Dept. of Electronics and Electrical Communication Engineering
Indian Institute of Technology Kharagpur

ANALOG COMMUNICATION THEORY (EC31001)

ASSIGNMENT-2 ANGLE MODULATION

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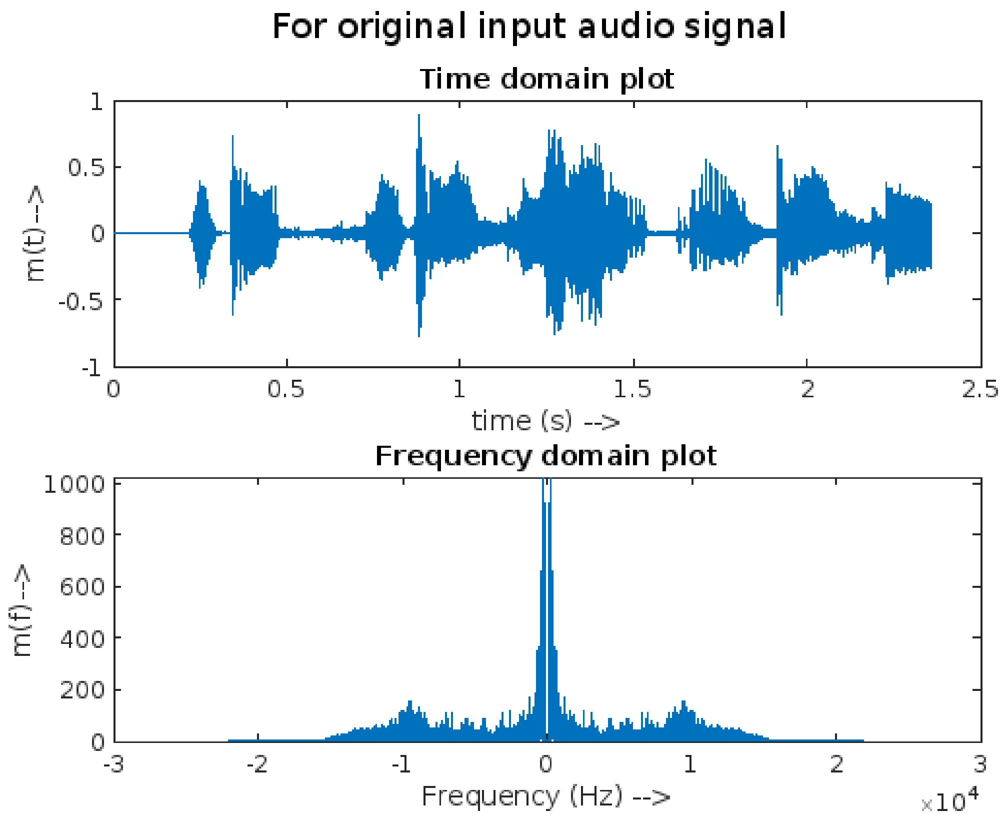
The input audio signal in time and frequency domain can be represented as:



For frequency spectra we use the inbuilt function ***fft*** and shift it to center by ***fftshift*** and plot the absolute value using ***abs***.

Modulation:

We represent FM signal as:

$$x_{FM}(t) = Ac * \cos(2 * pi * f_c * t + k_f * \int_{-\infty}^{\tau} m(\tau)d\tau\ )$$

We represent PM signal as:

$$x_{PM}(t) = Ac * \cos(2 * pi * f_c * t + k_p * m(\tau))$$

Where m(t) is the input audio signal (with sampling frequency 44100Hz) and c(t) is the carrier signal (with frequency 50kHz).

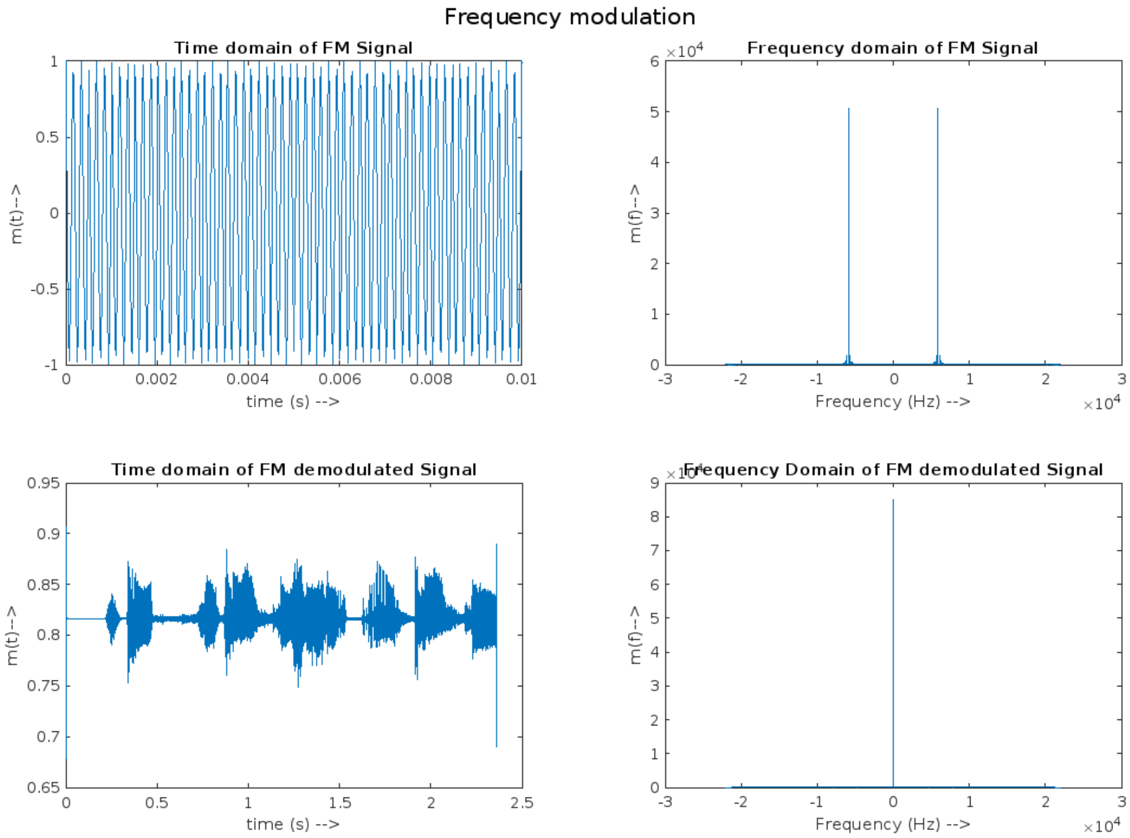
Demodulation:

