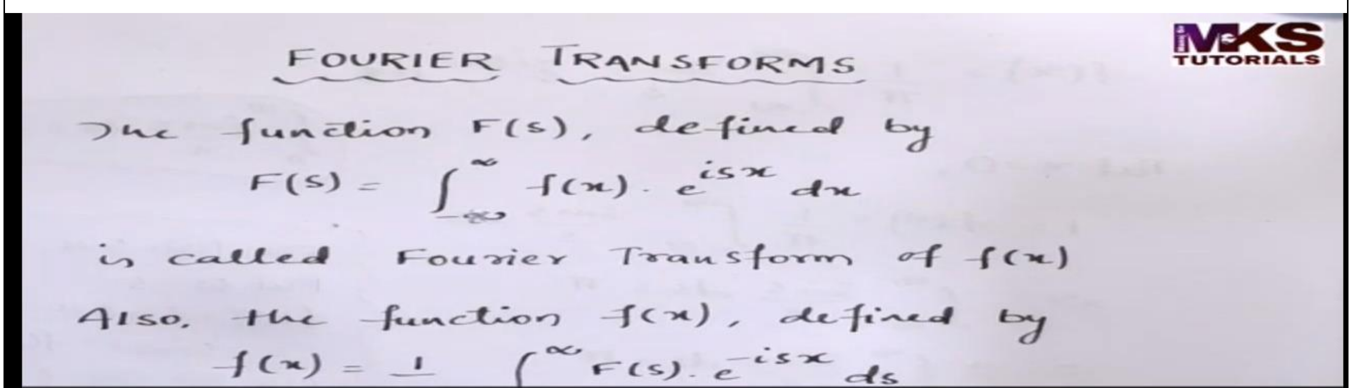
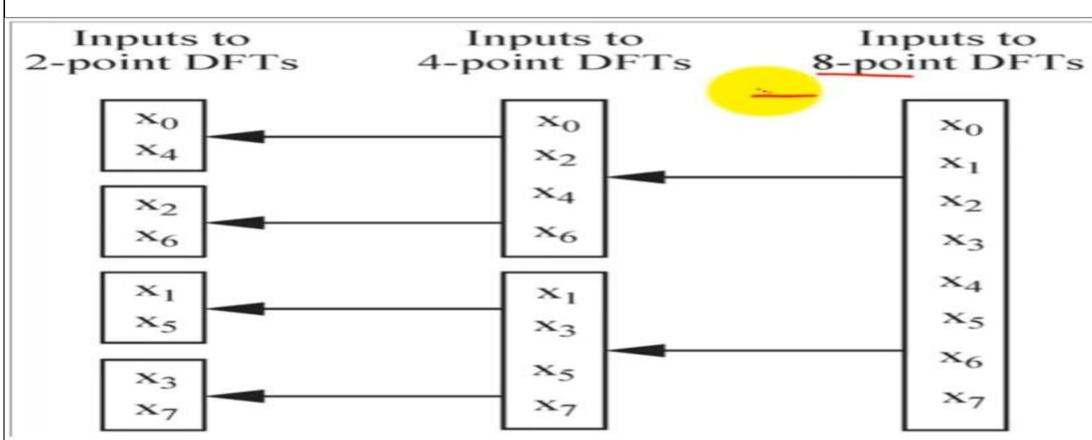
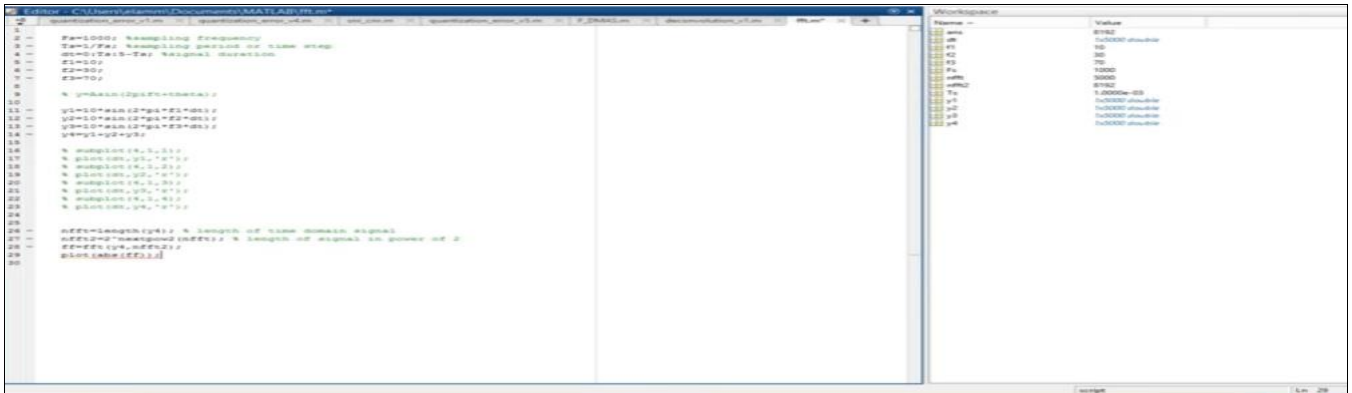
DAILY ASSESSMENT FORMAT

