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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

ACADEMIC YEAR : 2021 - 22 (ODD/EVEN SEMESTER)

COURSE FILE - CONTENT PAGE

YEAR & SEM	:	III / VI	**BATCH**	:	2019-2023
SUBJECT CODE	:	EC8562.	**SUBJECT NAME**	:	DSP. Lab.
REGULATION	:	R2017	**STAFF IN-CHARGE**	:	Mr. R. BALAKRISHNAN.

- Syllabus
- Course plan
- Student name list
- Individual time table
- Lab Manual
- Sample Observation notebook & Record
- Model Lab
  - Question paper
  - Sample answer sheet
  - Mark statement
- Content Beyond Syllabus
- Record of Internal Mark

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

SUBJECT: DIGITAL SIGNAL PROCESSING LABORATORY

SEMESTER: V

LAB MANUAL (EC8562)

(Version : 3)

PREPARED BY

Mr.R.BALAKRISHNAN,AP/ECE

EC 8562 DIGITAL SIGNAL PROCESSING LAB ORATORY

L T P C
0 0 4 2

SYLLABUS

(MATLAB / EQUIVALENT SOFTWARE PACKAGE)

1. Generation of elementary Discrete
2. Time sequences
3. Linear and Circular convolutions
4. Auto correlation and Cross Correlation
5. Frequency Analysis using DFT
6. Design of FIR filters (LPF/HPF/BPF/BSF) and demonstrates the filtering operation
7. Design of Butterworth and Chebyshev IIR filters (LPF/HPF/BPF/BSF) and
8. demonstrate the filtering operations

(DSP PROCESSOR BASED IMPLEMENTATION)

9. Study of architecture of Digital Signal Processor
10. Perform MAC operation using various addressing modes
11. Generation of various signals and random noise
12. Design and demonstration of FIR Filter for Low pass, High pass, Band pass and Band stop filtering
13. Design and demonstration of Butter worth and Chebyshev IIR Filters for Low pass, High pass, Band pass and Band stop filtering
14. Implement an Up sampling and Down sampling operation in DSP Processor

TOTAL: 60 PERIODS

6/8/21

SIGNATURE OF STAFF INCHARGE

6/9/21

HOD/ECE

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

COURSE PLAN

Sub. Code	: EC8562	**Branch / Year / Sem**	: B.E ECE / III /V
Sub. Name	: Digital Signal Processing Laboratory	**Batch**	: 2019-2023
Staff Name	: Mr.R.Balakrishanan	**Academic Year**	: 2021-22 (ODD)

COURSE OBJECTIVE

1. To perform basic signal processing operations such as Linear Convolution, Circular Convolution, Auto Correlation, Cross Correlation and Frequency analysis in MATLAB
2. To implement FIR and IIR filters in MATLAB and DSP Processor
3. To study the architecture of DSP processor
4. To design a DSP system to demonstrate the Multi-rate and Adaptive signal processing concepts.

LEARNING OUTCOME

At the end of the course, the student should be able to

- Carryout basic signal processing operations
- Demonstrate their abilities towards MATLAB based implementation of various DSP systems
- Analyze the architecture of a DSP Processor
- Design and Implement the FIR and IIR Filters in DSP Processor for performing
- filtering operation over real time signals
- Design a DSP system for various applications of DSP

PRE-REQUISITE

Programming in C and MATLAB

EQUIPMENTS / COMPONENTS / SOFTWARE REQUIRMENT

Hardware:

- PCs with Fixed / Floating point DSP Processors (Kit / Add-on Cards) 15 Units.

