Super-heterodyne Receiver

I. Introduction

The purpose of this project is to simulate the basic components of an analog communication system using MATLAB programming. Specifically, an AM modulator and a corresponding super-heterodyne receiver will be simulated using radio-station generated signals.

Figure 1 shows a typical AM transceiver. The input message is modulated and wirelessly transmitted. The receiver detects the message using multiple stages of mixing and filtering as we discussed in the lectures.

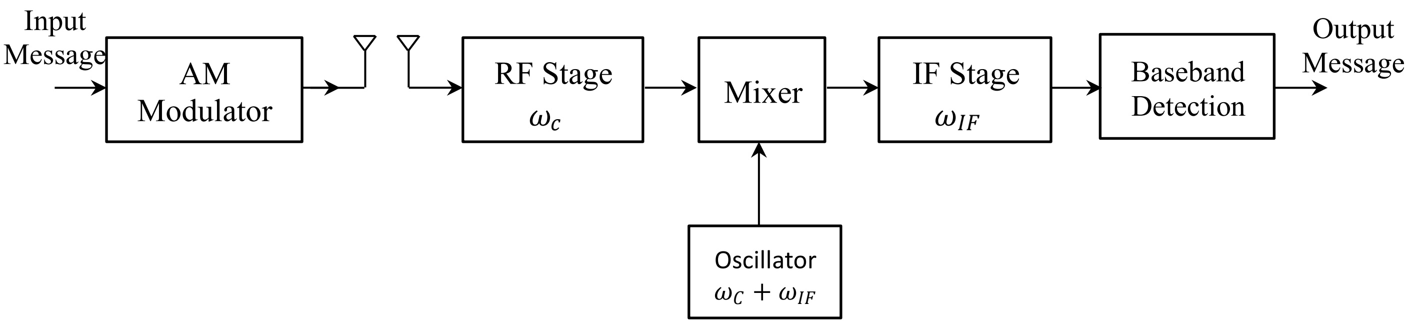

Figure 1: AM modulator and a super-heterodyne receiver

II. Project Statement

Given audio signals that represent the input message for the above system. The task is to design the block diagram in Figure 1 using MATLAB. The details are given as follows:

i. *The signals*: stereo (2-channel) signals are provided in the baseband. The signals are stored in files with extension ‘wav’. We can read these files in MATLAB using the command ‘audioread’ or ‘wavread’. For later processing, we will need to obtain the sampling frequency (which was used to digitize the recorded waves). The MATLAB function can return the sampling frequency.
Each modulating signal (the messages) has two channels representing a stereo signal. We are going to implement a monophonic receiver so there is no need for two separate channels. We can form a single channel stream by simply adding the two channels. Please note that the given signals are not of equal length. We can pad the short signals with zeros, so they have all equal length.

We can plot the spectrum of the signal using the 'fft' and 'plot' commands. We will have to adjust the axis scale, so the signal spectrum is plotted versus frequency. By doing so, you can estimate the bandwidth of the signal. You will use this bandwidth value when you design the filters.

ii. *The AM modulator*: the modulation type is DSB-SC for all the signals. The first signal is modulated with 100 KHz carrier. Each following signal is modulated with a carrier of frequency $\omega_n = 100 + n\Delta F$, where $\boldsymbol{\Delta F = 55}$ **KHz** and the index n is the signal index ($n = 0$ for the first signal that is modulated at 100 KHz). The modulated signals are used to construct an FDM signal (FDM: frequency division multiplexing).

Note that the values of the carrier frequencies are smaller than the actual AM/FM transmission values that are in the order of megahertz. The small values of the carrier frequencies used in this project were chosen to reduce the simulation complexity. We may still face some problems due to the audio file length; if this happens We may take half or quarter the size of the file.

Another important simulation parameter is the sampling frequency F_s as mentioned before. We should obtain F_s using the 'wavread' command (or other). Now, if we want to modulate a signal, generate the carrier: $\cos(\omega_n t)$ which, in the discrete domain (in MATLAB environment), is generated as $\cos(\omega_n n T_S)$ where T_S is the sampling interval. This could simply be $T_S = 1/F_s$. However, if the carrier frequency $F_n = \frac{\omega_n}{2\pi}$ turns out to be greater than $F_S/2$ then we cannot use the given F_s as this will violate the Nyquist criteria.

One solution is to increase the sampling frequency F_S, say by 10 times. This is effectively done by increasing the number of samples of the modulating signals. That is, if the modulating signals are of size Nwhich is obtained using a sampling frequency F_s, then increasing the number of samples to $10N$ is equivalent to increasing the sampling frequency to $10F_S$ (or reducing the sampling interval to $\frac{T_S}{10}$). To increase the number of samples of the modulating signals We can use 'interp' command in MATLAB which interpolates the signal to obtain more samples.

iii. *The wireless channel*: in real life the modulated FDM signal is transmitted over the air from the base station. Simulating this wireless channel is out of the scope of this project. So, we are not going to do anything for this part. We can go immediately to the receiver.

iv. *The RF stage*: This is the stage that performs interference-image rejection. For simplicity, this stage will be implemented as a band-pass Filter (BPF) only, centered at the carrier frequency ω_n (that is tunable at the desired station).

v. *The Oscillator*: this is the generator for a carrier frequency $\omega_c + \omega_{IF}$, where $\omega_c = \omega_n + n\Delta F$. The IF frequency $\boldsymbol{\omega_{IF} = 27.5}$ **KHz**. The mixer is a simple multiplier in this simulation.

vi. *The IF stage*: similar to the RF stage, this stage is simply modeled as a band-pass filter only, centered at the center frequency ω_{IF}.

vii. *Baseband detection*: this is the final stage where the signal is obtained in the baseband. This stage thus involves mixing with a carrier of suitable frequency (it is easy to guess its value) and filtering the signal using a low-pass filter (LPF). The design of LPF can be done similarly to the BPF.