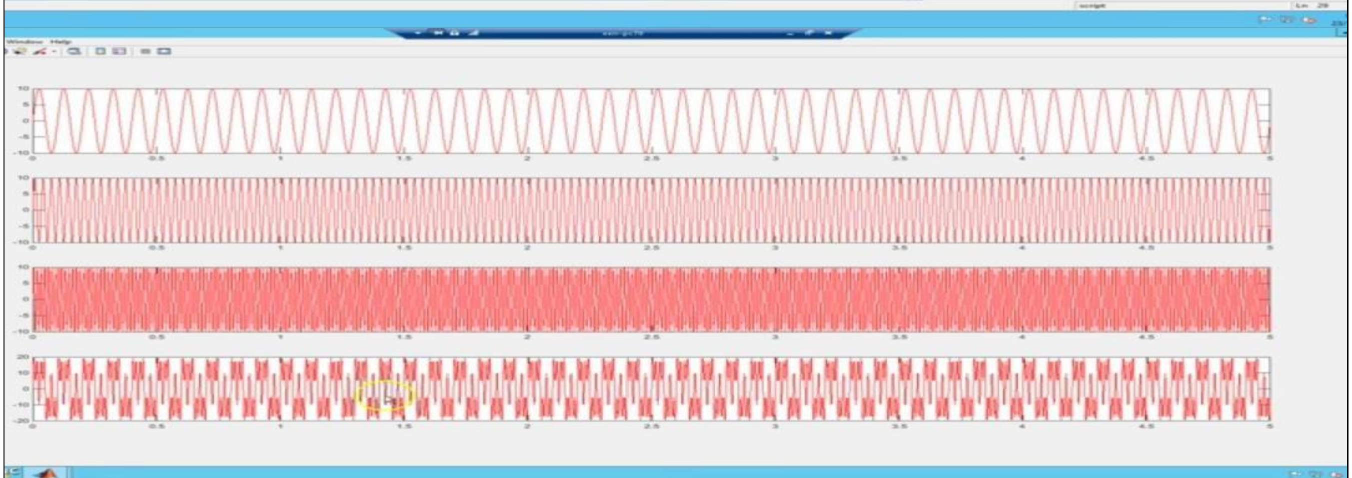
Date:	27-05-2020	Name:	Sahana S R
Course:	Digital signal processing	USN:	4al17ec083
Topic:	•Fourier Transform •FFT •FFT Fast Fourier transform matlab •FIR and IIR Filter •Study and analysis FIR and IIR using FDA tool in matlab •Introductions to WT •CWT and DWT •Implementation of signal filtering signal using WT in matlab •Short time Fourier transform and spectogram •Welch’s method and windowing • ECG signal analysis using matlab	Semester & Section:	6^{th} sem B sec
Github Repository:	sahanasr-course		

