Massachusetts Institute of Technology
Dept. of Electrical Engineering and Computer Science
Fall Semester, 2006
6.082 Introduction to EECS 2

Lab#3: Frequency-Division Multiplexing

Objectives

In this lab you will learn about a communication scheme known as frequency division multiplexing. You will then implement, in software, the signal processing operations necessary to receive voice signals transmitted using this communication scheme.

Introduction

Frequency Division Multiplexing (FDM) is a communication scheme that allows multiple information signals to be sent simultaneously across a communication channel. FDM assigns to each information signal a frequency band within the communication channel's spectrum. Since the frequency bands are chosen so they don't overlap, the information signals travel through the channel without interference. Figure 1 illustrates the transmission of two information signals using FDM.

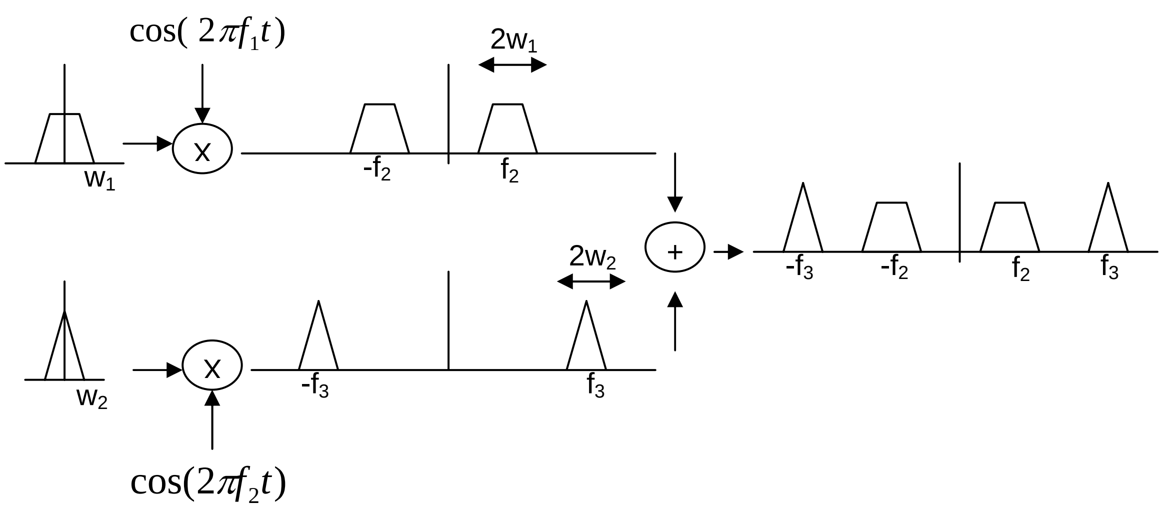


Figure 1

In Figure 1 we have two information signals with spectra shaped as a trapezium and a triangle respectively. To send these information signals simultaneously over the same communication channel using FDM, we *modulate* each of the information signals to a different frequency band. In this example, we modulated the information signal corresponding to the trapezium spectrum to a frequency band with center frequency f_2 and bandwidth w_1; and we modulated the information signal with the triangular spectrum to a frequency band with center f_3 and bandwidth w_2. By adding these two modulated information signals we obtain a *transmission signal* that can carry the spectra of both information signals across the communication channel simultaneously.

At the receiver the transmission signal has to be processed in order to extract the information signal of interest. This processing involves removing the unwanted spectral components from the transmission signal, and then demodulating the spectral component of interest back to DC (0Hz). As an example, Figure 2 illustrates the processing steps necessary in order to extract the information signal with the trapezium spectrum.

