EC6502 PRINCIPLES OF DIGITAL SIGNAL PROCESSING
UNIVERSITY – TWO MARK QUESTIONS & ANSWERS

UNIT I

1.What is the relationship between Fourier transform and DFT ? (Apr.2018)

DFT: It is obtained by performing sampling operation in both the time and frequency domains. It is discrete frequency spectrum.

Fourier transform: Sampling is performed in time domain only. It is a continuous function of frequency.

2.What is a twiddle factor? (Nov.2017)

A twiddle factor, in fast Fourier transform (FFT) algorithms, is any of the trigonometric constant coefficients that are multiplied by the data in the course of the algorithm.

$$W_N = e^{-j2\pi / N}$$

3.State an prove periodicity property of DFT.(Nov.2017)

If X(k) is N-point DFT of a finite duration sequence x(n), then

$x(n+N) = x(n)$ for all n

$X(k+N) = X(k)$ for all k

4.Calculate the 4-point DFT of the sequence x(n) = {1 0 -1 0} (Apr.2018)

$$X_k = \sum_{n=0}^{N-1} x_n e^{-j2\pi kn/N} \qquad k = 0,1,2,\ldots,N-1$$

$X(0) = x(0) + x(1) + x(2) + x(3) = 1 + 0 + -1 + 0 = 0$

$X(1) = x(0) + x(1)\, e^{-j\pi/2} + x(2)\, e^{-j\pi} + x(3)\, e^{-j3\pi/2} = 1 + 0 -1(-1) + 0 = 2$

$X(2) = x(0) + x(1)\, e^{-j\pi} + x(2)\, e^{-j2\pi} + x(3)\, e^{-j3\pi} = 1 + 0 -1(1) + 0 = 0$

$X(3) = x(0) + x(1)\, e^{-j3\pi/2} + x(2)\, e^{-j3\pi} + x(3)\, e^{-j9\pi/2} = 1 + 0 -1(-1) + 0 = 2$

$X(k) = \{ 0, 2, 0, 2\}$

5.What is the relation between DTFT and DFT? (Apr.2017)

DTFT output is continuous in time where as DFT output is Discrete in time.

DFT: It is obtained by performing sampling operation in both the time and frequency domains. It is discrete frequency spectrum.

DTFT: Sampling is performed in time domain only. It is a continuous function of frequency.

6.Compute the DFT of the sequence x(n) = { 1, -1, 1, -1}.(Apr.2017)

$$X_k = \sum_{n=0}^{N-1} x_n e^{-j2\pi kn/N} \qquad k = 0,1,2,\ldots,N-1$$

$X(0) = x(0) + x(1) + x(2) + x(3) = 1 + -1 + 1 + -1 = 0$

$X(1) = x(0) + x(1)\, e^{-j\pi/2} + x(2)\, e^{-j\pi} + x(3)\, e^{-j3\pi/2} = 1 + (-1)(-j) + 1(-1) + (-1)(j) = 0$

$X(2) = x(0) + x(1)\, e^{-j\pi} + x(2)\, e^{-j2\pi} + x(3)\, e^{-j3\pi} = 1 + (-1)(-1) + 1(1) + (-1)(-1) = 4$

$X(3) = x(0) + x(1)\, e^{-j3\pi/2} + x(2)\, e^{-j3\pi} + x(3)\, e^{-j9\pi/2} = 1 + (-1)(j) + 1(-1) + (-1)(-j) = 0$

$X(k) = \{ 0, 0, 4, 0\}$

7.***Compare Radix 2 DIT, DIF fft algorithms. (Nov.2016)***

Differences:

1) The input is bit reversed while the output is in natural order for DIT, whereas for DIF the output is bit reversed while the input is in natural order.

2) The DIF butterfly is slightly different from the DIT butterfly, the difference being that the complex multiplication takes place after the add-subtract operation in DIF.

Similarities:

Both algorithms require same number of operations to compute the DFT.Both algorithms can be done in place and both need to perform bit reversal at some place during the computation.

8.Test the causality and stability of y(n) = sin x(n). (Nov.2016)
If n = 0 , y(0) = sin x(0); If n = 1 , y(1) = sin x(1); If n = -1 , y(-1) = sin x(-1);
Output depends on present inputs only, so it is causal.
For infinite values of n, h(n) = infinity, so it is unstable.

9.Is h(n) = (-1/4)δ(n+1) + (1/2) δ(n) – (1/4)δ(n-1) is stable and causal? Justify.(Apr.2016)
If n = 0 , h(n) = (-1/4)δ(1) + (1/2) δ(0) – (1/4)δ(-1) = 1/2
If n = -1 , h(n) = (-1/4)δ(0) + (1/2) δ(-1) – (1/4)δ(-2) = -1/4
If n = 1 , h(n) = (-1/4)δ(2) + (1/2) δ(1) – (1/4)δ(0) = -1/4.
h(n) is not equal to zero for n < 0. so this is non-causal.
For all values of n, h(n)<infinity, so it is stable.