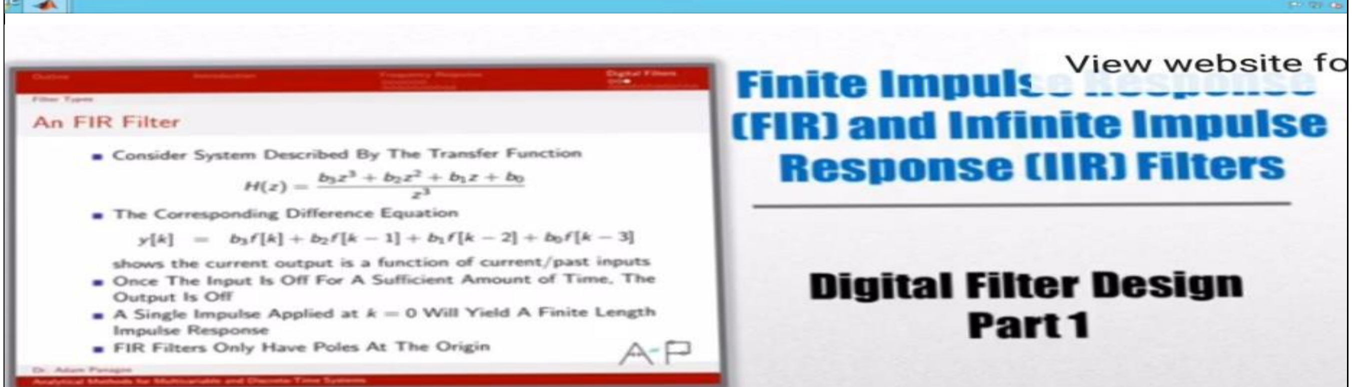
FORENOON SESSION DETAILS
Image of session



