Wired and Wireless Communication System

DSB-SC Communication System Final Project

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Executive Summary

The aim of this project is to design a DSB-SC communication systems for both modulation and demodulation of the baseband signal based on last 3 digits of student ID. The message signal and carrier wave will undergo amplitude modulation which form the DSB-SC signal and will pass through channel and then demodulator.

During the transmission to demodulation process, the signal will first undergo the distortion of the channel (e.g., wires) before going through demodulator. And afterward the demodulated signal will pass through the filter, LPF as known as low pass filter to filter unwanted signal such as noise by filtering out high frequencies. In the end, the filtered demodulated signal will reassembles the original message in the very beginning. It is also worth noticing that the filtered demodulated signal's amplitude will be smaller than the original message signal.

To make it easier to comprehend, the essential concept and principle of modulation and demodulation of signals are described in this paper with various illustrations using the Octave model. Since Octave has a range of tools for performing common numerical linear algebra problems, including such identifying the roots of nonlinear equations, integrating ordinary functions, manipulating polynomials, and integrating ordinary differential and differential-algebraic equations.

Introduction

Indeed, communications plays a crucial role in human existence. To communicate or transmit information over a long distance, modulation and demodulation is required.

A signal is any time-varying physical occurrence that transmits information in communications. Human speech, MP3 music, ECG signals, and MPEG films are examples of valuable or wanted signals, while noise and interference are examples of signals that are regarded worthless and unwelcome.

This report will focus mainly about the DSB-SC system, the double-sideband suppressed-carrier transmission is an amplitude modulated (AM) wave transmission, which consumes less power with larger bandwidth. The system that will be presented in this report will include the modulation and demodulation process as well as the channel distortion during the transmission process.

The report will begins with brief introduction of the system and signal that will be used in this report as an example for clearer interpretation with graphs. Modulation, channel characteristics and distortions as well as demodulation of signals will be presented both in graph and hand calculation and will be used to analyze the system in term of modulation, effect of channel, and demodulation.

Discussion

The basic structure of Double-sideband suppressed-carrier transmission or (DSB-SC) is shown in fig 1.

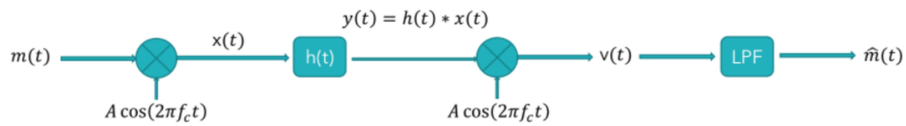

Figure 1 *Simple block diagram of DSB-SC*

1. Modulation

The process of encoding information in a transmitted signal is known as modulation. The modulator, as shown in fig. 1, combines the message signal to be delivered with a carrier signal. Carrier signal plays an important role to carry signals from one location to another. In this report, the system uses product modulator. (See Appendix A for the code.)

The baseband signal is represented by the equation (consist of last 3 digits of student ID):

$$m(t) = \cos(2\pi F m_1 t) - \frac{1}{3}\cos(2\pi F m_2 t) + \frac{1}{5}\cos(2\pi F m_3 t) \qquad (1)$$

Where F_m = Message frequency; in this report uses 200, 10, 3 respectively.

T = Time; in this report uses -0.1:1/ 10000:0.1 as a time vector

The baseband signal model is in Fig 2.

The carrier signal is represented by the equation:

$$c(t) = A(cos 2\pi F_c t) \qquad (2)$$

Where A = Modulation index; this report uses 1

Fc = Carrier wave frequency; 1000

The carrier wave model is shown below in Fig 3.

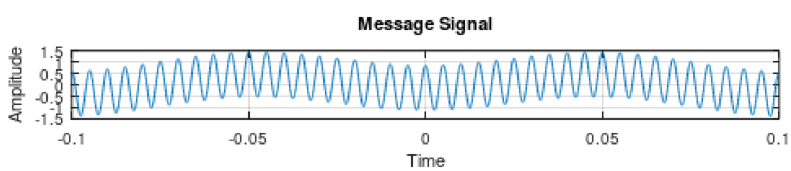

Figure 2 *Baseband Signal Mode m(t)l*

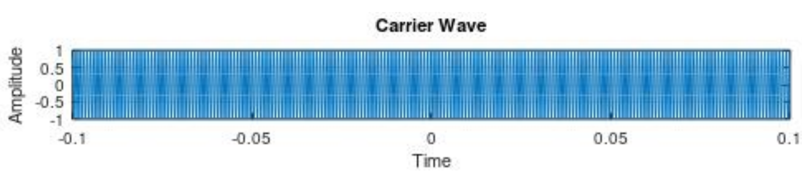

Figure 3 *Carrier Wave Model c(t)*

Double Sideband Suppressed Carrier (DSBSC) is an amplitude modulation technique in which the modulated wave contains both the sidebands along with the suppressed carrier.