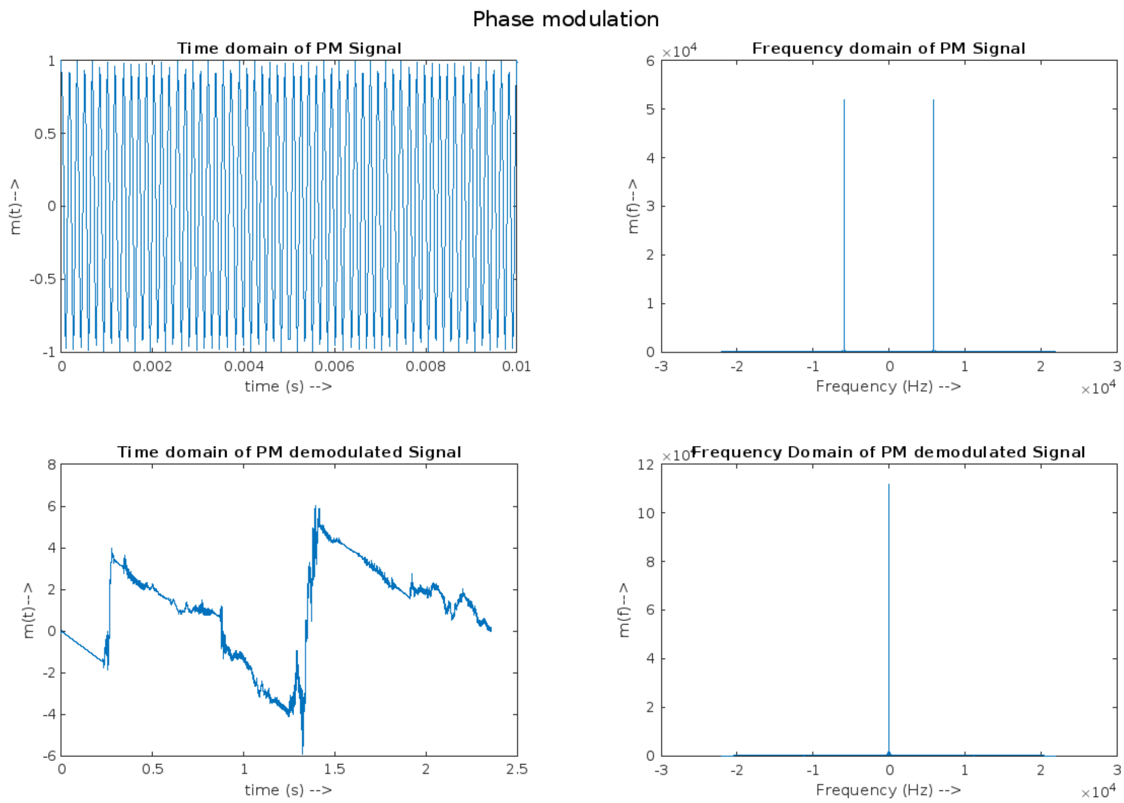
For frequency demodulation we pass it through differentiator and then envelope detector (using Hilbert transform) of the modulated signal.

For phase demodulation we pass it through FM demodulator (using differentiator and envelope detector) and then through integrator.

We subtract the mean of the signal obtained after passing through the FM demodulator to remove the offset and extract the information bearing modulated signal.

Here we can integrate the message signal either through **cumsum** or **cumtrapz** function.


Frequency modulation
Time domain of FM Signal
m(t)-->
time (s) -->
Frequency domain of FM Signal
m(f)-->
Frequency (Hz) -->
Time domain of FM demodulated Signal
m(t)-->
time (s) -->
Frequency Domain of FM demodulated Signal
m(f)-->
Frequency (Hz) -->


Phase modulation
Time domain of PM Signal
m(t)-->
time (s) -->
Frequency domain of PM Signal
m(f)-->
Frequency (Hz) -->
Time domain of PM demodulated Signal
m(t)-->
time (s) -->
Frequency Domain of PM demodulated Signal
m(f)-->
Frequency (Hz) -->