Software:

- MATLAB with Simulink and Signal Processing Tool Box or Equivalent Software in desktop systems -15 Nos
- Signal Generators (1MHz) – 15 Nos
- CRO (20MHz) -15 Nos

Ex.No	Planned Date	Title of the Experiment	No. of Hrs. required	Cumulative No. of periods
CYCLE : I MATLAB / EQUIVALENT SOFTWARE PACKAGE				
1.		Generation of elementary Discrete Time sequences	4	4
2.		Linear and Circular convolutions	4	8
3.		Auto correlation and Cross Correlation	4	12
4.		Frequency Analysis using DFT	4	16
5.		Design of FIR filters (LPF/HPF/BPF/BSF) and demonstrates the filtering operation	4	20
6.		Design of Butterworth and Chebyshev IIR filters (LPF/HPF/BPF/BSF) and demonstrate the filtering operations	8	28
CYCLE : II DSP PROCESSOR BASED IMPLEMENTATION				
7.		Study of architecture of Digital Signal Processor	4	32
8.		Perform MAC operation using various addressing modes	4	36
9.		Generation of various signals and random noise	4	40
10.		Design and demonstration of FIR Filter for Low pass, High pass, Band pass and Band stop filtering	4	44
11.		Design and demonstration of Butter worth and Chebyshev IIR Filters for Low pass, High pass, Band pass and Band stop filtering	8	52
12.		Implement an Up-sampling and Down-sampling operation in DSP Processor	8	60

CONTENT BEYOND SYLLABUS

1. Study of sampling theorem, effect of under sampling.
(Link:http://www.digital.iitkgp.ernet.in/dsp/expts/expt01/index3.php
2. Study of FIR filter design using window method: Low pass and high pass filter. (Link:http://www.digital.iitkgp.ernet.in/dsp/expts/expt08/)

INTERNAL ASSESSMENT DETAILS

MODEL	I
PORTIONS	CYCLE 1 & 2 EXPERIMENTS
Date	

6/9/21
Prepared by
Mr.R.BALAKRISHNAN

J. [illegible] 14/8/2021

6/9/21
Verified By
HOD/ECE

Approved by
PRINCIPAL

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

ACADEMIC YEAR 2021 – 2022 / ODD SEMESTER

CLASS: III ECE

HALL NO:ONLINE

CLASS COORDINATOR: Mrs.U.JEYAMALAR

STRENGTH: 42

ROLL NO.	REGISTER NUMBER	NAME OF THE STUDENT	ROLL NO.	REGISTER NUMBER	NAME OF THE STUDENT
01	821119106001	ABIMANEU S	22	821119106025	MADHUMITHA G
02	821119106002	AGALYA P	23	821119106026	MAHESWARI V
03	821119106004	BLESSON MANUEL J	24	821119106027	MATHIVANAN K
04	821119106005	DHARMADURAI A	25	821119106028	NITHITHA U
05	821119106006	DHARSHINI C	26	821119106029	NIVETHITHA S
06	821119106007	DURGA SRI R	27	821119106030	PAVITHRA P
07	821119106008	GANGA L	28	821119106031	PRAKASH A
08	821119106009	GANGA R	29	821119106032	PRETHIYA B
09	821119106010	GAYATHRI K	30	821119106033	PRIYANKA K
10	821119106011	GAYATHRI S	31	821119106034	RAMANA BHARATHI S
11	821119106012	ISHWARYA K	32	821119106035	RENUKA K
12	821119106013	JAYAKUMAR A	33	821119106036	RUTHRA R
13	821119106015	JOTHIKA R	34	821119106037	SABARINATHAN S
14	821119106016	KABILAN R	35	821119106039	SARASWATHI K
15	821119106017	KABISHENA P	36	821119106040	SATHYA G
16	821119106019	KARIKALAN G	37	821119106042	SHATHANA B
17	821119106020	KARTHICK N	38	821119106043	SOUNDHARYA R
18	821119106021	KARTHIKA DEVI M	39	821119106044	SURIYA C
19	821119106022	KIRUBADHARSHINI S	40	821119106045	SUSIKUMAR T
20	821119106023	KRISHNADEVI G	41	821119106046	SWETHAA S M
21	821119106024	LOGESHWARAN P	42	821119106048	VAISHNAVI G

Format: QF06

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING
FACULTY INDIVIDUAL TIME TABLE (AUG 2021 – NOV 2021, ODD SEM)
B.E – ECE (Regulation 2017) - With Effect from 18.08.2021

