ECE 09.351
DIGITAL SIGNAL PROCESSING – LAB 04
<u>DISCRETE TIME FOURIER TRANSFORM</u>
DUE FEB 27, 2009

In this lab, we will look at the computation and various properties of DTFT. First recall the following:
A system can be characterized either by its impulse response h[n], or its CCLDE coefficients, a_n and b_n, also called the denominator and numerator coefficients, respectively. The (discrete time) Fourier transform of the impulse response of a discrete system, H(ω), is then its frequency response. Matlab does not compute DTFT directly, it only computes DFT (which we will see later). However, one can approximate the DTFT in Matlab, either by directly computing it from its definition, or by using the **freqz()** function, we have seen in Lab 2. Recall that

[H, f]=freqz(b,a,N,Fs); Plots the frequency response of the discrete filter that is characterized by the CCLDE coefficients **a** and **b**. **Fs** is the sampling frequency of the signal to be filtered, **H** is the complex frequency response computed over **N** points, and **f** is a frequency base computed based on the sampling frequency. If ***Fs*** is not used, the output frequency base will be in angular frequency, not in Hz.

If using the freqz() function, you must choose N sufficiently large to simulate a continuous frequency domain to compute DTFT. Some other useful functions are:

abs(): computes the absolute value of its argument, if the argument is complex, computes magnitude
angle():computes the angle of its (complex) argument, can be used to compute the phase response
unwrap(): unwraps the phase response so that a suitable –π to π phase reponse can be computed.
fftshift(): flips its argument around the middle value. Convert the magnitude spectrum from [0 2π] to [-π π] range.
Lab:

1. Using the **freqz()** function as a starting point, write a short self-contained program that accepts numerator and denominator coefficients, the number of coefficients desired and the sampling frequency to compute DTFT and plots its real, imaginary parts as well as the magnitude and phase responses in the [-π π] range, as well as in the linear frequency range. Recall that the [-π π] range corresponds to [$-f_s/2$ $f_s/2$] range in linear frequency.
2. Using the above written function, compute and plot the frequency response of the following system. Based on your plots, comment on what kind of system this is (lowpass, highpass, etc.).

$$y[n]+2.37y[n-1]+2.7y[n-2]+1.6y[n-3]+0.41y[n-4]=$$
$$0.008x[n]-0.033x[n-1]+0.05x[n-2]-0.033x[n-3]+0.008x[n-4]$$

3.
    a. Write a program that computes the DTFT and its inverse from its definition. Check your code on the above sequences to ensure that it works correctly. Your code should also plot its magnitude and phase responses in the [-π π] range.
    b. Repeat part (a), however without using any “for” loops in your code! Once the auxiliary sections of the code is written (preparing the time, frequency axes, measure length of the signal, etc.), both the DTFT and reconstruction should be computed in one line!!

4. Using Matlab, verify the following properties of DTFT on sample signals of your choice.
    a. Linearity
    b. Time-shifting
    c. Frequency-shifting
    d. Convolution
    e. Parseval’s relation
5. Through several experiments, determine the “frequency resolution” of the DTFT, which is the width of the main lobe in the spectrum. You should find an expression that is a function of the signal length.