10.What is the smallest no. Of DFTs and IDFTs needed to compute the linear convolution of a length 50 sequence with a length of 800 sequence is to be computed using 64 pt DFT & IDFT? (Apr.2016)
Overlap-add method:
M = 50
L + M – 1 = 64
L = 15
No.of blocks = 800 / 15 =54
No.of DFTs 54 + 1 = 55
No..of IDFTs = 54

Overlap save method:
No.of data points lost = 49
Total length of sequence = 849
Each convolution results 15 correct values
Total No.of DFTs [849/15]+ 1 = 58
Total No..of IDFTs = 57

UNIT II

1.What is warping effect? (Apr.2018) (Apr.2016)
The relationship between analog and digital frequencies in bilinear transformation is given by Ω = ((2/T) tan (ω/2).
For smaller values of ω there exist linear relationship between ω and Ω.but for larger values of ω the relationship is nonlinear. This introduces distortion in the frequency axis. This effect compresses the magnitude and phase response. This effect is called warping effect.

2.Discuss the need for pre warping. (Apr.2017) (Nov.2016)
The effect of the non linear compression at high frequencies can be compensated. When the desired magnitude response is piecewise constant over frequency, this compression can be compensated by introducing a suitable rescaling or prewarping the critical frequencies.

3.State the condition for a digital filter to be causal and stable? (Apr.2017)
A digital filter is causal if its impluse response h(n)=0 for n<0.
A digital filter is stable if its impulse response is absolutely summable .

4.Why impulse invariant method is not preferred in the design of IIR filters other than low pass filter?(Apr.2016)
In this method the mapping from s plane to z plane is many to one. Thus there are an infinite number of poles that map to the same location in the z plane, producing an aliasing effect. It is inappropriate in designing high pass filters. Therefore this method is not much preferred.

5. What are the methods used for digitizing the analog filter into a digital filter ? (Apr.2018)
a) Impulse Invariance method
b) Bilinear Transformation method

c) Approximation of derivatives
d) Matched z-transform

6.List the different types of filters based on frequency response?(Nov.2017)
Based on frequency response the filters can be classified as
1. Low pass filter
2. High pass filter
3. Band pass filter
4. Band reject filter

7.What are the properties of bilinear transformation?(Nov.2017) (Nov.2016)
The mapping for the bilinear transformation is a one-to-one mapping that is for every point Z, there is exactly one corresponding point S, and vice-versa. The j Ω-axis maps on to the unit circle |z| =1,the left half of the s-plane maps to the interior of the unit circle |z|=1 and the half of the s-plane maps on to the exterior of the unit circle |z|=1.

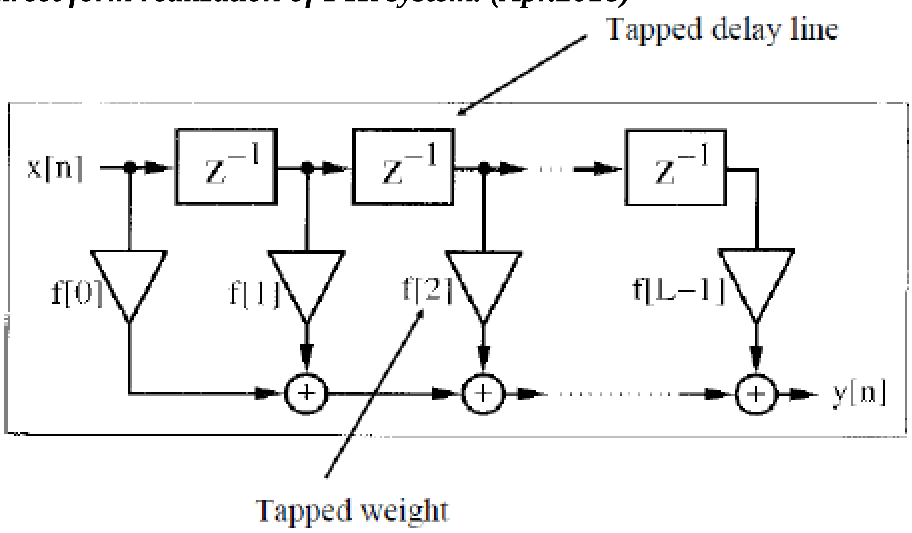
UNIT III

1.List out the advantages of FIR filter? (Apr.2016)
Linear phase FIR filter can be easily designed .
Efficient realization of FIR filter exists as both recursive and non-recursive structures.
FIR filter realized non-recursively stable.
The round off noise can be made small in non recursive realization of FIR filter.