NAME OF THE FACULTY: Mr.R.Balakrishnan

Session	1	2		3		4	5
Day	09.50am - 10.50am	10.50am - 11.50am	11.50am - 12.05pm	11.45am - 12.45pm	12.45pm - 01.50pm	01.50pm - 02.50pm	02.50pm - 04.00pm
MON	EC8553		BREAK		LUNCH BREAK	EC8395	
TUE		EC8553					EC8395
WED		EC8395					
THU				EC8395			
FRI	EC8395			EC8553			
SAT		EC8553					

EC6... 2

EC8553 EC8562

Day/Time	5 (04.00PM -05.00PM)	7 (05.00PM -06.00PM)	8 (06.00PM -07.00PM)
MON – THU		EC8562	

CLASS/SEM	SUB CODE	NAME OF THE SUBJECT	PERIODS/WEEK
III ECE/V	EC8553	Discrete Time Signal Processing	5
II CSE/III	EC8395	Communication Engineering	5
III/V	EC8562	Digital Signal Processing Laboratory	8

DEPT. TTC

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

Lab Manual

Subject Title: DIGITAL SIGNAL PROCESSING LABORATORY

Subject Code: EC8562

Name : S.M. Swethaa

Reg. No. : 821119106046

Degree : **B.E.**

Branch : **ECE**

Year/ Sem : **III / V**

Academic Year: 2021 – 2022

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

SUBJECT: DIGITAL SIGNAL PROCESSING LABORATORY

SEMESTER: V

LAB MANUAL (EC8562)

(Version : 3)

PREPARED BY

Mr.R.BALAKRISHNAN,AP/ECE

EC 8562 **DIGITAL SIGNAL PROCESSING LAB ORATORY** **L T P C**
0 0 4 2

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14. Implement an Up sampling and Down sampling operation in DSP Processor

TOTAL: 60 PERIODS

6/8/21
SIGNATURE OF STAFF INCHARGE

HOD/ECE 6/8/21

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING
COURSE PLAN

Sub. Code	: EC8562	**Branch / Year / Sem**	: B.E ECE / III /V
Sub. Name	: Digital Signal Processing Laboratory	**Batch**	: 2019-2023
Staff Name	: Mr.R.Balakrishanan	**Academic Year**	: 2021-22 (ODD)

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1.	19.08.2021	Generation of elementary Discrete Time sequences	4	4
2.	23.08.2021 26.08.2021	Linear and Circular convolutions	4	8
3.	02.09.2021	Auto correlation and Cross Correlation	4	12
4.	06.09.2021	Frequency Analysis using DFT	4	16
5.	09.09.2021 13.09.2021	Design of FIR filters (LPF/HPF/BPF/BSF) and demonstrates the filtering operation	4	20
6.	16.09.2021 20.09.2021	Design of Butterworth and Chebyshev IIR filters (LPF/HPF/BPF/BSF) and demonstrate the filtering operations	8	28
CYCLE : II DSP PROCESSOR BASED IMPLEMENTATION				
7.	23.09.2021	Study of architecture of Digital Signal Processor	4	32
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CONTENT BEYOND SYLLABUS

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INTERNAL ASSESSMENT DETAILS

MODEL	I
PORTIONS	CYCLE 1 & 2 EXPERIMENTS
Date	

6/8/21
Prepared by
Mr.R.BALAKRISHNAN

14/8/2021
Approved by
PRINCIPAL

6/8/21
Verified By
HOD/ECE

CONTENTS

S.NO.	DATE OF EXPERIMENT	TITLE OF THE EXPERIMENT	PAGE	MARK (10)	SIGNATURE
		CYCLE 1(MATLAB / EQUIVALENT SOFTWARE PACKAGE)			
1.	19.08.2021	Generation of elementary Discrete Time sequences	06	10	
2.	23.08.2021 26.08.2021	Linear and Circular convolutions	11	10	
3.	02.09.2021	Auto correlation and Cross Correlation	15	10	
4.	06.09.2021	Frequency Analysis using DFT	18	10	
5.	09.09.2021 13.09.2021	Design of FIR filters (LPF/HPF/BPF/BSF) and demonstrates the filtering operation	21	10	
6.	16.09.2021 20.09.2021	Design of Butterworth and Chebyshev IIR filters (LPF/HPF/BPF/BSF) and demonstrate the filtering operations	25	10	
		CYCLE 2(DSP PROCESSOR BASED IMPLEMENTATION)			
7.	23.09.2021	Study of architecture of Digital Signal Processor	29	10	
8.	27.09.2021	Perform MAC operation using various addressing modes	35	10	
9.	30.09.2021 4.10.2021	Generation of various signals and random noise	38	10	
10.	7.10.2021 11.10.2021	Design and demonstration of FIR Filter for Low pass, High pass, Band pass and Band stop filtering	42	10	
11.	18.10.2021 21.10.2021	Design and demonstration of Butter worth and Chebyshev IIR Filters for Low pass, High pass, Band pass and Band stop filtering	81	10	
12.	25.10.2021 28.10.2021 01.11.2021	Implement an Up-sampling and Down-sampling operation in DSP Processor & CBS.	69	10	

