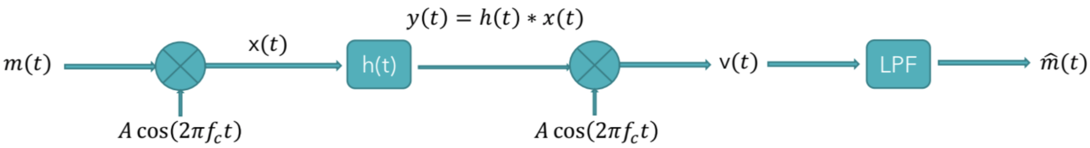
Double-Sideband Suppressed Carrier Report

Final Project

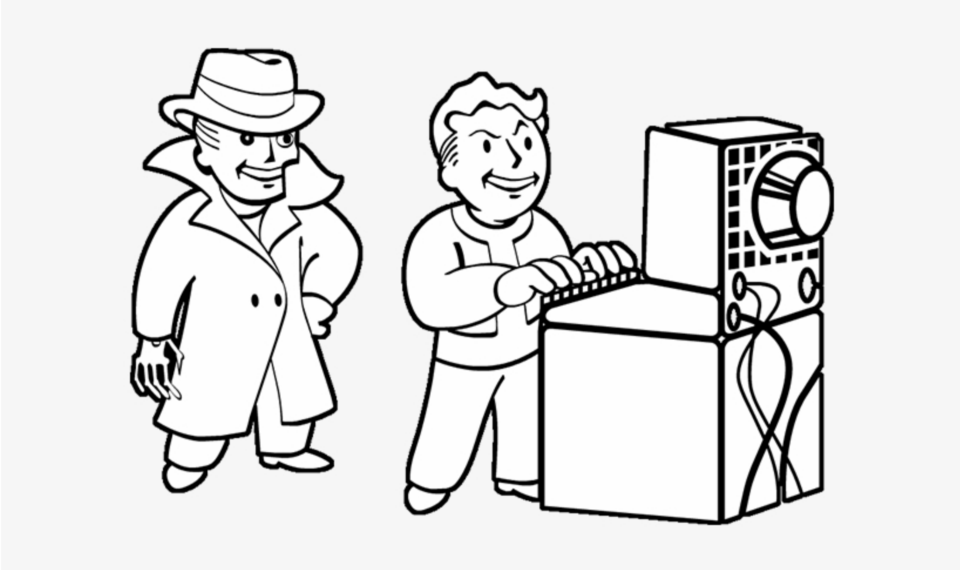
Design a DSB-SC communication system following below diagram



Given $H(f) = e^{-2j\pi 30t} + e^{-2j\pi 5t}$

Baseband signal $m(t)$ is composed with three frequencies based on your last 3 digits of your student ID.

E.g., your last 3 digits are 77-7, $m(t) = \cos(2\pi 700t) - \frac{1}{3}\cos(2\pi 70t) + \frac{1}{5}\cos(2\pi 7t)$



Introduction to le signal

(Double-Sideband Suppressed Carrier)le signal's goal

- To transmit a baseband signal
  - to perfom **modulation** on baseband signal
  - to perfom **demodulation** on signal
  - to perfom Fourier Transfrom on signal
  - to use digital filter on signal

Terminology

- ▼ The low frequency or the message signal is known as the **baseband signal**

  example of baseband signal
  - Human voice
- high-frequency periodic signal is known as **carrier signal.**
- **Modulation** changes the shape of a carrier wave to somehow encode the speech or data information that we were interested in carrying. Modulation is like hiding a code inside the carrier wave.
- **DSB-SC** is an amplitude modulated wave transmission scheme in which only sidebands are transmitted and the carrier is not transmitted as it gets suppressed. DSB-SC is an acronym for **D**ouble **S**ideband **S**uppressed **C**arrier.
  - **Advantages of DSB-SC**
    - provides a larger bandwidth
    - consumes less power.
  - **Disadvantages of DSB-SC**
    - Sometimes it is difficult to recover a signal at the receiver
    - A very expensive technique because it sends the signal double sided (upper and lower sideband)

Purpose

Can't we just send baseband signal on their own?? No, we can't. There are quite a lot of challenges of transmitting baseband signal on their own, and even tougher task if you want to transmitt the signal to further away target. If you want to transmitt a **baseband signal** in a long distance, it is a must to use high-frequency periodic signal to carries baseband signal. This high-frequencey signal is called **carrier signal**