2.What is Gibbs phenomenon? (Apr.2017)
One possible way of finding an FIR filter that approximates H(ejw)would be to truncate the infinite Fourier series at n= ± (N-1/2).Abrupt truncation of the series will lead to oscillation both in pass band and is stop band .This phenomenon is known as Gibbs phenomenon.

3.What are the desirable characteristics of the windows? (Nov.2016)
The desirable charaterstics of the window are
1.The central lobe of the frequency response of the window should contain most of the energy and should be narrow.
2.The highest side lobe level of the frequency response should be small.
3.The sides lobes of the frequency response should decrease in energy rapidly as w tends to p .

4. Draw the direct form realization of FIR system. (Apr.2018)



5. How the zeros in FIR filter is located ? (Apr.2018)

1.Z_1 is a real zero with $|z_1| < 1$. Then z_1^{-1} is also a real zero and there are two zeros in this group.
2. Z_2 = -1. Then, Z_2 -1 = Z_2 and this group contains only one zero.
3. Z_3 is a complex zero with $|Z_3|$ is not equal 1 then = Z_3^* and there are two zeros in this group.
4. Z_4 is a complex zero with $|Z_4|$ not equal 1. This group contains four zeros Z_4, Z_4^{-1}, Z_4^*, $(Z_4^*)^{-1}$.

6.Write the steps involved in FIR filter design?(Nov.2017)

Choose the desired frequency response Hd(w)
Take the inverse fourier transform and obtain Hd(n)
Convert the infinite duration sequence Hd(n) to h(n)
Take Z transform of h(n) to get H(Z)

Using windows:

1. For the desired frequency response Hd(w), find the impulse response hd(n) using Equation

$$hd(n)=1/2\pi\int_{-\pi}^{\pi} Hd(w)e^{jwn}dw$$

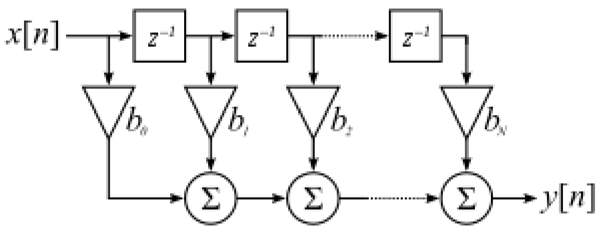
2. Multiply the infinite impulse response with a chosen window sequence w(n) of length N to obtain filter coefficients h(n),i.e.,

h(n)= hd(n)w(n) for |n|≤(N-1)/2
= 0 otherwise

3. Find the transfer function of the realizable filter

$$H(z)=z^{-(N-1)/2} [h(0)+\sum_{n=0}^{(N-1)/2} h(n)(z^{n}+z^{-n})]$$

7.Draw the block diagram representation of FIR system?(Nov.2017)



8.Compare Hamming window and Blackmann window. (Apr.2017)

Hamming window	***Blackmann window***
The main lobe width is 8π / N and the peak side lobe level is – 41 dB.	The main lobe width is 12π / N and the peak side lobe level is 57 dB.
The low pass FIR filter designed will have first side lobe peak of -53 dB.	The low pass FIR filter designed will have first side lobe peak of -74 dB.
$W_H(n)= 0.54 + 0.46\cos\frac{2\pi n}{N-1}$; $-(\frac{N-1}{2})$ to $(\frac{N-1}{2})$ = 0 ; other n Or $W_H(n)= 0.54 - 0.46\cos\frac{2\pi n}{N-1}$; 0≤n≤N-1 = 0 ; other n	$W_B(n)= 0.42 + 0.5\cos(\frac{2\pi n}{N-1}) + 0.08\cos(\frac{4\pi n}{N-1})$; $-(\frac{N-1}{2})$ to $(\frac{N-1}{2})$ = 0 ; other n Or $W_B(n)= 0.42 - 0.5\cos(\frac{2\pi n}{N-1}) + 0.08\cos(\frac{4\pi n}{N-1})$; 0≤n≤N-1 = 0 ; other n

9.What do you understand by linear phase response? (Nov.2016)

The phase function should be a linear function of w, which inturn requires constant group delay and phase delay.

UNIT IV

1.What is meant by "dead band" of the filter?(Apr.2017) (Apr.2016)

The limit cycle occurs as a result of quantization effect in multiplication. The amplitudes of the output during a limit cycle are confined to a range of values called the dead band of the filter.

