB. Questions 2 and 3 are analytical questions and need to be done **without** the aid of a computer.
  - Analytic expressions should be simplified and explicit as much as possible.
- All solutions and graphs should be submitted as a PDF file. MATLAB code should be attached to the end of the file.
- **Submission should be done in pairs.**
- **Only one person should upload the assignment to the moodle, and the pdf file should contain the names and IDs of both students.**

1 AM and FM Modulation- MATLAB (50%)

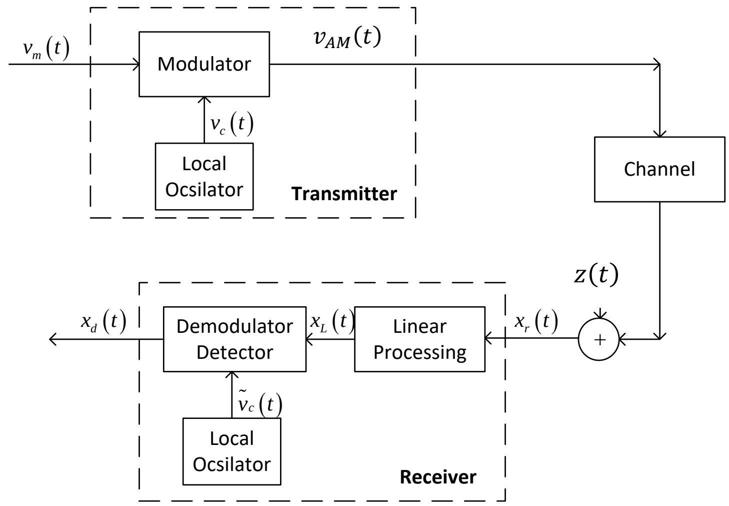

Figure 1-Analog Communication System

1.1 Data Generation (5%)

To generate the information signal $v_m(t)$, consider the following steps:

- Download the wav file "in-the-air.wav" from Moodle.
- Load the file in MATLAB using the command:

  "[v_m,fs] = audioread('in-the-air.wav')"

  The MATLAB vector 'v_m' is of dimension 107968×2, where the two columns of the vector are referred to the right and left channel. The parameter 'f_s' is the sample rate for the data signal 'v_m'.
- To make things simpler, we will work with a one dimensional signal (107968×1), i.e., extract only the left channel by using the command: "v_m = v_m(: ,1)".
- You can listen to the information signal by using the command: "sound(v_m,fs)".
- Define the sampling interval $T_s = 1/f_s$, and define the number of sample points N as the length of the vector v_m (i.e., use the MATLAB commend "N=length(v_m)").
- create the time vector "t", and the frequency vector "f", using the commands:

  "t = 0: Ts: (N-1)*Ts"

  "f = linspace(-fs/2,fs/2,N)"

  From now on, when we refer to the time t and the frequency f, we will refer to the discrete-time vectors created above.

- Finally, to evaluate the Fourier transform of $v_m(t)$, $V_m(f) = \mathcal{F}\{v_m(t)\}$, use the "fft" command: "V_m = fftshift(fft(v_m)) / sqrt(N)"

1.1.1. Plot on the same figure (using "subplot") the data signal $v_m(t)$ and its Fourier transform $V_m(f)$.

1.1.2. Evaluate the bandwidth of the information signal.

1.2 Modulator (10%)

In this section, we will simulate the AM with Carrier modulation learned in class, for the information signal $v_m(t)$. The modulated signal will be denoted by $v_{AM}(t)$.

Choose the carrier frequency $f_c = 15 \cdot 10^3 [Hz]$ and the modulation index $k_{AM} = 0.02$.

1.2.1. Create the signal $v_{AM}(t)$. You can use the MATLAB function "ammod".

1.2.2. Compute in MATLAB the Fourier transform of $v_{AM}(t)$, denoted $V_{AM}(f)$, by using the "fft" command (as explained above), and **plot $V_{AM}(f)$. Explain.**

1.2.3. Listen to the modulated signal $v_{AM}(t)$ (using the "sound" command). Do you hear something? **Explain?**

1.3 Channel (10%)

In this section, we will generate the noise sequence $z(t)$.

In order to simulate the continuous-time noise in MATLAB, you will generate a discrete time sequence $z(t)$ (again, remember that t is a discrete-time vector). The sequence is of independent and identically distributed random variables, each drawn from a zero-mean normal distribution with variance $N_0/2$, i.e., $z(t) \sim N(0, N_0/2), \forall t$. $z(t)$ is further generated independently of the data signal.

1.3.1. Generate in MATLAB a real white Gaussian noise sequence $z(t)$ as described above.

- MATLAB instructions: To generate the noise in MATLAB, use the function "randn" in the following command: $z = \sqrt{\frac{N_0}{2}} * \text{randn}(1, N)$.

  Make sure that the noise vector has the same size as the signal vector.

  Choose $\sqrt{N_0/2} = 0.02$.

1.3.2. Generate the received signal at the channel output: $x_r(t) = v_{AM}(t) + z(t)$.

Compute and **plot** in MATLAB the Fourier transform of $x_r(t)$, $X_r(f)$. **Explain**.