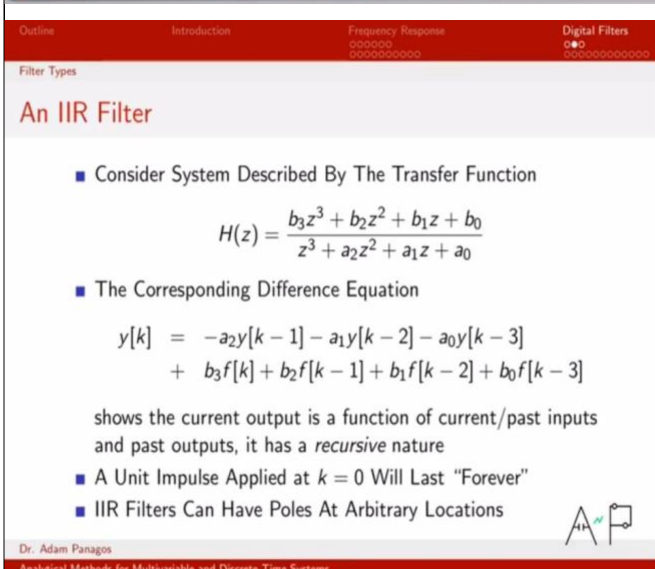




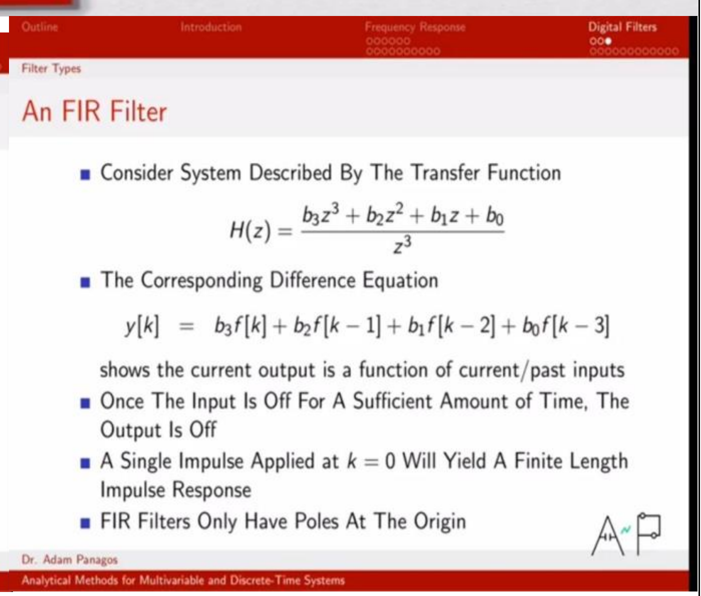




An FIR Filter
Consider System Described By The Transfer Function
$H(z) = \frac{b_3z^3 + b_2z^2 + b_1z + b_0}{z^3}$
The Corresponding Difference Equation
$y[k] = b_3f[k] + b_2f[k-1] + b_1f[k-2] + b_0f[k-3]$
shows the current output is a function of current/past inputs
Once The Input Is Off For A Sufficient Amount of Time, The Output Is Off
A Single Impulse Applied at $k = 0$ Will Yield A Finite Length Impulse Response
FIR Filters Only Have Poles At The Origin
View website fo
Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) Filters
Digital Filter Design
Part 1



Outline
Introduction
Frequency Response
Digital Filters
Filter Types
An IIR Filter
Consider System Described By The Transfer Function
$H(z) = \frac{b_3z^3 + b_2z^2 + b_1z + b_0}{z^3 + a_2z^2 + a_1z + a_0}$
The Corresponding Difference Equation
$y[k] = -a_2y[k-1] - a_1y[k-2] - a_0y[k-3] + b_3f[k] + b_2f[k-1] + b_1f[k-2] + b_0f[k-3]$
shows the current output is a function of current/past inputs and past outputs, it has a *recursive* nature
A Unit Impulse Applied at $k = 0$ Will Last "Forever"
IIR Filters Can Have Poles At Arbitrary Locations
Dr. Adam Panagos
Analytical Methods for Multivariable and Discrete-Time Systems



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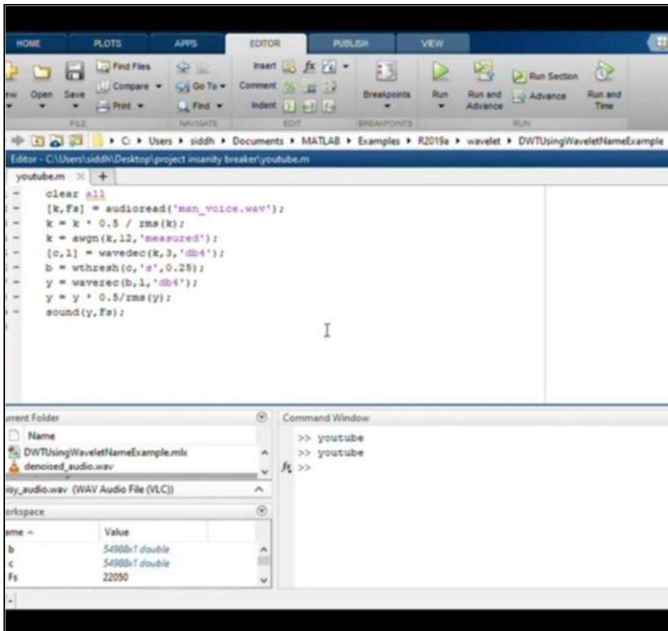
Analytical Methods for Multivariable and Discrete-Time Systems


Editor - C:\Users\siddh\Desktop\project insanity breaker\youtube.m
youtube.m
clear all
[k,Fs] = audioread('man_voice.wav');
k = k * 0.5 / rms(k);
k = awgn(k,12,'measured');
[c,l] = wavedec(k,3,'db4');
b = wthresh(c,'s',0.25);
y = waverec(b,l,'db4');
y = y * 0.5/rms(y);
sound(y,Fs);
Command Window
>> youtube
>> youtube
Name
DWTUsingWaveletNameExample.mlx
denoised_audio.wav
b 54988x1 double
c 54988x1 double
Fs 22050