We can represent the equation of DSBSC wave as the product of the message signal m(t) and the carrier wave c(t).

$$x(t) = m(t) \times c(t) \qquad (3)$$

$$x(t) = A(cos(2\pi Fm_1 t) - \frac{1}{3}\cos(2\pi Fm_2 t) + \frac{1}{5}\cos(2\pi Fm_3 t)) \times (cos2\pi F_c t))$$

Replacing the variables with values from Equation 1 and 2 will give us:

$$x(t) = (cos(2\pi 200t) - \frac{1}{3}\cos(2\pi 10t) + \frac{1}{5}\cos(2\pi 3t)) \times (cos2\pi 1000t))$$

The modulated signal x(t) is plotted below in fig.4.

Notice that x(t) undergoes phase reversal whenever m(t) crosses zero. The upper and lower envelope of the graph is the message/baseband signal, inside the envelope is the suppressed carrier. This is an example of Amplitude Modulation or AM.

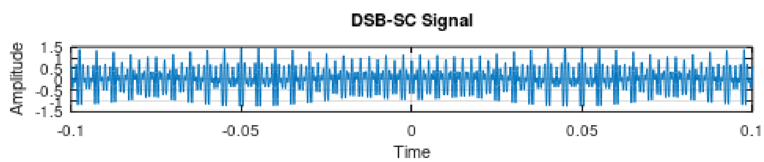

Figure 4 *DSBSC transmission Signal*
Modulating DSB-SC Signal in Time Domain x(t)

By converting to the frequency domain, we can see what frequencies have been added to the signal due to the modulation. A Fourier transform of a signal tells you what frequencies are present in your signal and in what proportions.

Performing Fourier transform on message signal, m(t) will gives:

$$m(f) = \frac{1}{30}(15\delta(w-200) - 5\delta(w-10) + 3\delta(w-3) + 3\delta(w+3) - 5\delta(w+10) + 15\delta(w+200))$$

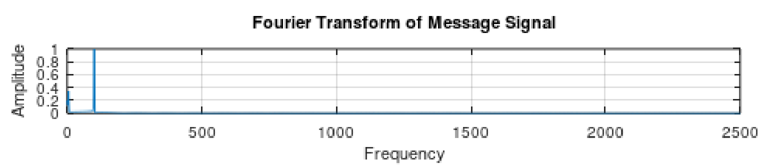

Figure 5 *Fourier Transform of Message Signal*

While performing Fourier transform on modulated signal, x(t) will gives:

$$x(f) = \frac{1}{60}(15\delta(w-1200) - 5\delta(w-1010) + 3\delta(w-1003) + 3\delta(w-997) - 5\delta(w-990) + 15\delta(w-800) + 15\delta(w+800) - 5\delta(w+990) + 3\delta(w+997) + 3\delta(w+1003) - 5\delta(w+1010) + 15\delta(w+1200))$$

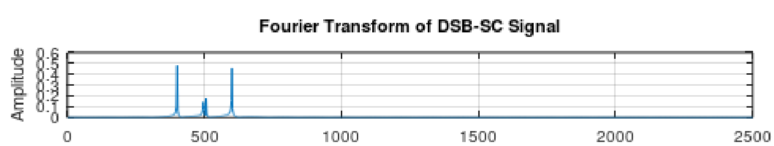

Figure 6 *Fourier Transform of DSB-SC Signal*

2. Channel Distortion

Distortion usually refers to a degradation of the signal.