7/12/21

SIGN OF STAFF INCHARGE

7/12/21

SIGN OF HOD

completed 7/12/21

USING MATLAB

Expt. No.: 1	GENERATION OF ELEMENTARY DISCRETE TIME SEQUENCES
Date: 19.08.2021	

AIM:

To generate elementary time sequences of unit impulse, unit step, delayed unit step Exponential Sine wave, Cosine wave, Ramp and Random function using MATLAB.

APPARATUS REQUIRED:

PC with MATLAB 9.0

FUNCTIONS USED:

plot (x,y);
stem (x,y)
supplot (2,1,1);

PROGRAM:

```
% GENERATION OF INPUT SIGNALS
clc;
% UNIT IMPLUSE
X=-10:1:10;
Y=((X-0)==0);
subplot(7,1,1);
stem(x,y);
xlabel('n');
ylabel('amp');
title('unit impluse');
% DELAYED UNIT IMPLUSE
X=-10:1:10;
Y=((X-4)==0);
subplot(7,1,3);
stem(x,y);
xlabel('n');
ylabel('amp');
title('delayed unit impluse');
% UNIT STEP
X=-10:1:10;
Y=((X-0)>=0);
subplot(7,1,5);
stem(x,y);
xlabel('n');
ylabel('amp');
title('unit step');
% DELAYED UNIT STEP
X=-10:1:10;
Y=((X-4)>=0);
subplot(7,1,7);
stem(x,y);
xlabel('n');
ylabel('amp');
title('delayed unit impluse');
```

```
%EXPONENTIAL FUNCTION
figure(2);
x=0:0.01:10;
subplot(7,1,1);
plot(x,exp(x));
xlabel('t');
ylabel('amp');
title('exponential function');
% RAMP FUNCTION
x=0:1:20;
y=0:1:20;
subplot(7,1,3);
plot(x,y);
xlabel('t');
ylabel('amp');
title('ramp function');
% RAMDOM FUNCTION
x=rand(1,10);
%y=rand(1,10);
subplot(7,1,5);
plot(x);
xlabel('t');
ylabel('amp');
title('random function');
%SINUSOIDAL FUNCTION
t=linespace(0,2*pi,1000);
disp(   '  INPUT   ');
disp('   ');
a=input('enter the amplitude       :');
f=input('enter the frequency       :');
th=input('enter the angle          :');
theta=th*pi/180;
y=a*sin((2*pi*f*t)+theta);
subplot(7,1,7);
plot(t,y);
xlabel('t');
ylabel('amp');
title('sinusoidal function');
%COSINUSOIDAL FUNCTION
figure(3);
t=linespace(0,2*pi,1000);
disp(   '  INPUT   ');
disp('   ');
a=input('enter the amplitude       :');
f=input('enter the frequency       :');
th=input('enter the angle          :');
theta=th*pi/180;
y=a*cos((2*pi*f*t)+theta);
subplot(7,1,1);
plot(t,y);
xlabel('t');
ylabel('amp');
title('cosinusoidal function');
```

Model output signals:

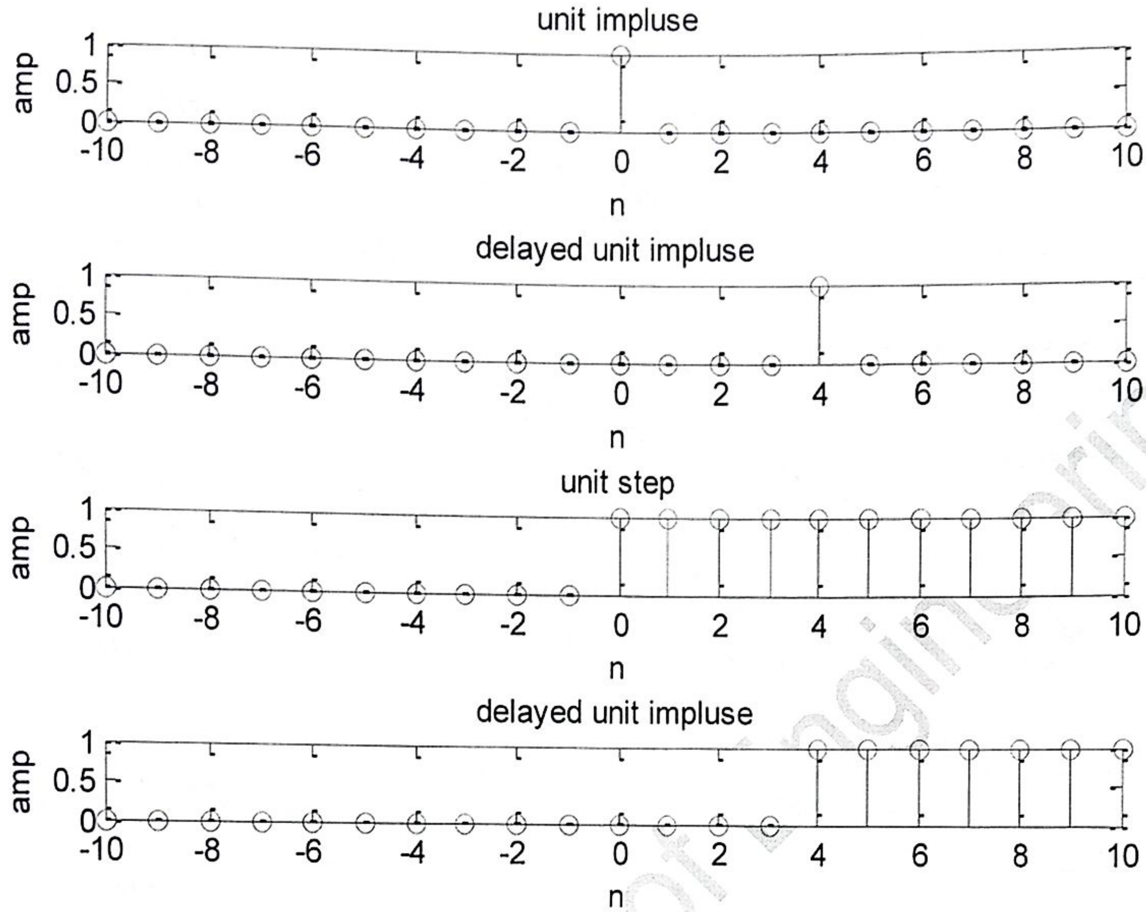
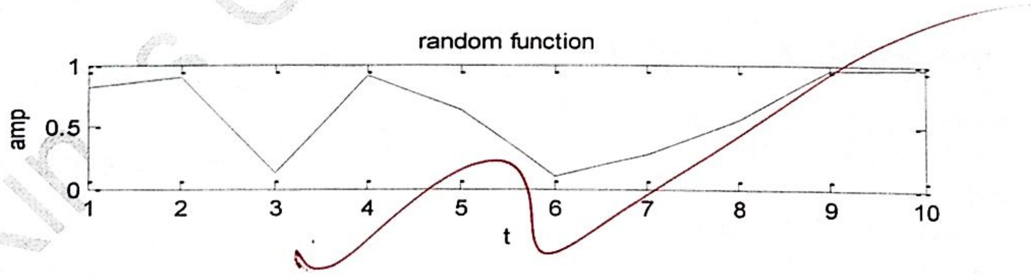