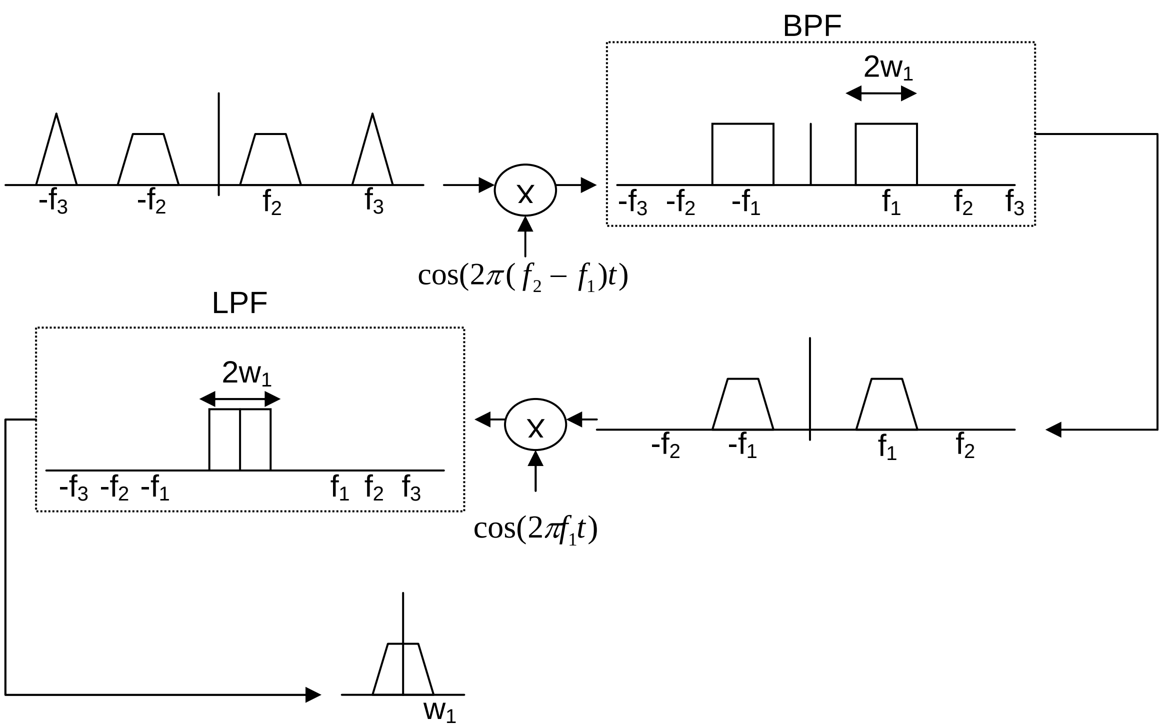


Figure 2

First the transmission signal is modulated so that the spectral component of interest is positioned at the center frequency of a bandpass filter; in this case the trapezium spectrum is shifted so that it is centered at f_1. The bandpass filter keeps only the spectral components of the information signal of interest (the trapezium) and removes the spectral components corresponding to the second information signal (the triangle).

Next, the trapezium spectrum is demodulated to DC (the original center frequency of the information signal spectrum) and low pass filtered in order to remove the higher frequency demodulation products (at $2f_1$ and $-2f_1$). The result is the information signal with the trapezium spectrum.

The same process is used to extract the information signal with the triangular spectrum except now the receiver modulates the received signal by $\cos(2\pi(f_3 - f_1)t)$ as opposed to $\cos(2\pi(f_2 - f_1)t)$. Modulation by the f_3-f_1 frequency cosine centers the triangular spectrum at f_1. Now when the bandpass filter is applied only the spectral components of the information signal of interest (the triangle) are kept. The steps necessary to extract the triangular spectrum are illustrated in Figure 3.

The receiver just discussed is known as the *superheterodyne receiver*. It was developed by Edwin Armstrong in 1918. The main advantage of the superheterodyne receiver is that it allows the majority of the processing blocks of a radio receiver (blocks such as signal amplifiers not illustrated in Figure 2) to be designed and operated at a narrow, low frequency range ([f_1-w_1 f_1+w_1] in Figure 2) as opposed to operating in a wide, high frequency range ([f_2-w_1 f_3+w_2] in Figure 2). Designing high performance processing blocks that operate in a narrow, low frequency range is far easier than designing processing blocks to operate in a wide, high frequency range.

In summary, the key step in receiving an information signal from a frequency division multiplexed transmission signal is selection of the modulation frequency at the receiver that will center the spectrum of the information signal of interest in the passband of the bandpass filter.