2.what are the two kinds of limit cycle behavior in DSP? (Apr.2016)

1.zero input limit cycle oscillations
2.Overflow limit cycle oscillations

3. Distinguish between fixed point arithmetic and floating point arithmetic. (Apr.2018)(Nov.2017)

Fixed point arithmetic	**Floating point arithmetic**
1. Fast operation	1. Slow operation
2. Relatively economical	2. More expensive because of costlier hardware.
3. Small dynamic range	3. Increased dynamic range
4. Round off errors occur only in addition	4. Round off errors occur in both addition and multiplication.
5. Overflow occurs in addition	5. Overflow does not arise.
6. Used in small computers	6. Used in large, general purpose computers

4. Why is rounding preferred over truncation in realizing a digital filter ? (Apr.2018)

i) The quantization error due to rounding is independent of the type arithmetic
ii) The mean of rounding error is zero
iii) The variance of the rounding error signal is low

5.What are the methods used to prevent overflow? (Apr.2017)(Apr.2016)

There are two methods used to prevent overflow
1. Saturation arithmetic 2. Scaling

6.What is meant by finite word length effects in digital system?(Nov.2017)

1.Input quantization errors
2.Coefficient quantization errors
3.Product quantization errors
4.zero input limit cycle oscillations
5.Overflow limit cycle oscillations

7.What are the different types of fixed point representation? (Nov.2016)

Depending on the negative numbers are represented there are three forms of fixed point arithmetic. They are sign magnitude,1's complement,2's complement .

8.Name the three quantization error due to finite word length registers in digital filters. (Nov.2016)

1.Input quantization errors
2.Coefficient quantization errors
3.Product quantization errors

UNIT V

1. Show that the up sampler and down sampler are time invariant system. (Apr.2018)

Consider a factor of L upsampler defined by

y(n) = x(n/L)

The output due to delayed input is

y(n,k) = x($(n/L)-k$)

The delayed output is

y(n-k) = x((n-k)/L)

y(n,k) != y(n-k) so, is time variant system.

Similarly for down sampler y(n) = x(nM)

y(n-k) = x(M(n-k))

y(n,k) != y(n-k) so, is time variant system.

2.Write the expression for the output y(n) as a function of the input x(n) for the given multirate system as in Figure 1. (Apr.2018)

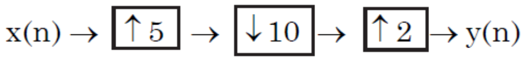
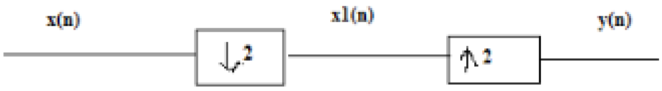


Figure 1

Solution:



x1(n) = x(2n) and y(n) =x1 (n/2) for n = 2k
= 0 otherwise

y(n) = x (n) for n = k
= 0 otherwise

3.Write the input output relationship for a decimator?(Nov.2017)

The output signal y(n) is a down sampled signal of the input signal x(n) and can be represented by y(n) = x(Mn).

4.State the applications of adaptive filtering.(Nov.2017)

Adaptive filters are used widely in communication systems, control systems and various other systems like adaptive antenna systems, digital communication receivers, adaptive noise cancelling and system modelling etc.

5.Define adaptive filtering. (Apr.2017) (Nov.2016)

An adaptive filter is a system with a linear filter that has a transfer function controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm. A filter with adjustable coefficients is called as adaptive filter.

6.List the applications of multirate signal processing. (Apr.2017)

Applications of multirate DSP are

a) Design of phase shifts
b) Interfacing of digital systems with different sampling rates
c) Implementation of digital filter banks
d) Subband coding of speech signals
e) Quadrature mirror filters (QMFs)
f) Transmultiplexers
g) Oversampling A/D and D/A conversion

7.What is the need for anti imaging filter after upsampling signal? (Nov.2016)

The frequency spectrum of upsampled signal with a factor L, contains (L-1) additional images of the input spectrum. Since we are not interested in image spectra, LPF with cut off frequency π/L can be used after upsampler. This filter is known as anti-imaging filter. The insertion of zeros effectively attenuates the signal by L, so the output of the anti-imaging filter must be multiplied by L, to maintain the same signal magnitude.

8.What is the need for antialiasing filter? (Apr.2016)

The spectra obtained after down sampling a signal by a factor M is the sum of all the uniformly shifted and stretched version of original spectrum scaled by a factor 1/M, then down sampling will cause aliasing. In order to avoid aliasing the signal x(n) is to be band limited to plus or minus π/M . This can be done by filtering the signal x(n) with a low pass filter with a cutoff frequency of π/M. This filter is known as antialiasing filter.

9.If the spectrum of a sequence x(n) is X(ejw), then what is the spectrum of the signal down sampled by 2? (Apr.2016)

Y(exp(jw))= 0.5[X(exp(jw/2))+X(exp(j(w/2-pi)))