EE-232 Semester Project

Task 1:

Frequency Division Multiplexing:

Frequency division multiplexing (FDM) is a technique of multiplexing which means combining more than one signal over a shared medium. In FDM, signals of different frequencies are combined for concurrent transmission.

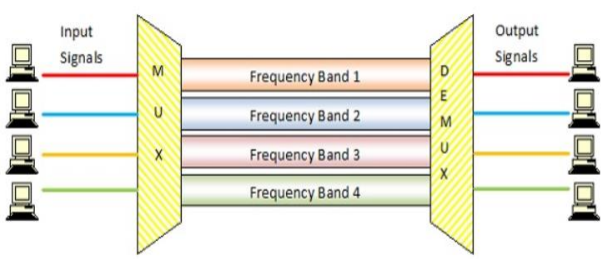
Concept and Process:

In FDM, the total bandwidth is divided to a set of frequency bands that do not overlap. Each of these bands is a carrier of a different signal that is generated and modulated by one of the sending devices. The frequency bands are separated from one another by strips of unused frequencies called the guard bands, to prevent overlapping of signals.

The modulated signals are combined together using a multiplexer (MUX) in the sending end. The combined signal is transmitted over the communication channel, thus allowing multiple independent data streams to be transmitted simultaneously. At the receiving end, the individual signals are extracted from the combined signal by the process of demultiplexing (DEMUX).

Example:

The following diagram conceptually represents multiplexing using FDM. It has 4 frequency bands, each of which can carry signal from 1 sender to 1 receiver. Each of the 4 senders is allocated a frequency band. The four frequency bands are multiplexed and sent via the communication channel. At the receiving end, a demultiplexer regenerates the original four signals as outputs.



Here, if the frequency bands are of 150 KHz bandwidth separated by 10KHz guard bands, then the capacity of the communication channel should be at least 630 KHz (channels: 150 × 4 + guard bands: 10 × 3).

Uses and Applications:

It allows sharing of a single transmission medium like a copper cable or a fiber optic cable, among multiple independent signals generated by multiple users.

FDM has been popularly used to multiplex calls in telephone networks. It can also be used in cellular networks, wireless networks and for satellite communications.

Perform the following tasks in order to do FDM:

1) Take 3-4 input sound signals from user. These should be 3-4 recorded audio signals from different members of your team. Each recorded audio should have a duration of about 10 seconds.
2) Design a low pass filter for 3kHz. (Can use FDA tool to design LPF)
3) Pass each signal separately through it.
4) Modulate each of the filtered signal by multiplying with a unique Cosine function using frequencies 3kHz, 9 kHz, 15 kHz and 21 kHz respectively.
5) Add the four modulated signals. Plot spectrum of their sum.
6) Design four band-pass filters for 3kHz, 9 kHz, 15 kHz and 21 kHz. Use appropriate bandwidth of the filters.
7) Pass the SUM through each band-pass to get respective signals (impulse response, two sided).
8) Again multiply the signals obtained from the bandpass filters with cosine at respective frequencies and pass through Low-pass filter to retrieve original signals.

Task 2: Audio Compression in MATLAB

Use one of the audio file from any member of the group. You are required to compress this audio signal and then reconstruct the audio signal using the compressed signal. The idea is to process the given signal in frequency domain. Save the significant frequency components (which have high energy) and ignore the ones with low energy. This will be the compressed signal.

Perform the following tasks:

1. Read the audio file into MATLAB. What is the sampling frequency F_s of the audio signal?
2. What is the length of the audio signal? Plot the signal. Label the axes appropriately.
3. Obtain the frequency spectrum of the audio signal. Plot this frequency spectrum against frequency in Hz and label the axes properly.
4. Select the first L significant frequency components. Make a vector with the significant frequency components of the signal. This will be the compressed signal. Note that this signal is in the frequency domain.
5. Reconstruct the audio signal in time using the compressed signal (which is currently in frequency domain). Plot this reconstructed frequency spectrum against frequency in Hz and label the axes properly.
6. Using this reconstructed frequency spectrum, obtain the audio signal in time domain. Plot the reconstructed audio signal. Label the axes appropriately.
7. You may also play the original and reconstructed audio. Is the reconstructed audio signal understandable?
8. Calculate the compression ratio as

   Compression ratio = $100 - \frac{length\ of\ significant\ frequency\ components}{length\ of\ original\ signal}(100)$
9. Propose and calculate some metric to quantify signal distortion in the reconstructed signal.
10. By varying the value of L, calculate and plot signal distortion. Comment on the observed trend? Specify the underlying cause of the observed trend.

Submission Guidelines:

1. Each group should comprise of 3-4 members. Groups with more than 4 members are not acceptable.

2. Submission on LMS should be 1 per group in the form of a zipped folder.

3. The zipped folder should contain 2 folders, with the titles “Task 1” and “Task 2”.

4. Each folder should contain the following:

    i): Matlab file with source code

    ii): Report which includes screenshots of results of each step in the task and participation of group members at the end of the report.

    iii): Sound signals that are used in the tasks.

5. Evaluation would be done individually on the performance of each group member in the project. The evaluation will include 8-10 minutes presentation per group with each member presenting a part of the work. The presentation will be followed by a viva on the project as well as associated topics from the course. More instructions on the presentation will be given later.