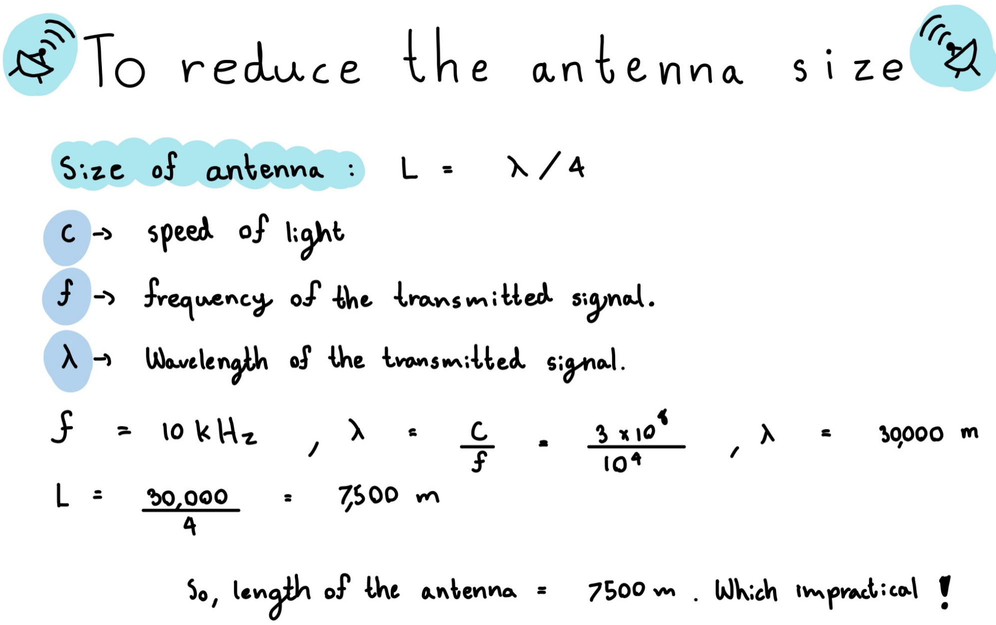
> In the modulation process, one of the carrier **signal properties** will be changed according baseband signal

💡 **Signal basic properties: Amplitude, Frequency and Phase.**

Benefits of modulation?

Antenna size

When the signal is transmitted wirelessly, the size of the antenna is a very important factor. Size of the antenna is proportional to the wavelength of the transmitted signal



Using modulation will shrink the frequency of the transmitted signal. For example, f = 10kHz could shrink into 10MHz, so L = 7.5, therefore the length of the antenna is more manageble and more practical.

$$\boxed{\lambda = \frac{c}{f}}$$

To reduce the interference

Signals that are around in the same frequency range are transmitted without the modulation through a signal channel, then there will be interference between them, and at the receiver end won't get any signals. However if each signal is modulated at different carrier freuencies, and all carrier frequencies are relatively remotely away from each other, then there won't be any interference and all the signals can be demodulated at the receiver and the message signal of each transmitter can be retrieved.

To all multiplexing of the signals

▼ Example of the frequency division multiplying

The different message signals are modulated at the different carrier frequencies and because of that, they can be transmitted simultaneously.

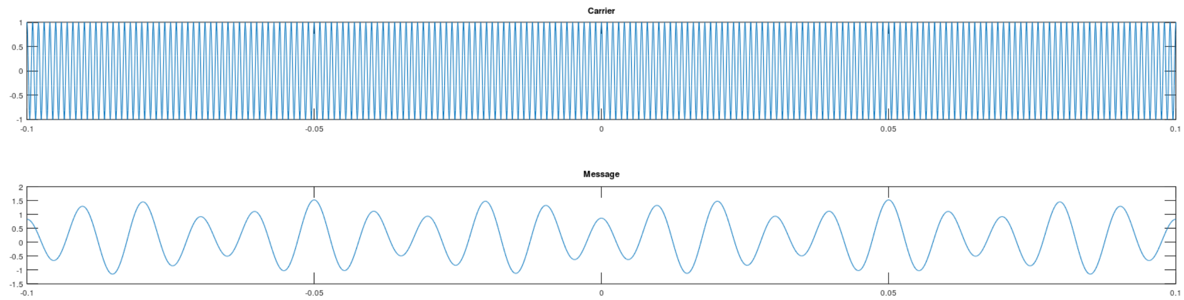
le signal program

> Make sure pkg load signal everytime

First of all, we going to have to initialised variables.