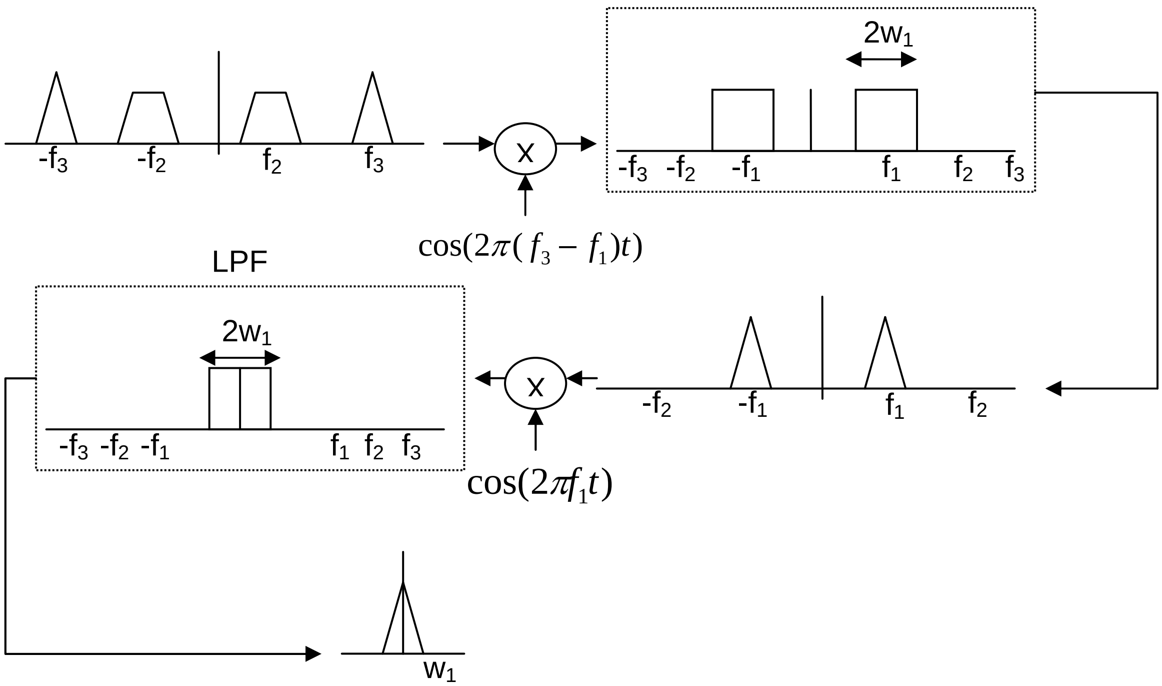


Figure 3

Building a FDM Receiver

In this lab a transmission signal carrying the spectra of three different voice signals is being broadcast. The voice signal spectra each have a bandwidth of 4kHz, and they are centered at f_1=30kHz, f_2=70kHz, and f_3=90kHz respectively. The spectrum of the transmission signal is illustrated in Figure 4. Your task is to extract and listen to each of the voice signals embedded in the transmission signal of Figure 4; each voice signal is a word. What is the word formed by concatenating the first letter of the words in Signal 1,2,3 in that order?

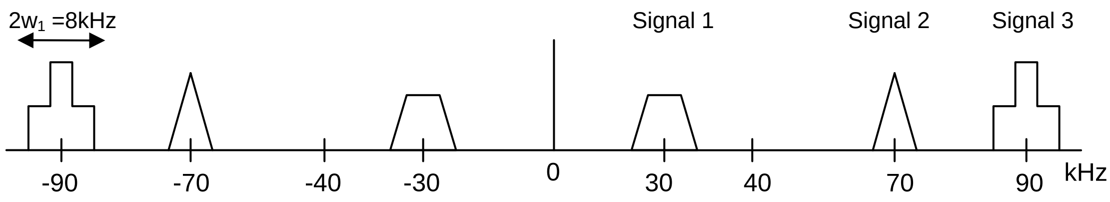


Figure 4

To complete this task, you will fill the blanks in the file "receive_FDM_SuperHet.m" This file is an implementation of the FDM receiving scheme discussed in the introduction and illustrated more compactly in Figure 5. In Figure 5 f_k for $k=1, 2, 3$ refers to the center frequency of voice signal k (e.g. f_3 = 90 kHz and is the center frequency of voice signal 3). Also note that in Figure 5 a correction term Δf has been added to the first modulator. The purpose of this correction term is to allow YOUR receiver to compensate for a frequency offset introduced by the transmitter; that is to allow your receiver to recognize that signal 1, 2, and 3 may not actually be at 30 kHz, 70 kHz, and 90 kHz but rather at 30 kHz + Δf , 70 kHz + Δf, and 90 kHz + Δf. A frequency offset can exist between the transmitter and receiver because the hardware components used to implement the modulators and demodulators in Figures 1 and 2 are not exact; that is both the transmitter and receiver modulators/demodulators operate at frequencies that are different from those we specify. The correction term Δf is automatically computed for you in the file "receive_FDM_SuperHet.m".

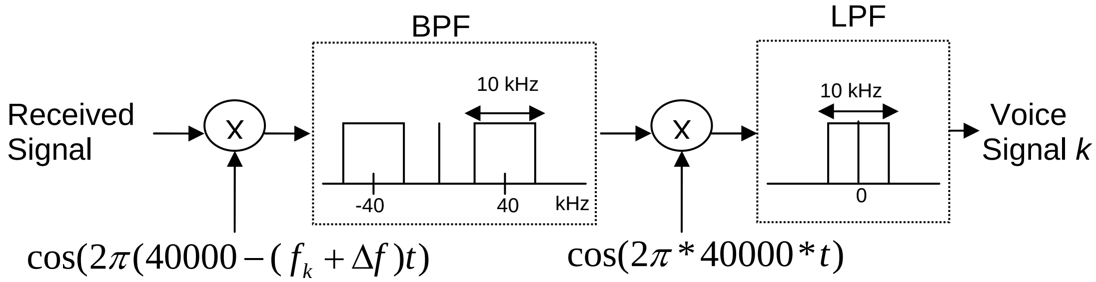


Figure 5

The file "receive_FDM_SuperHet.m" includes comments and blanks (marked using the symbol "??") where you should insert values and commands in order to implement the block diagram in Figure 5. The contents of the file "receive_FDM_SuperHet.m" are shown below; commands are in bold and comments are preceded by the symbol "%". Once you have filled all the blanks, listen to each of the voice signals in the FDM transmission signal and identify the word being transmitted.

