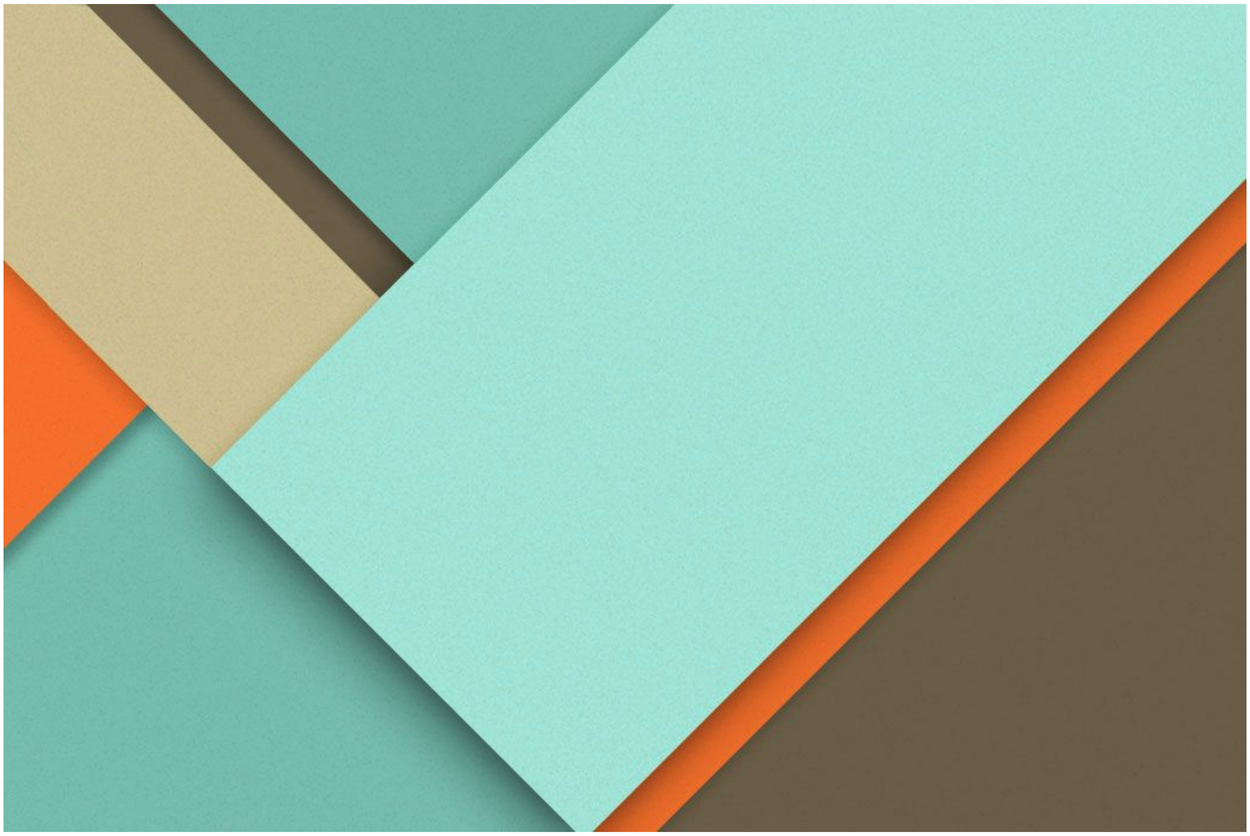

Communication mini project

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Overview

The code is separated in 2 Matlab files, the first one is for amplitude modulation(**AM**) and the second one is the frequency modulation (**FM**), Code in the files has the same concept, first reading the audio file and resample it with the carrier sampling rate then calculate the modulated signal from the equation discussed at lectures after that looping 20 times and changing **SNR** that added to the signal as a noise then calculate the demodulated signal and **MSE** and at the end of the loop plotting **SNR** vs **MSE.**

Choosing the sampling rate

According to the **Nyquist sampling theorem** the sampling rate must be more than twice the frequency of the carrier signal so we choose the sampling rate 3 times carrier frequency.