Figure 1.1.Generation of Signals



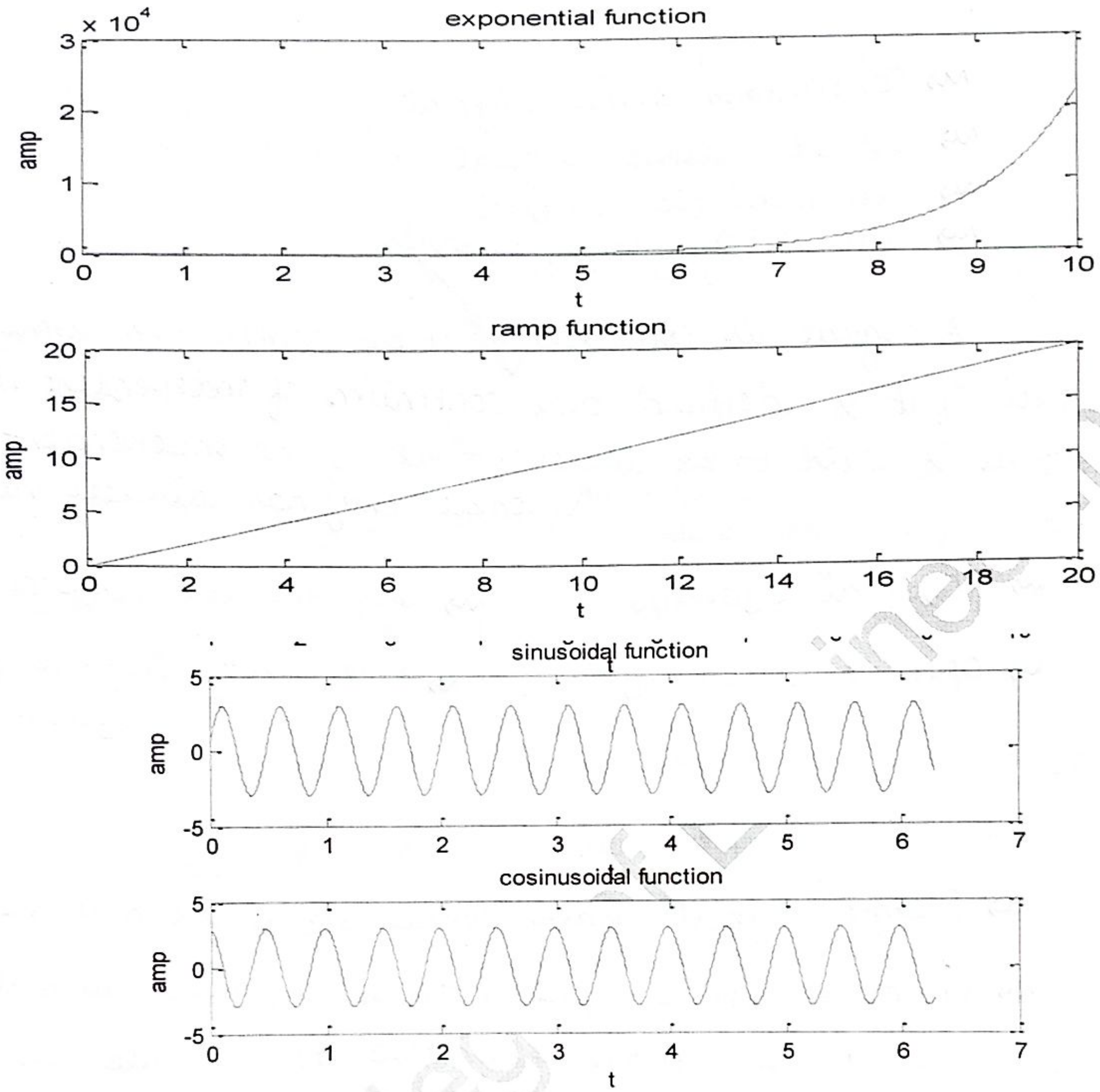


Figure 1.2.Generation of Signals

RESULT:

Thus the graphical representation of unit impulse, unit step, delayed unit step, Exponential, Sine wave, Cosine wave, Ramp and Random function was generated and verified.