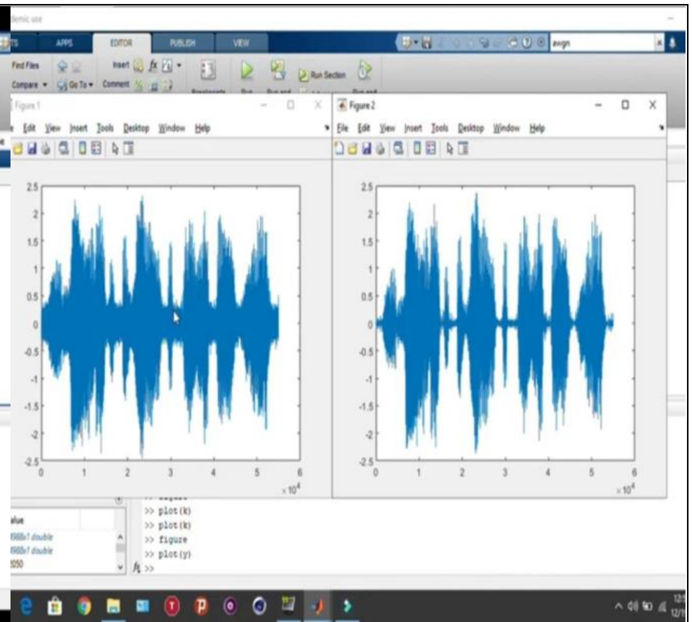
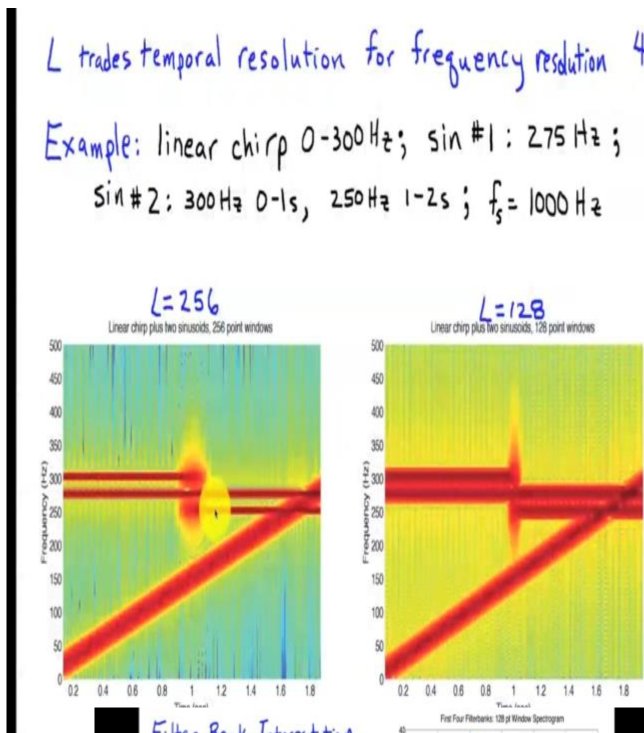
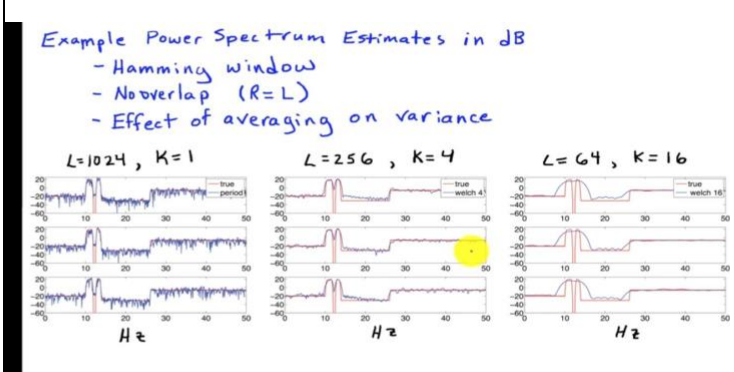
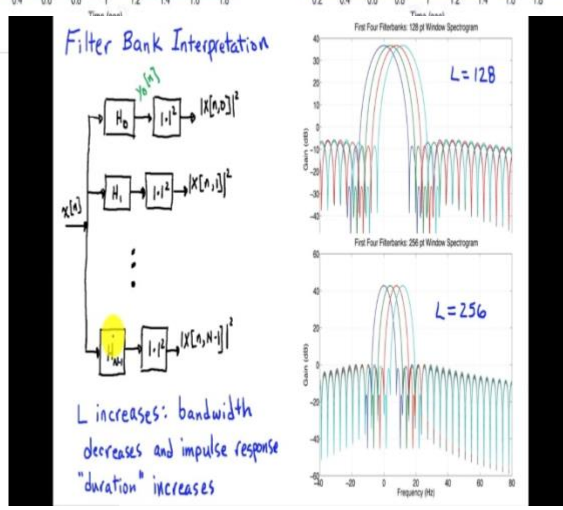