Setting the modulation index

At **AM** the *modulation index* $= m_p / A_c$, so it helps us to get the DC component A_c.

Setting the deviation ratio

At **FM** the *deviation ratio = delta f / B* and *delta f* $= k_f * m_p / 2\pi$ so it helps us to get k_f.

SNR increases in both cases

Increasing in **SNR** in both cases makes the sound more clear as **SNR** increases.

Small values of β

As $\boldsymbol{\beta}$ decreases the sound becomes more worse.

MSE vs SNR

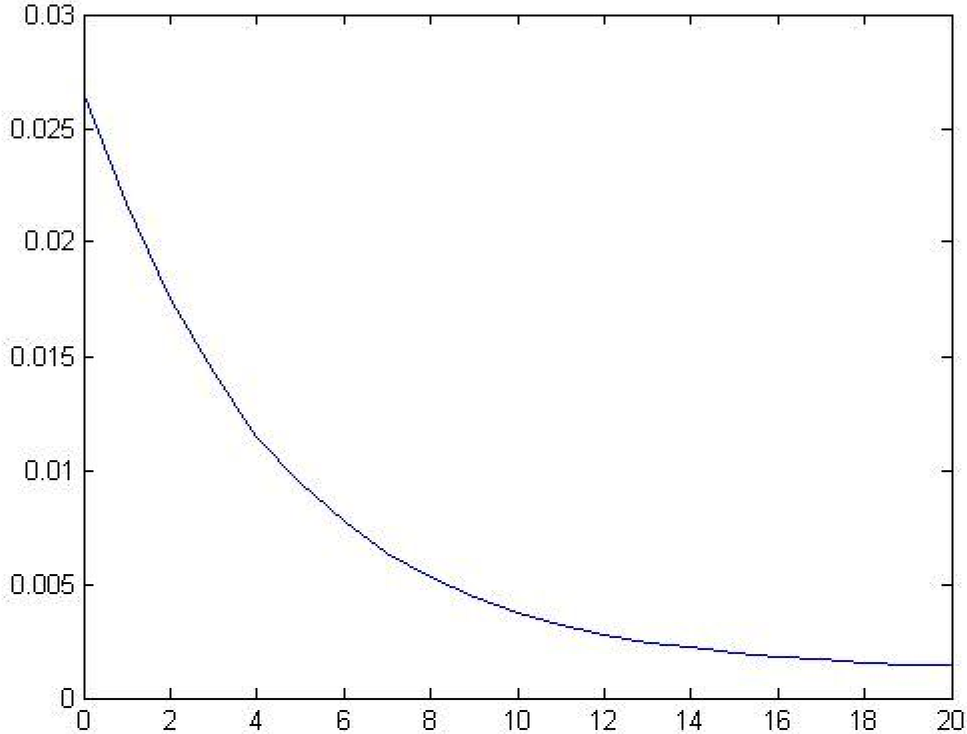

Fig 1**: **MSE** vs **SNR** at **AM

As **SNR** increases the **MSE** decreases because of the good change in sound quality.