I've set my time vector "t" to t = -0.1:1/Fs:0.1. My carrier frequency "fc" is 1000. My sampling frequency "fs" to frequency of carrier times 10.

These two signals are our **message signal(baseband signal)** and **carrier signal**

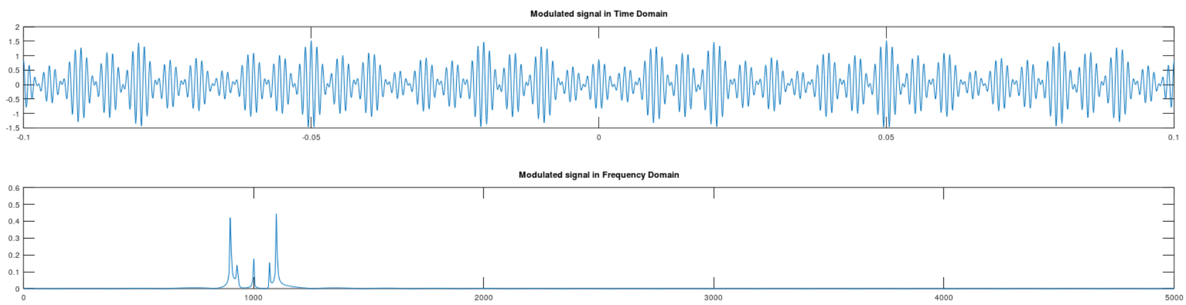


Modulation

$$m(t) = \cos(2*\pi*100*t) - \frac{1}{3}*\cos(2*\pi*70*t) - \frac{1}{5}*\cos(2*\pi*1*t)$$

$$c(t) = A*\cos(2*\pi*f_c*t)$$

Then we achieve modulation through multiplication in the time domain. To translate to code, we multiply of arrays M and C by multiplying corresponding elements or in other words scalar multiplication. As a result, we get a modulated signal



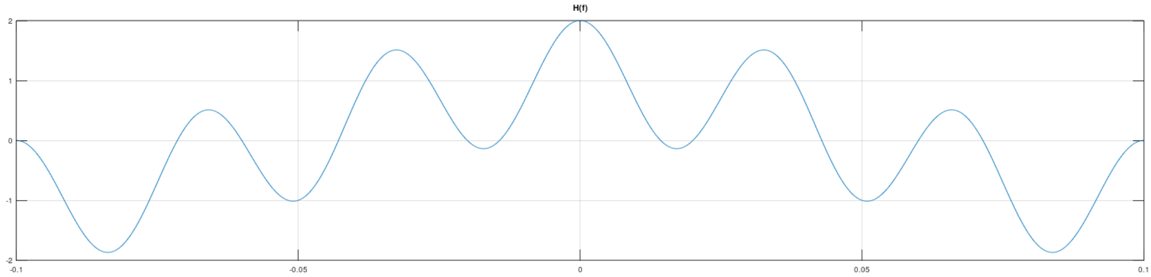
$$x(t) = [cos(200*\pi*t) - \frac{1}{3}cos(140*\pi*t) + \frac{1}{5}\cos(2*\pi*t)] * \cos(2000*\pi*t)$$

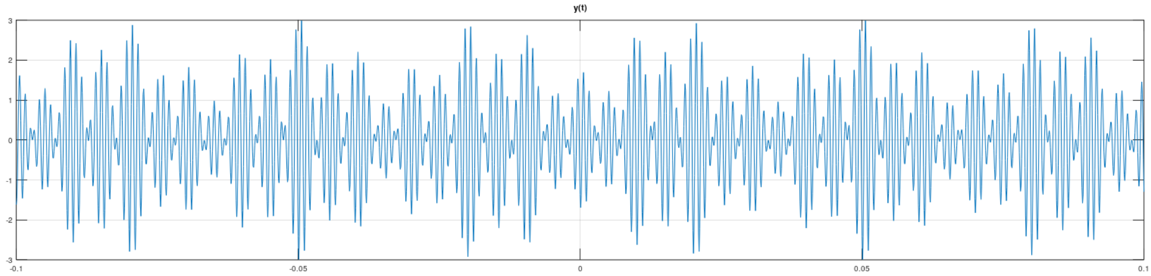
$$x(f) = \frac{1}{2}[\pi\delta(f-1800\pi) + \delta(f+1800\pi)] - \frac{1}{6}[\pi\delta(f-1860\pi) + \delta(f+1860\pi)] + \frac{1}{10}[\pi\delta(f-1998\pi)$$

