Cairo University

Faculty of Engineering

Dept. of Electronics and Electrical Communications

Second Year

**Signal Project**

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**Part1: Image filtering and restoration**

1. In the beginning, there will be a recording of sound, and the sound will contain noise, which is a very high frequency in the range of 5000 to 10,000 hertz, audio recording object with a sampling rate of 44100 Hz, a bit depth of 16 bits per sample, and a single audio channel. These parameters are commonly chosen based on standard audio recording practices.
2. I use function called **getaudiodata** to extract audio data from a recorded audio, this function retrieves the recorded audio samples as a column vector.
3. I use **audiowrite** function to store the recorded audio.
4. We generate waveform plot of audio signal; this type of plot visualizes the changes in the amplitude of the audio signal over time.
5. perform some analysis on two audio signals

L1 and L2 store the lengths of the audio signals, k1 and k2 store indices for the audio signals.

Performing FFT (Fast Fourier Transform)

**Plotting results:**

Three separate plots are created:

The first plot displays the magnitude of the FFT of ad against the indices.

The second plot visualizes the magnitude against frequency for ad.

The third plot shows the shifted FFT of ad.

1. We designed a Low pass filter with Fpass = 4000 and Fstop =5000 based on the recorded voice and wanted range of frequency; we loaded the stored low pass filter from .mat file
2. We applied low pass filter to the two recorders and repeat step 5 to the filtered audio
3. **Modulation:** setting the modulation parameters to be 5000,1500 to avoid over write, t1&t2 are time vectors created for modulation based on the lengths of filtered audios.

We modulated the signals with cosine functions at the specified carrier frequency, then calculate Fast Fourier Transform to the modulated signal.

1. trans1 and trans2 perform frequency shifting on the FFT results, f is a frequency vector, fun represents the sum of the magnitudes of the shifted FFTs for both modulated signals, then plots the frequency spectrum (f) against the combined magnitude spectrum (fun), The plot appears to show the frequency spectrum of the sum of the modulated signals. The frequencies are shifted using FFT, and the result is visualized in the plot.

The first pair of subplots shows the modulated signals in the time domain

The second pair of subplots displays the magnitude spectrum of each modulated signal in the frequency domain.

1. **Demodulation:**  are obtained by multiplying the modulated signals with a cosine function at the carrier frequencies, this process effectively shifts the modulated signals back to the baseband.
2. The low-pass filter is applied to the demodulated signals using the filter function, the factor of 2 is applied to scale the amplitude of the demodulated signals.

This sequence of demodulation and low-pass filtering is a common process in analog communication systems to retrieve the original baseband signal, the low-pass filter is used to remove high-frequency components introduced during modulation, leaving the demodulated audio signals.

**Part2: Communication system simulation**