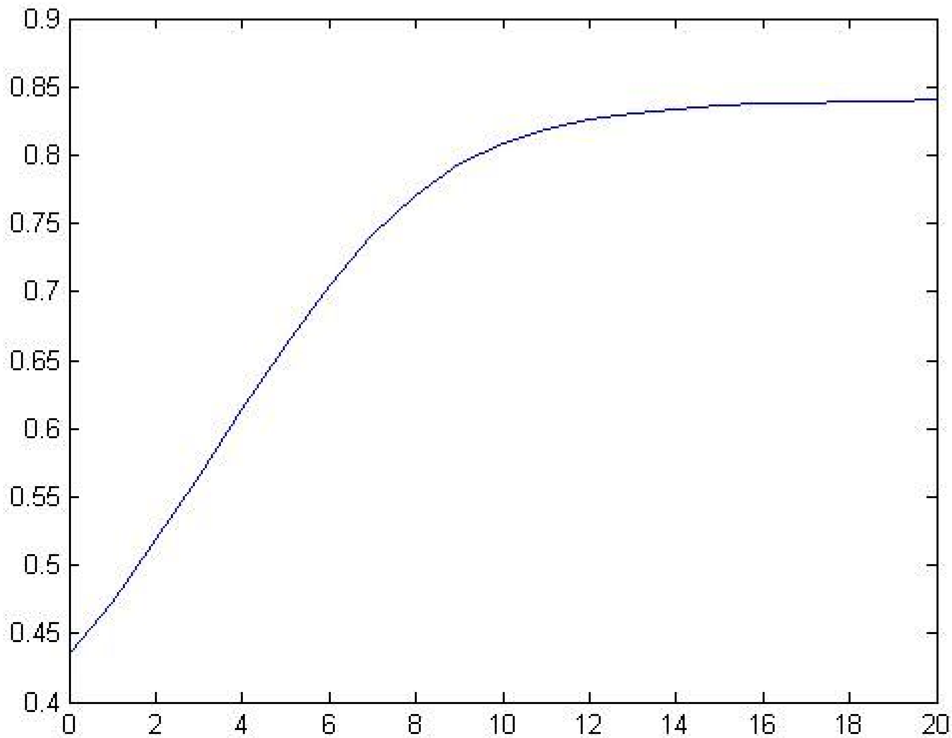

Fig 2**: **MSE** vs **SNR** at **FM

Although the increasing in the quality of the sound the **MSE** increases as unexpected.

Code

AM file

```
% read from audio file
[audio, sample_rate] = audioread('speech_dft_8khz.wav');
% make sure that the audio is 1D vector
audio = audio(:,1);
% DC component
Ac = abs(min(audio) / 0.9);
% carrier frequency and sample rate
carr_freq = 100000;
carr_sample_rate = carr_freq* 3;
% resample the audio to the carrier sample rate
audio_resample = resample(audio, carr_sample_rate, sample_rate);
% carrier omega
Wc = 2 * pi * carr_freq;
% time vector
size_of_audio = size(audio_resample);
t = (1:size_of_audio(1));
t = transpose(t);
% modulated signal
St = (Ac + audio_resample(t)) .* cos(Wc * t);
% loop in SNR from 0 to 20 dB
MSE = (0:20);
for SNR = 0:20
   % calculate the signal with noise
   audio_noise = awgn(St, SNR);
   % calculate the demodulated signal
   demod = abs(hilbert(audio_noise)) -
mean(abs(hilbert(audio_noise)));
   % resample it to the base sample rate
   demod = resample(demod, sample_rate, carr_sample_rate);
   new_MSE = mean((audio - demod).^2);
   MSE(SNR+1) = new_MSE;
   % listen to the signal
   sound(demod);
```

```
   pause(5);
end
plot((0:20), MSE);
```

FM file

```
% read from audio file
[audio, sample_rate] = audioread('speech_dft_8khz.wav');
% make sure that the audio is 1D vector
audio = audio(:,1);
% DC component
Ac = abs(min(audio) / 0.9);
% carrier frequency and sample rate
carr_freq = 100000;
carr_sample_rate = carr_freq* 3;
% resample the audio to the carrier sample rate
audio_resample = resample(audio, carr_sample_rate, sample_rate);
% carrier omega
Wc = 2 * pi * carr_freq;
% time vector
size_of_audio = size(audio_resample);
t = (1:size_of_audio(1));
t = transpose(t);
% modulated signal
St = (Ac + audio_resample(t)) .* cos(Wc * t);
% loop in SNR from 0 to 20 dB
MSE = (0:20);
for SNR = 0:20
   % calculate the signal with noise
   audio_noise = awgn(St, SNR);
   % calculate the demodulated signal
   demod = abs(hilbert(audio_noise)) -
mean(abs(hilbert(audio_noise)));
   % resample it to the base sample rate
   demod = resample(demod, sample_rate, carr_sample_rate);
   new_MSE = mean((audio - demod).^2);
   MSE(SNR+1) = new_MSE;
   % listen to the signal
```

```
   sound(demod);
   pause(5);
end
plot((0:20), MSE);
```