Faculty of Science and Engineering



ELEC4844/8844 Practical Class – Week 7, 2024 Analogue Communication 1

TASK 1

- a) Refer to the textbook Exercise 6.1 (pp. 207–209), use the Simulink model
- "...\am\simulation\am_dsb_sc.slx" and follow the steps to explore the double sideband suppressed carrier amplitude modulation (AM-DSB-SC).
- b) Refer to the textbook Exercise 6.2 (pp. 214–216), use the Simulink model
- "...\am\simulation\am_dsb_tc.slx" and follow the steps to explore the double sideband transmitted carrier amplitude modulation (AM-DSB-TC).

TASK 2

The file 'audio1.wav' contains an audio signal sampled at 44.1 kHz. Write a MATLAB program to:

- a) Read the file to obtain the audio message signal x(t). Plot the message signal in the time domain, and its PSD in the unit of dBW/Hz.
- b) Interpolate x(t) by 6 times to $x_u(t)$ at sampling frequency of 264.6 kHz. Plot the interpolated signal in the time domain, and its PSD in the unit of dBW/Hz.
- c) AM-DSB-TC modulate the signal to $x_{AM}(t) = [1 + x_u(t)]\cos 2\pi f_c t$, with $f_c = 105$ kHz. Plot $x_{AM}(t)$ in the time domain, and its PSD in the unit of dBW/Hz.
- d) Demodulate $x_{AM}(t)$ using the optimised envelope detection as follows:
 - 1. take the magnitude of $x_{AM}(t)$;
 - 2. downsample by 6 times;
 - 3. remove the DC bias (i.e. mean value);
 - 4. apply a lowpass filter to obtain $y_{\text{dem}}(t)$.

The lowpass filter in the last step is a FIR filter with order of 50 and cutoff frequency at 5 kHz. Plot the demodulated signal $y_{\text{dem}}(t)$ in the time domain, and its PSD in the unit of dBW/Hz.

- e) Now consider an additive white Gaussian noise $\eta(t)$ to be added to the modulated audio signal $x_{AM}(t)$, with SNR = 10, 20, 40, and 60 dB. Create $\eta(t)$ and plot the PSD in the unit of dBW/Hz.
- f) Using the same method of optimised envelope detection above to AM-demodulate the received signal containing the additive while Gaussian noise. Discuss the quality of the demodulated signal as the SNR varies in terms of the time-domain and frequency-domain plots and the audio quality.

TASK 3

Write a MATLAB program to:

- a) Record your own voice at 44.1 kHz for 10 seconds to produce a new file named 'audio2.wav'.
- b) Similar to Task 2, interpolate the message signal by 6 times to obtain $x_u(t)$. Then, AM-DSB-SC modulate the signal to $x_{AM}(t) = x_u(t)\cos 2\pi f_c t$, with $f_c = 105$ kHz.
- c) AM demodulate the received signal using the coherent method as follows:
 - 1. multiply $x_{AM}(t)$ with a local carrier $\cos 2\pi f_{dem}t$, with $f_{dem} = 105$ kHz;
 - 2. downsample by 6 times;
 - 3. apply a lowpass filter to obtain $y_{\text{dem}}(t)$.

The lowpass filter can be the same FIR filter designed in Task 2d.

d) Induce a frequency error $\Delta f = 5$, 20, and 100 Hz to the local carrier frequency, i.e. $f_{c,\text{dem}} = 105.005$, 105. 02, and 105.1 kHz. Explore the influence on the coherently AM demodulated signal as Δf varies in terms of the audio quality.