Channel characteristic (in frequency domain) equation:

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

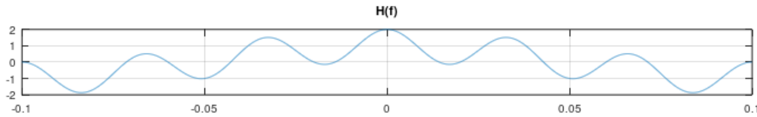

Figure 7 *H(f) Channel Characteristic Graph*

Then perform fast Fourier transform on the modulated signal, x(t) to get modulated signal in frequency domain, x(f) in order to do matrix multiplication with h(f).

The FFT function is calculated along the first non-singleton dimension of the array.

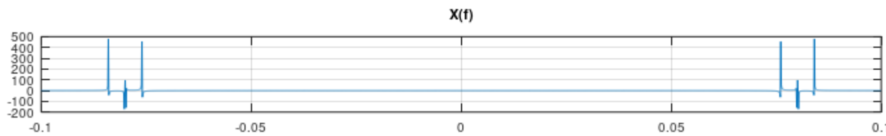

Figure 8 *Modulated Signal in Frequency Domain, x(f)*

Perform matrix Multiplication of x(f) and h(f) to get y(f), the distorted signal in frequency domain.

$$y(f) = x(f) \times h(f) \qquad (4)$$

$$y(f) = \frac{1}{60}(15\delta(w-1200) - 5\delta(w-1010) + 3\delta(w-1003) + 3\delta(w-997) - 5\delta(w-990) + 15\delta(w-800) + 15\delta(w+800) - 5\delta(w+990) + 3\delta(w+997) + 3\delta(w+1003) - 5\delta(w+1010) + 15\delta(w+1200))) \times (e^{-2j\pi 30t} + e^{-2j\pi 5t})$$

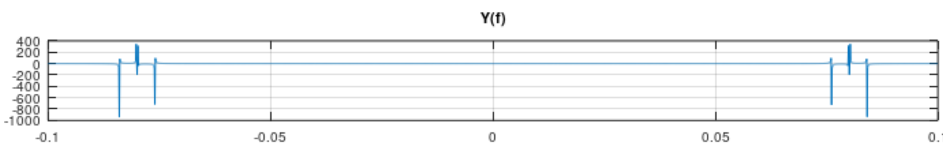

Figure 9 *Distorted Modulated Signal in Frequency Domain*

Lastly, perform inverse Fourier Transform of Y(f) to convert back to time domain to obtain y(t), message signal that undergo channel distortion in time domain. The graph below can be obtain by using inverse fast Fourier transform, ifft function. The ifft function, compute the inverse discrete Fourier transform by using a Fast Fourier Transform (FFT) algorithm. Notice that after the signal pass through channel, there's an interference with the modulated signal. It can be observe that the signal now doesn't resemble the DSB-SC signal.

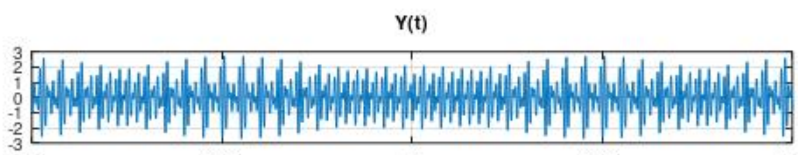

Figure 10 *Distorted Modulated Signal in Time Domain*

3. Demodulation

Demodulation is the process of retrieving information from a transmitted signal by using the demodulator to extract the original message or the baseband signal from the modulated carrier. In short, it's the reverse process of modulation. This is one of the disadvantages of using a DSBSC signal modulation, extracting information at the receiver is difficult and can be quite challenging. (See Appendix C)

The demodulated signal, v(t) can be obtain by multiplying distorted signal, y(t) to the carrier wave, c(t).

$$V(t) = y(t) \times c(t) \qquad (5)$$

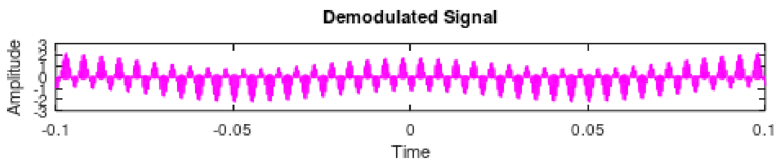

Figure 11 *Demodulated Signal v(t)*

Then perform low pass filter (LPF) on the demodulated signal to remove unwanted noise. A low pass filter removes undesirable frequencies from a signal that seem to be beyond a certain cutoff frequency. To put it another way, it allows low frequencies travel through during filtering out high frequencies.

