

AM and FM

Communication Project

Prepared By:	Sec:	BN:
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-Evram Youssef	1	9
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-Mahmoud Othman	2	21
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Delivered to: TA\ Alaa Abd El-Nasser

Under supervision of: Dr\ Mikel Melek

How to run:

First Installing Dependencies:

First install Python 3.7, with pip

Then run this line (windows or linux)

```
$ python3 -m pip install --user -r requirements.txt
```

Then install Octave, with communications package (in case of windows).

To install them on Ubuntu/debian linux:

```
$ sudo apt update
```

```
$ sudo apt install -y octave octave-communications
```

AM:

```
$ python3 am.py
```

FM:

```
$ octave fm.m
```

Show Plot:

```
$ python3 plot-mse.py
```

AM Code:

0- Reading the audio sample.

1- Arranging the audio file into sample rates.

2- Dividing the peek by the modulation index (0.9) to get A_c .

3- Adding A_c to the audio signal, and multiply it by $\cos(Wct)$.

4- Adding normal noise to the modulated signal, with different SNR [0,1, 10, 20].

5- Demodulating the signal using envelope detector, by "Hilbert" library.

6- Writing the output sample.wav audio file, with different SNR's.

FM Code:

1- Read sample.wav

2- Resample to $4 \cdot f_c$

3- Let max frequency $B = \text{sample rate}$

4- $\text{freqdev} = \beta * B$

5- Calculate Kf with $kf = \text{freqdev} * 2\pi / mp$

6- Modulate with $\text{modulated}(t) = A_c * \cos(2\pi * f_c * t + kf * \text{integration}(\text{audio}(t)))$

7- for each β (5, .1):

a- for each SNR (0, 1, 10, 20):

a- Add random noise relative to $1/\text{SNR}$

b- Demodulate with *fmdemod*

c- Resample back to original sample rate

d- Write to /out

Choosing The Sampling Rate:

AM:

The library used for reading the audio sample, returned a specific sampling rate, and it was audible, so no need for resampling.

FM:

'Audioread' function returns a sampling rate from the file's metadata, and another resampling ratio was chosen for better analysis which was approximate 4.

How to set Modulation Index to 0.9:

We calculated the absolute of the minimum peak of the audio file, A_m , and then set $A_c = A_m / 0.9$

How to set the Deviation Ratio to 5 in FM (WBFM):

We multiplied the Sample Rate (F_m) by the Modulation index (B), and set the frequency deviation to that.

Observation on SNR:

As SNR increases the Noise's effect decreases, this could be observed via the Plots, or simply by listening to different audio file in 'out' folder.

For Small values of B (0.1):

As observed from the Noise plot, for small values of SNR, small *modulation index* (deviation ratio) creates a very low quality audio, but as SNR increases, (after 10), small *modulation index* gets better than both larger B and AM.

GRAPH:

Comment:

AM results are worse than FM in general due to addition of noise.

NBFM is better than WBFM when SNR > 10, otherwise WBFM is better.