$$+\delta(f+1998\pi)] + \frac{1}{10}[\pi\delta(f-2002\pi) + \delta(f+2002\pi)] - \frac{1}{6}[\pi\delta(f-2140\pi) + \delta(f+2140\pi)] + \frac{1}{2}[\pi\delta(f-2200\pi$$

$$+\delta(f+2200\pi)]$$

There's a reason why there are two words "suppressed carrier" in the name of **DSB-SC.** It is because carrier signal is suppressed, it does not appear in output (Modulated signal) signal.





Signal distortion

Now, signal distortion, Distortion, in electronics, an change in a signal that alters the basic waveform or the relationship between various frequency components.

Major types of distortions:

- Linear distortion
  - Amplitude distortion, refers to unequal amplification or attenuation of the various frequency components of the signal. For example,h(f) is not constant with frequency.
  - Phase distortion, refers to changes in the phase relationships between harmonic components of a complex wave.
- Non-linear distortion
  - Non linear distortion results from systems where the output signal is not exactly proportional to the input signal

Two properties that needed to be satisfied in order to acheive distortionless channel

- linear shift phase
- flat frequency response

first step of this process is we initialise variable h(f).

h(f) = $e^{-2j\pi 30t}+e^{-2j\pi 5t}$(First graph), then we use Fourier transfrom to transfrom our modulated signal into frequency domain. In Octave or Matlab, there is **fft() function(fast Fourier transform)** to simplifies the process and saves time in transfromation process. lastly, as the result, we'll get y(f). To get y(t), we have to do a fourier transfrom on y(f), this could be easily achieved through the use of **ifft() fuction (inverse fast Fourier transfrom)**

$$y(t)=[e^{-\pi*t*10i}+e^{-\pi*t*60i}]*\frac{1}{2}[\pi(e^{-\pi t1800i}+e^{\pi t1800i})]-\frac{1}{6}[\pi(e^{-\pi t1860i}+e^{\pi t1860i})]+\frac{1}{10}[\pi(e^{-\pi t1998i}+e^{\pi t1998i})].$$

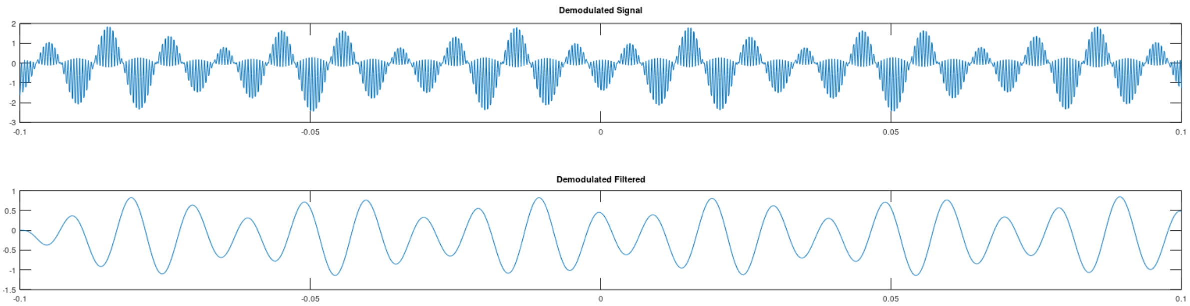
$$+\frac{1}{10}[\pi(e^{-\pi t2002i}+e^{\pi t2002i})]+\frac{1}{6}[\pi(e^{-\pi t2140i}+e^{\pi t2140i})]+\frac{1}{2\pi}[\pi(e^{\pi t2200i}+e^{\pi t2200i})]$$

$$y(f)=[e^{-\pi t10i}+e^{-\pi t60i}]*\frac{1}{2}[\pi\delta(f-1800\pi)+\delta(f+1800\pi)]-\frac{1}{6}[\pi\delta(f-1860\pi)+\delta(f+1860\pi)]$$