The graph below Fig. 12 represents the filtered demodulated signal, m'(t) using butter function in octave. After the noise is removed the signal pattern can be observe that it's reassemble the original message but with smaller amplitude.

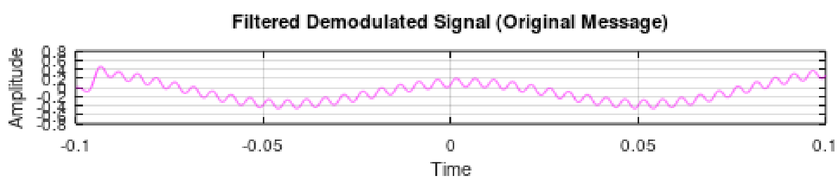

Figure 12 *Filtered Demodulated Signal (Original Message)*

The graph below Fig. 13 is the graph of Fourier transform v(f) of demodulated signal v(t).

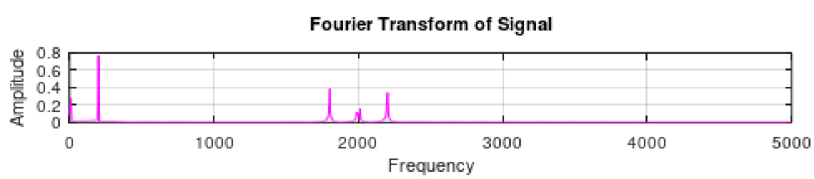

Figure 13 *Fourier Transform of Demodulated Signal*

Filtered the Fourier transform of the demodulated signal, v'(f). Same as above graph, using butter function in octave to apply low pass filter on the signal.

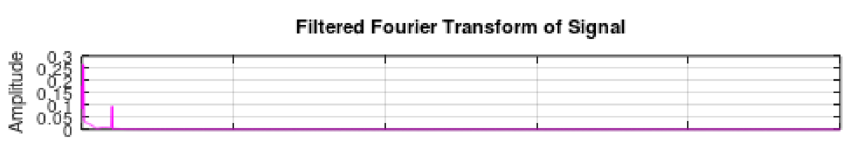

Figure 14 *Filtered Fourier Transform of Demodulated Signal*

Conclusion

This report has been able to implemented DSB-SC communication system in Octave as well as hand calculation. We were able to generate modulated and demodulated signals as well as the channel distortion’s signal. From Fig. 2 to Fig. 6 shows the modulation section of the system. In this report, the message frequency consists of 3 different message frequencies, 200, 10, and 3 respectively. And the carrier wave frequency is 1000. The modulation index is 1. The modulated signal can easily be obtain by multiplication between message or the baseband signal with the carrier wave. In this way we will obtain DSB-SC signal, which is an example of amplitude modulation or AM.

The second part of the report discussed about channel distortion that degraded the signal during the transmitting process to the receiver. Channel distortion refers to degradation of signals. In this case the modulated signal is converted to frequency domain by fast Fourier Transform in order to do matrix multiplication with the channel characteristic. Now, the signal is distorted by the channel and converted back to time domain by using inverse fast Fourier transform (Fig. 7 - 10). Now the signal will go through demodulation process.

And lastly extracting the information from signal by demodulation. The demodulated signal can be obtain by multiplying the distorted message signal by the channel to the carrier wave. The result, then, undergo the low pass filter, LPF to remove unnecessary signal such as noise to see the original message. Note that the message signal that receiver received will be slightly distorted due to transmission process (effect from the channel). Furthermore, both the demodulated signal and the filtered version of it can undergo Fourier transform. (Fig.11 - 14)

Appendices

Note: Source code is also attached in the google classroom assignment.