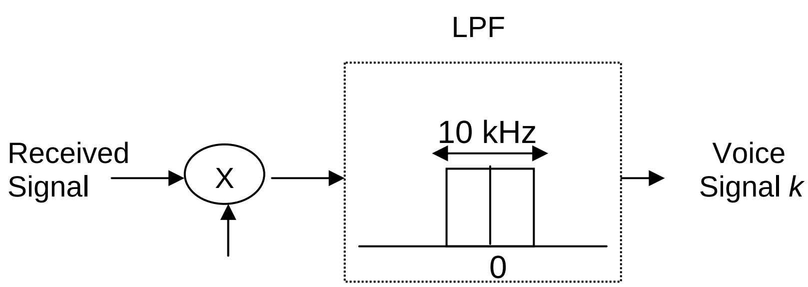
```
%Specify Sampling Frequency equal to 250kHz
Fs = 250e3;

%Specify center frequency (in FDM transmission signal) of voice signal of interest
fsignal = ???;                    % Can be 30kHz, 70kHz, or 90kHz
N = 3e5;                          % N is the number of samples we will capture from the signal

%Make the bandpass filter
Filter_Order = ???;
Cuttoff_Frequency_Low  = ???*(2/Fs);
Cuttoff_Frequency_High = ???*(2/Fs);
B_Coeffecients_BPF = ???;

%Make the low pass filter
Filter_Order = ???;
Cuttoff_Frequency = ???*(2/Fs);
B_Coeffecients_LPF = ??? ;

%Specification of RF hardware gains, oscillator frequencies, pilot tone
vco_freq = 452e6; dco_freq = 5.04e6;   adc_gain = 0;  mixer_gain = -10; f_tone = 110e3;

%Receiving Loop
for i = 1:20

   %Receive from the hardware the transmission into variable rx_a
   [rx_a,rx_b] = rf_receive(vco_freq,dco_freq,adc_gain,mixer_gain,N);

   %Automatic computation of frequency offset
   delta_f = frequency_offset(rx_a, N, Fs, f_tone);

    %Make the cosine to modulate the voice signal to 40kHz
    t = 0:1/Fs:(N-1)/Fs;
   f1 = ????;
   f2 = ????;
   modulation_Signal_1 = cos(???*t);
```

```
    %Modulate the received signal to the fixed bandpass filter
    rx_demod = rx_a .* ????;

    %Apply the bandpass filter
    rx_bpf =????;

    %Make the cosine to modulate the bandpass filtered signal to DC
    f3 = ???;
    modulation_Signal_2 = cos(???*t);

    %Modulate the bandpass filtered signal to DC
    rx_dc = ??? .* ????;

    %5)Apply the low pass filter to remove higher frequency demodulation products
    rx_dc_lpf = ????;

    %6)Listen to the recieved signal
    soundsc(rx_dc_lpf, Fs);

end
```

Save your file and ask for TA check off ____TA

Another FDM Receiver Architecture

In this section of the lab we will use an alternate FDM receiver architecture shown in Figure 6 to extract each of the voice signals in Figure 4. To receive voice signal *k* using this architecture, one chooses the demodulator frequency so that it centers the spectrum of voice signal *k* at DC. Next, the demodulated signal is low pass filtered so that only the spectrum of voice signal *k* appears at the output. The receiver architecture outlined in Figure 6 is known as the *homodyne receiver*. It was developed by a team of British scientists in 1932. The main advantage of the homodyne receiver is its low complexity (easier and cheaper to build) and low power consumption.



$$\cos(2\pi(f_k + \Delta f)t)$$

Figure 6

Edit your superheterodyne file to build the homodyne receiver shown in Figure 6. Once again you must include the correction term Δf when specifying the frequency of the homodyne receiver demodulator; the correction Δf is automatically computed for you. Listen to each of the voice signals in the FDM transmission signal and identify the word being transmitted.

Save your file and ask for TA check off ____TA

Investigate Effect of Frequency Mismatch

Add a known frequency mismatch between your receiver and the transmitter by adding to the modulation signal and additional offset x = ±500 Hz and ±1000 Hz and listen to the results. Now you see the reason we need frequency matching between the transmitter and receiver.

$$\cos(2\pi(f_k + \Delta f + x)t)$$

Save your file and ask for TA check off ____TA