EEE3030 Signal Processing and Machine learning

Semester 1 assignment – Digital Signal Processing

Submission format: pdf document submitted to Canvas

Submission deadline: 14:00 on 12th December 2025

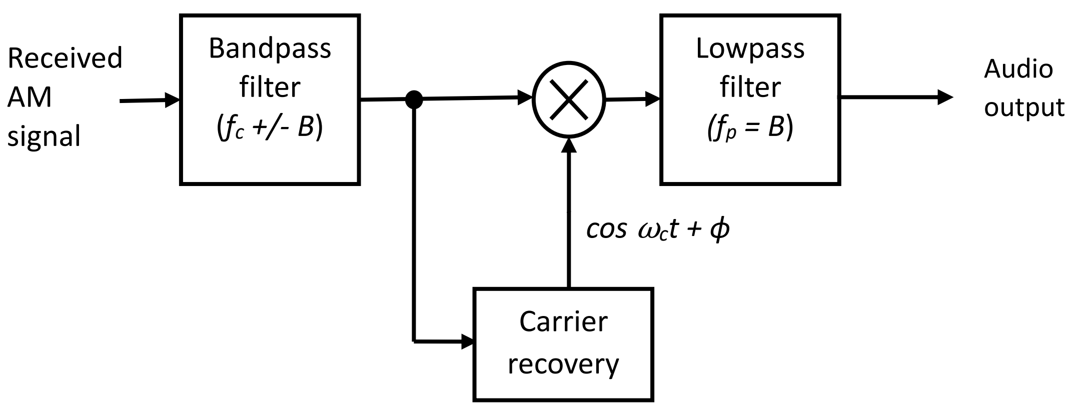
Introduction

An amplitude modulation (AM) waveform is generated according to the equation below:

$$s(t) = \cos(2\pi f_c t)\, Km(t)$$

Where f_c is the carrier frequency, *m(t)* is the modulation signal (message) and *K* is the modulation depth. This results in a double sideband suppressed carrier (DSB-SC) signal with "sidebands" either side of the carrier frequency determined by the frequency content of the modulation signal. If the modulation signal has bandwidth *B* then the spectrum of the AM signal will occupy the band between $(f_c - B)$ and $(f_c + B)$.

A demodulator for DSB-SC is as shown in the diagram below. First the received AM signal is passed through a bandpass filter to remove most of the noise outside the band of interest. Next the filtered signal is fed through a coherent AM demodulator or product detector (https://en.wikipedia.org/wiki/Product_detector) which multiplies the signal by the original carrier signal and then applies a low pass filter to remove the unwanted signal at $2f_c$ and remaining out of band noise. The resulting signal is then the original modulation signal plus any noise contained within the AM frequency band. For the demodulation process to work correctly the frequency and phase of the local carrier signal ($\cos \omega_c t + \phi$) must match that of the incoming signal, and this is estimated by a process called *carrier recovery*.



You are provided with a personalised .wav file which contains a unique AM signal with noise added. The sampling frequency for the signal is 96 kHz. Using MATLAB you will analyse this signal in time and frequency domain to measure the parameters of the

AM signal and added noise. You will then implement the above demodulator structure, designing appropriate digital filters, estimating the carrier frequency and phase, and extracingt the modulation signal (B = 4kHz) which contains a spoken 3 letter message. The assignment report will describe your analysis and filter design process and present all relevant results of filter testing and application to the AM signal. The assignment should also include and describe any MATLAB code produced. The submitted report will be assessed according to the grading scheme below for a total of 80 available marks. **Submit your assignment as a pdf document with filename EEE3030_<YOURNAME>.pdf.**

Task 1 – Time and frequency domain analysis [15 marks]

First read the provided signal into MATLAB, using audioread(), then plot and inspect the signal in the time and frequency domain (using an FFT with the output scaled appropriately in amplitude and frequency), recording your observations on the signal and noise properties. Estimate the upper and lower limits of the spectrum of the AM signal (f_{min} and f_{max}).

Task 2 – Bandpass filter [20 marks]

Design a finite impulse response (FIR) bandpass filter, using the impulse response truncation method described in lectures, with the following specification:

Passband edge frequencies (f_p)	f_{min} and f_{max} (determined in task 1)
Stopband edge frequencies (f_{stop})	f_{min} – 2kHz, f_{max} + 2 kHz
Max passband ripple (dB)	0.1
Stopband attenuation (dB)	> 50 dB

Verify the frequency response of this filter, then apply this to the AM signal. Plot and describe the output signal in the time and frequency domain. If designed correctly, this should remove much of the out-of-band noise, preserving the AM signal spectrum.

Task 3 – Carrier recovery and mixing [15 marks]

Apply a square law ($| \, |^2$) to the bandpass filtered signal (from task 2) and compute the spectrum (FFT) of this signal. The squaring process will generate a peak at double the carrier frequency ($2f_c$) which you can use to estimate f_c (which will be an exact multiple of 1 kHz).
Once you have estimated f_c, generate the local carrier signal ($\cos \omega_c t + \phi$) and multiply with the bandpass filtered signal (task 2 output), plotting the output in time and frequency domain. (Use $\phi = 0$ as the optimum value will be determined in task 5).

Task 4 – Lowpass filter [20 marks]

Design an infinite impulse response (IIR) lowpass filter with the following specifications:

Order	4
Cut off frequency	4 kHz
Polynomial	Butterworth

Verify the frequency response of this filter, then apply this to the signal from task 3 and plot the output signal in the time and frequency domain.

Task 5 – Audio signal output [10 marks]

Determine ϕ by adjusting in the range (0 – π) until you observe the maximum output signal magnitude (in the time domain) at the output of the lowpass filter. Finally play back this signal using the MATLAB function sound() and listen to your 3 letter message.

<u>Grading scheme (rubric)</u>

Task 1
[0-8 marks] The student has read in the signal file and plotted in time and frequency domain using in built MATLAB functions e.g. pspectrum(). The broad characteristics of the signal have been described and the AM frequency band has been accurately measured.
[9–15 marks] As above but using basic fft() function, then applying correct amplitude normalisation and frequency scaling, logarithmic magnitude scaling (dB) and demonstrating awareness of frequency resolution, window size and spectral leakage control via window functions.

Task 2
[0-9 marks] The student has chosen appropriate cut off frequencies, designed the FIR filter correctly using MATLAB filter design functions, applied the filter to the AM signal using the MATLAB filter() function and demonstrated that it removes most of the out-of-band noise.
[10-20 marks] As above but the student has verified the frequency response of the filter and **implemented the FIR filter via their own code** to perform convolution.

Task 3
[0-5 marks] The student has correctly squared the filtered AM signal and plotted the result.
[6-10 marks] as above and the student has correctly identified the carrier frequency from the spectrum.
[11-15] above and multiplied with the filtered AM signal, plotting and explaining the output signal in time and frequency domain.

Task 4
[0-15 marks] The student has Correctly designed the IIR filter, verified the response and applied to the output of task 3 using the MATLAB filter() function.
[16-20 marks] As above but the student has verified the filter response and the **IIR filter is implemented via the student's own code** to perform the convolutions.

Task 5
[0-5 marks] The student has plotted the final demodulated signal, output as an audio signal and correctly identified the 3-letter spoken message.
[6-10 marks] As above but the student has adjusted the carrier phase to obtain the highest amplitude/SNR on the message signal.

In all of the above, marks are awarded for the quality of description, depth of understanding, presentation of results (e.g. correct scaling/labelling of graphs) and code comments/explanation.