Figure 1
Figure 2
>> plot(k)
>> plot(k)
>> figure
>> plot(y)

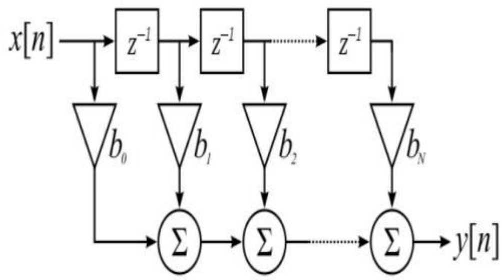

The Short-Time Fourier Transform
and the Spectrogram


L trades temporal resolution for frequency resolution 4
Example: linear chirp 0-300 Hz; sin #1 : 275 Hz;
sin #2: 300 Hz 0-1s, 250 Hz 1-2s; f_s = 1000 Hz
L=256
Linear chirp plus two sinusoids, 256 point windows
L=128
Linear chirp plus two sinusoids, 128 point windows
Frequency (Hz)

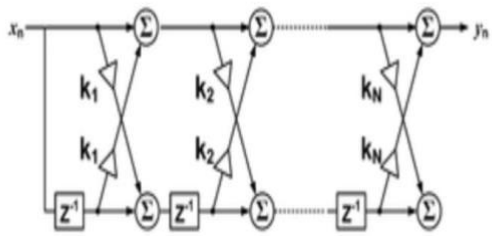

Example Power Spectrum Estimates in dB
- Hamming window
- No overlap (R=L)
- Effect of averaging on variance
L=1024, K=1
L=256, K=4
L=64, K=16
Hz
Hz
Hz


Filter Bank Interpretation
x[n]
H_0
$|X[n,0]|^2$
H_1
$|X[n,1]|^2$
$|X[n,N-1]|^2$
L=128
L=256
L increases: bandwidth decreases and impulse response "duration" increases

Report – Report can be typed or hand written for up to two pages.



A direct form discrete-time FIR filter of order N. The top part is an N-stage delay line with $N+1$ taps. Each unit delay is a z^{-1} operator in Z-transform notation.



A lattice-form discrete-time FIR filter of order N. Each unit delay is a z^{-1} operator in Z-transform notation.

For a causal discrete-time FIR filter of order N, each value of the output sequence is a weighted sum of the most recent input values:

$$y[n] = b_0 x[n] + b_1 x[n-1] + \cdots + b_N$$
$$= \sum_{i=0}^{N} b_i \cdot x[n-i],$$

A function $\psi \in L^2(\mathbb{R})$ is called an **orthonormal wavelet** if it can be used to define a Hilbert basis, that is a complete orthonormal system, for the Hilbert space $L^2(\mathbb{R})$ of square integrable functions.

The Hilbert basis is constructed as the family of functions $\{\psi_{jk} : j, k \in \mathbb{Z}\}$ by means of dyadic translations and dilations of ψ,

$$\psi_{jk}(x) = 2^{\frac{j}{2}} \psi\left(2^j x - k\right)$$

for integers $j, k \in \mathbb{Z}$.

If under the standard inner product on $L^2(\mathbb{R})$,

$$\langle f, g \rangle = \int_{-\infty}^{\infty} f(x)\overline{g(x)}dx$$

this family is orthonormal, it is an orthonormal system:

$$\langle \psi_{jk}, \psi_{lm} \rangle = \int_{-\infty}^{\infty} \psi_{jk}(x)\overline{\psi_{lm}(x)}dx$$
$$= \delta_{jl}\delta_{km}$$

where δ_{jl} is the Kronecker delta.