Appendix A

```
function moddemod = project

   %Variables
  Fc = 1000; #Carrier Frequency
  Fs = Fc*10; #sampling frequency
  Fm1 = 200; #Message Signal 1
  Fm2 = 10; #Message Signal 2
  Fm3 = 3; #Message Signal 3
  ET = 0.1; # End-Time
  t = -0.1:1/Fs:ET; #Time vector
  A =  1;  #Modulation Index

   % Equation stuff
  mt = cos(2*pi*Fm1*t) - (1/3)*cos(2*pi*Fm2*t) + (1/5)*cos(2*pi*Fm3*t); #message signal
  cw = A.*(cos(2*pi*Fc*t)); #carrier wave
  xt = mt.*cw; #modded signal

  %-----------Modding-------------

  %Graph 1
  figure(1);
  subplot(4,1,1)
  plot(t,mt);
  xlabel('Time'), ylabel('Amplitude'); #Message Signal
  grid on
  title('Message Signal')
```

```
%Graph 2
figure(1);
subplot(4,1,2)
plot(t,cw);
xlabel('Time'), ylabel('Amplitude'); #Carrier Wave
grid on
title('Carrier Wave')


%Graph 3
figure(1);
subplot(4,1,3)
grid on;
plot(t,xt);
xlabel('Time'), ylabel('Amplitude'); #DSB-SC Signal
grid on
title('DSB-SC Signal')


%Graph 4
#  L = length(mt);
L = length(xt);
NFFT = 2^nextpow2(L);
#  y_fft = 2*abs(fft(mt,NFFT)/L);
y_fft = 2*abs(fft(xt,NFFT)/L);

freq = Fs/4*linspace(0,1,NFFT/2+1);
#freq = Fs/2*linspace(0,1,NFFT/2+1);

figure(1);
subplot(4,1,4) #Fourier
plot(freq,y_fft(1:NFFT/2+1));
grid on
```

```
  xlabel('Frequency');
  ylabel('Amplitude');
#  title('Fourier Transform of Message Signal')
  title('Fourier Transform of DSB-SC Signal')
```

Appendix B

```
%------------Channel Distortion----------------
  hf = exp(-2i*pi*5*t)+exp(-2i*pi*30*t); #channel charac
  xf = fft(xt); #fast fourier transform -> freq domain
  yf = xf.*hf; #Matrix multiplication
  yt = ifft(yf); #inverse fourier transform -> time domain

    % uncomment to see the graph
#  figure(3);
#  subplot(4,1,1);
#  plot(t,hf); grid on; title("H(f)")

#  figure(3);
#  subplot(4,1,2);
#  plot(t,xf); grid on; title("X(f)")

#  figure(3);
#  subplot(4,1,3);
#  plot(t,yf); grid on; title("Y(f)")

#  figure(3);
#  subplot(4,1,4);
#  plot(t,yt); grid on; title("Y(t)")
```

Appendix C

```
%------------Demodding----------------

 %Graph 1
 vt = yt.*cw;
 figure(2);
 subplot(4,1,1)
 plot(t,vt,'m');
 xlabel('Time'), ylabel('Amplitude');
 title('Demodulated Signal');



 %Graph 2
 fvt = vt;
 L = length(fvt);
 NFFT = 2^nextpow2(L);
 y_fft = 2*abs(fft(fvt,NFFT)/L);
 freq = Fs/2*linspace(0,1,NFFT/2+1);
#  freq = Fs*linspace(0,1,NFFT/2+1);
 figure(2);
 subplot(4,1,2)
 plot(freq,y_fft(1:NFFT/2+1),'m');
 grid on
 xlabel('Frequency');
 ylabel('Amplitude');
 title('Fourier Transform of Signal');

#LPF

 %Graph 3
 Fs1 = Fs/2;
 [b,a]=butter ( 4, 120 / Fs1);
```

```
  filtered = filter(b,a,vt);
  figure(2);
  subplot(4,1,3)
  plot(t,filtered,'m');
  grid on
  xlabel('Time');
  ylabel('Amplitude');
  title('Filtered Demodulated Signal (Original Message)');


  %Graph 4
  [b,a]=butter ( 4, 120 / Fs1);
  filtered = filter(b,a,fvt);
  L = length(filtered );
  NFFT = 2^nextpow2(L);
  y_fft = 2*abs(fft(filtered ,NFFT)/L);
#  freq = Fs*linspace(0,1,NFFT/2+1);
#  freq = Fs/2*linspace(0,1,NFFT/2+1);
  freq = Fs/4*linspace(0,1,NFFT/2+1);
  subplot(4,1,4)
  plot(freq,y_fft(1:NFFT/2+1),'m');
  grid on
  xlabel('Frequency');
  ylabel('Amplitude');
  title('Filtered Fourier Transform of Signal');

endfunction
```