$$+\frac{1}{10}[\pi\delta(f-1998\pi)+\delta(f+1998\pi)]+\frac{1}{10}[\pi\delta(f-2002\pi)+\delta(f+2002\pi)]-\frac{1}{6}[\pi\delta(f-2140\pi)+\delta(f+2140\pi$$

$$+\frac{1}{2}[\pi\delta(f-2200\pi)+\delta(f+2200\pi)]$$

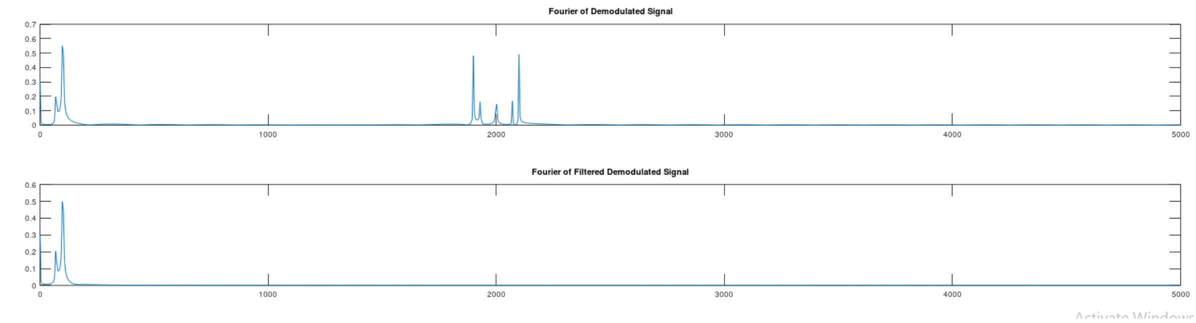
Demodulation and filter



$$v(t)=[\cos(2000\pi t)*e^{-\pi t10i}+e^{-\pi t60i}]*\frac{1}{2}[\pi(e^{-\pi t1800i}+e^{\pi t1800i})]-\frac{1}{6}[\pi(e^{-\pi t1860i}+e^{\pi t1860i})]+$$

$$\frac{1}{10}[\pi(e^{-\pi t1998i}+e^{\pi t1998i})]+\frac{1}{10}[\pi(e^{-\pi t2002i}+e^{\pi t2002i})]+\frac{1}{6}[\pi(e^{-\pi t2140i}+e^{\pi t2140i})]+\frac{1}{2\pi}[\pi(e^{\pi t2200i}+e^{\pi t2200i})]$$

Finally, demodulation and filter, to recover our message signal we will need to perform demodulation on our **y(t)**. To perform demodulation on signal, simply just like modulation, we multiply our signal **y(t) with carrier signal**, as the result we get demodulated signal **v(t)**. Also, It is necessary to filter out unwanted noise signals from the measurement. we want to extract our message signal so we will be using **low pass filter**. With this filter, only low frequency will be able to pass through, dropping all other high frequency noises.



Here are the graphs for Demodulated signal and filtered demodulated signal in frequency domain.

Conclusion

In this project, the goal was to observe and study the mechanic of DSB-SC, and a way to ustilise this very technique to transmit a baseband signal. The process can be simplified into baseband signal getting put through modulation by multiplying with the carrier signal, signal distortion, **demodulation** by multiplying the DSB-SC signal with the carrier signal (with the same phase as in the modulation process) and eventually getting fillltered by low pass band filter. As the result, if everything goes according to plan, baseband signal will be recovered. While DSB-SC consumes less power and provides 100% modulation efficiency, Therefore allows us to have a transmission at the lower power consumption. It is rather expensive when you are performing demodulation to retrieve original baseband signal and other thing to consider, it could be troublesome during demodulation as well. Please be informed that all things that've been written in this report could be partially differ to other implementations. This project main focus is to only perform DSB-SC procedure on a low frequency baseband signal, everything else is out of the scope of this very project. DSB-SC number of applications is not limited to just transmit low freqency signal like human voice. It is known that DSB-SC technique is also being implentment in Television broadcasting which was not covered in this project, because it may required to tweak DSB-SC procedures, processes ,and etc in order to work. The writer would like to end this report by recommending DSB-SC to ameteur DSP engineers, for learning purposes because the amount of experiences and knowledges that could be obtain during researching and practising this transmission technique whether by hand or coding calculation, is very rewarding.