1.4 Demodulator (15%)

In this section, you will demodulate the channel output obtained in the previous section. We first filter the received noisy signal with a bandpass filter.

1.4.1. Generate the signal $x_L(t)$ by passing the received signal $x_r(t)$ through a bandpass filter. You can use the MATLAB function "bandpass". Note that you need to choose the passband frequency range of the filter appropriately.

Compute and **plot** in MATLAB the Fourier transform of $x_L(t)$, $X_L(f)$. **Explain.**

1.4.2. Demodulate the received signal $x_L(t)$. You can use the MATLAB function "amdemod". Let $x_d(t)$ denote the decoded signal at the receiver. **Plot** the decoded signal, together with $v_m(t)$ on the same graph.

Plot the decoded Fourier transform of the signals $X_d(f)$ and $V_m(f)$ on the same graph.

NOTE: You may need to lowpass filter the received signal $x_d(t)$. You can use the MATLAB function "lowpass" if needed.

1.4.3. Listen to the signal $x_d(t)$. How does it sound?

1.4.4. Finally, use the MATLAB function "xcorr" to determine the correlation between the original signal $v_m(t)$ and the decoded signal $x_d(t)$. You can use the following command: **"**xcorr(x_d, v_m, 0, 'coeff')**". Write the result.**

1.5. Now with different noise and modulations (10%)

1.5.1. Repeat items 1.3 and 1.4, this time choosing $\sqrt{N_0/2} = 0.1$.

1.5.2. Repeat items 1.3-1.5.1, this time for FM modulation. Choose the frequency deviation $\Delta f_d = 10 \cdot 10^3 [Hz]$. You can use the MATLAB functions "fmmod" and "fmdemod".

1.5.3. Conclude by writing the four correlation coefficients from item 1.4.4. Explain the differences.

2 PM Modulation (25%)

Consider the information signal (a periodic signal with period $T_m = 1/f_m$):

$$v_m(t) = \begin{cases} 1, & |t| \le \frac{T_m}{4} \\ 0, & \frac{T_m}{4} < |t| < \frac{T_m}{2} \end{cases}$$

Assume that $v_M(t)$ is a PM modulation of the signal $v_m(t)$, with maximal phase deviation $\Delta\varphi_{\max}$.

2.1 (10%) Show that the low pass equivalent of $v_M(t)$ is given by:

$$v_{LPE}(t) = \sum_{n=-\infty}^{\infty} C_n \cdot e^{j2\pi f_m n \cdot t},$$

and find the coefficients $\{C_n\}_{n=-\infty}^{\infty}$ explicitly.

2.2 (5%) Show that:

$$|C_n| = \begin{cases} \left|\frac{1}{2}\cos\left(\frac{\Delta\varphi_{\max}}{2}\right)\right|, & n = 0 \\ \left|\frac{1}{\pi n}\sin\left(\frac{\Delta\varphi_{\max}}{2}\right)\right|, & n \text{ odd} \\ 0, & n \text{ even}, \ n \ne 0 \end{cases}$$

2.3 (10%) We define the cut-off index: $n_c(\varepsilon) \triangleq min\{n: |C_n| \le \varepsilon\}$.

Find $n_c(\varepsilon)$ (as a function of ε) for:

2.3.1 $\Delta\varphi_{\max} = \frac{\pi}{2}$

2.3.2 $\Delta\varphi_{\max} << 1$ (you can use the approximation $\sin(\alpha) \approx \alpha$ for $\alpha << 1$).

3 AM and FM Modulations (25%)

Consider the information signal $v_m(t)=\cos(2\pi f_m t)$.

The signal $v_m(t)$ is modulated to the signal $z(t)$ using amplitude modulation, around a carrier frequency $\overline{f}=2f_m$, where the carrier signal is $v_c(t)=\cos(2\pi\overline{f}\cdot t)$.

Then, the signal $z(t)$ is modulated to the signal $x(t)$ using frequency modulation, around a carrier frequency $f_c >> f_m$:

$$x(t)=\cos\left(2\pi f_c t+2\pi\Delta f_d\int_{\tau=-\infty}^{t} z(\tau)\,d\tau\right)$$

where Δf_d is the maximum deviation frequency.

3.1 (10%) Assume that $z(t)$ is an **SSB modulation** of $v_m(t)$.

Find the bandwidth of the signal $x(t)$ by using Carson's rule. Your answer should depend on the parameters f_m, Δf_d. Consider the two methods of SSB modulation: USSB and LSSB (i.e., you should provide two different answers according to each method).

3.2 (15%) Assume that $z(t)$ is a **DSB modulation** of $v_m(t)$.

Derive an expression for the spectrum of the transmitted signal, $X(f)$. Your answer should depend on the parameters f_m, Δf_d and the Bessel function, defined by:

$$J_n(\beta)\triangleq\frac{1}{2\pi}\int_{\alpha=-\pi}^{\pi} e^{j(\beta\sin(\alpha)-n\cdot\alpha)}\,d\alpha$$