ANALOG MODULATION

In this project, we will learn how to emulate analog AM and FM modulation/demodulation by discrete-time signal processing in MATLAB using a sampled speech signal.

The sampled speech signal we will work with is stored in the file `speech_dft_8kHz.wav`. The sampling rate is 8 kHz. You can read this audio file via the MATLAB command
`x = audioread('speech_dft_8kHz.wav');`
Then, you can see the samples of the speech signal in the variable `x` in your MATLAB workspace. You can listen to it with the command `soundsc(x, 8000)`.

Since you need to execute pointwise multiplication or addition to implement AM and FM respectively, you have to use a sampling rate that is at least twice the carrier frequency. This necessitates upsampling the sampled speech signal to higher sampling rates before modulating it. After demodulation, you need to downsample the recovered message signal back to 8 kHz to be able to listen to it. You can use `interp` and `decimate` functions of MATLAB for upsampling and downsampling, respectively.

PLEASE NOTE THAT YOU ARE NOT ALLOWED TO USE MODULATION FUNCTIONS OF MATLAB.

Problem 1 (*AM Modulation*)

1. Modulate the speech signal $m(t)$ with carrier through the following steps:

   (a) Upsample the speech signal. What is your new sampling frequency? How did you choose that specific sampling rate? What is the total number of time samples? Explain your answers.

   (b) Modulate the speech signal with the carrier frequency 250 kHz according to:
   $$y(t) = (A + m(t))\cos(2\pi f_c t)$$
   where $m(t)$ is the message, f_c is the carrier frequency and A is a constant. Choose A such that the modulation index is 0.6. Explain how you determine A in your code.

   (c) Plot the Fourier magnitude spectrum of $m(t)$ and $y(t)$, in dB, on separate figures. Don't forget to label axes of the plot. Explain what you see.

2. Demodulate $y(t)$ using the envelope detection technique through the steps:

   (a) First, rectify the signal (pass it through the ideal diode) and plot its spectrum. How do you emulate a diode in your code?

   (b) Then, pass it through the RC circuit. The RC circuit is a low pass filter with the impulse response
   $$h(t) = e^{-\frac{t}{\tau}}u(t)$$
   where $\tau = RC$ is the time constant. Set $\tau = 5 \times 10^{-5}$ s and plot the magnitude spectrum of both the filter and filtered signal. Also, plot the demodulated signal and the original message $m(t)$ in time domain. Now set $\tau = 5 \times 10^{-4}$, is there any difference in the demodulated signal? Why? Comment on the effect of τ on the performance of the envelope detector.

   (c) Now, downsample the filtered signal to its original rate, 8 kHz. Plot the original and the demodulated signal in both time and frequency domain. Also, listen to the output. Did you recover the speech back? Is there any difference?

3. Add 20 dB white, Gaussian noise to $y(t)$ and repeat part 2.

Problem 2 (*FM Modulation*)

1. Modulate the speech signal $m(t)$ through the following steps:

   (a) Upsample the speech signal. What is your new sampling frequency? How did you choose that specific sampling rate? What is the total number of time samples? Explain your answers.

   (b) Modulate the speech signal with $f_c = 450$ kHz and $k_f = 300000\pi$ rad/V·s according to

   $$y(t) = \cos\left(2\pi f_c t + k_f \int_{-\infty}^{t} m(t)dt\right)$$

   You can approximate the integral using the MATLAB function `cumsum`. You have to divide the result of `cumsum` by f_s. Why?

   (c) Plot the Fourier magnitude of the modulated signal. What can you say about the bandwidth of the signal compared to AM modulation?

   (d) Estimate the bandwidth of the signal with Carson's rule. Is your estimate close to what you see in the spectrum?

2. Demodulate the FM signal using the superheterodyne FM receiver. Remember that for this purpose you need a differentiator followed by an envelope detector.

   (a) Approximate the differentiator using the MATLAB function `diff`. Do not forget to multiply the result by f_s. Why?

   (b) For the envelope detector, take the time constant $\tau = 10^{-3}$.

   (c) Compare the spectrum of $m(t)$ with the demodulated signal (before downsampling). Additionally, plot $m(t)$ and demodulated signal in the time domain. Is there any difference in the plots? If there is, why?

   (d) Now downsample the demodulated signal to 8 kHz. Did you recover the speech back?

Question (*Superhet Receiver*)

Explain why you did not have to use the tuner and the intermediate frequency (IF) stages of the superhet receiver in this project.