SCHOOL OF ENGINEERING
Faculty of Science and Engineering

MACQUARIE University

ELEC4844/8844 Practical Class – Week 8, 2024

Analogue Communication 2

TASK 1

The file 'audio1.wav' supplied previously for Practical W7 contains an audio signal sampled at 44.1 kHz for 10 seconds. Write a MATLAB program to implement single sideband lower sideband (SSB-LSB) AM.

a) Read the file to obtain the audio signal $m(t)$ and interpolate it by 10 times. Plot $m(t)$ as a function of time and its PSD in the unit of dBW/Hz.

b) AM-SSB-LSB modulate $m(t)$ to f_c = 150 kHz by

$$s_{\mathrm{LSB}}(t) = \frac{1}{2}[m(t)\cos 2\pi f_c t + \hat{m}(t)\sin 2\pi f_c t]$$

in which $\hat{m}(t)$ is the Hilbert transform of $m(t)$. Plot $s_{\mathrm{LSB}}(t)$ as a function of time and its PSD in the unit of dBW/Hz.

c) Obtain the complex envelope of the AM-SSB-LSB signal as the following

$$\tilde{s}_{\mathrm{LSB}}(t) = \frac{1}{2}[m(t) - j\hat{m}(t)]$$

Plot the PSD of $\tilde{s}_{\mathrm{LSB}}(t)$ in the unit of dBW/Hz.

d) Verify in the time domain that

$$s_{\mathrm{LSB}}(t) = \mathrm{Re}\left[\tilde{s}_{\mathrm{LSB}}(t)e^{j2\pi f_c t}\right]$$

TASK 2

Write a MATLAB program to:

a) Use your RTL-SDR device to receive the RF signal from an FM radio station of your choice (e.g. FM 96.9, with f_c = 96.9 MHz) at sampling rate of f_s = 960 kHz for a total period of 5 seconds. Specifically, set the number of samples per frame as L = 96000, and record 50 consecutive frames.

Note that it may be desirable to disable the automatic gain control (AGC) of the tuner and set a constant tuner gain instead, e.g. 30 dB. Also, mind the output data type and make sure it facilitates numeric calculations later.

b) Inspect the received FM signal, which is complex-valued. Plot the PSD of the received signal in the unit of dBW/Hz. Note the RTL-SDR device has already translated the signal to the baseband.

c) The complex-valued FM signal received by the RTL-SDR device contains both the in-phase and the quadrature-phase components, therefore allowing direct FM demodulation based on phase differentiation. Perform this FM demodulation to obtain the audio signal, and further downsample it to f_s = 48 kHz. Plot the audio signal as a function of time, and play the audio signal.

Note that you can use MATLAB function soundsc (instead of sound) to help normalise the volume.

TASK 3

a) Refer to the textbook Exercise 9.1 (pp. 336–339), use the Simulink model "…\fm\simulation\fm_nfm.slx" and follow the steps to explore the narrowband frequency modulation.

b) Refer to the textbook Exercise 9.2 (pp. 343–347), use the Simulink model "…\fm\simulation\fm_wfm.slx" and follow the steps to explore the wideband frequency modulation.

c) Refer to the textbook Exercise 10.6 (pp. 402–404), use the Simulink model "…\fm\rtlsdr_rx\rtlsdr_fm_discrim_stereo_demod.slx" and follow steps (l) – (p) to explore stereo demodulation of FM radio signals received by your RTL-SDR device.