VIVA QUESTIONS:

1. Give the classification of signals.

→ Continuous time signal
→ Discrete time signal
→ Deterministic signals
→ Non - deterministic signals

2. What is meant by continuous and discrete time signal?

A signal is considered to be continuous time signal if it is defined over continuum of independent variable. A signal is said to be discrete time if the independent variable only has discrete value.

3. What are the types of systems?

→ Physical systems → Deterministic systems
→ Open or closed systems → Man-made information system.

4. What is a energy signal? How to calculate energy of a signal?

→ Energy signal have power signal at a time.
→ We have 1 by 2n plus 1 then we have summation. n and we have a square. Because the inside we write x of n square. correct so mod of x of n square is square.

5. What is a power signal? How to calculate power of a signal?

The power of the signal is the sum of absolute squares of time-domain samples divided by signal length, or equivalently, the square of RMS level.

Calculate, the power for one period. For the analog sine we have $pa = 11 \int 10 \sin 2(2\pi t)dt = 12$.

Expt. No.: 2	LINEAR AND CIRCULAR CONVOLUTION
Date: 23.8.2021 26.8.2021	

AIM:

To write the program for finding the linear and circular convolution of two signals using MATLAB.

APPARATUS REQUIRED:

PC with MATLAB 9.0

ALGORITHM:

Step 1: Get the input x(n) and h(n) sequence.
Step 2: Use conv ( ) function.
Step 3: Plot the output sequence.

PROGRAM:

```
% LINEAR CONVOLUTION
clc;
clear all;
close all;
x=input('Enter the x(n) sequences');
y=input('Enter the y(n) sequences');
z=conv(x,y);
subplot(3,1,1);
stem(x);
xlabel('time');
ylabel('amp');
title('input x(n) sequences');
subplot(3,1,2);
stem(y);
xlabel('time');
ylabel('amp');
title('impluse input h(n) sequences');
subplot(3,1,3);
stem(z);
xlabel('time');
ylabel('amp');
title('output h(n) sequences');
% CIRCULAR CONVOLUTION
clc;
clear all;
close all;
x=input('enter the x1(n) sequence');
h=input('enter the x2(n) sequence');
N1=length(x);
N2=length(h);
N=max(N1,N2);
N3=N1-N2;
if(N3<0)
x=[x,zeros(1,-N3)];
else
x=[h,zeros(1,N3)];
end
for n=1:N
```

```
y(n)=0;
for i=1:N
j=n-i+1
if(j<=0)
j=N+j
end
y(n)=y(n)+x(i)*h(j);
end
end
subplot(3,1,1);
stem(x);
xlabel('time');
ylabel('amp');
title('input x1(n) sequences');
subplot(3,1,2);
stem(h);
xlabel('time');
ylabel('amp');
title('input x2(n) sequences');
subplot(3,1,3);
stem(y);
xlabel('time');
ylabel('amp');
title('output y(n) sequences');
```

Various Values of Linear and Circular Convolution:

S.NO.	X(n)	H(n)	Y(n)
Linear Convolution			
1.	{1,2,3,1,2,1,1}	{1,2,1}	{1.4,8,9,7,6,5,3,1}
2.	{1,1,1,1}	{2,2}	{2,4,4,4,2}
Circular Convolution			
3.	{1,2,3,4}	{2,1,3,0}	{15,17,11,17}
4.	{1,2,3,1}	{4,3,2,1}	{15,16,21,18}

Various values of linear and circular convolution

CALCULATION :-

Linear Convolution :

$x(n) = \{1, 2, 3, 1, 2, 1, 1\}$
$H(n) = \{1, 2, 1\}$
$L = 7 \quad , \quad M = 3$
$n = L+M-1 = 7+3-1 = 9$

h(n) \ x(n)	1	2	3	1	2	1	1
1	1	2	3	1	2	1	1
2	2	4	6	2	4	2	2
1	1	2	3	1	2	1	1

Add values

1, 2+2, 1+4+3, 2+6+1, 3+2+2, 1+4+1,
2+2+1, 1+2+1

$x(n) * h(n) = y(n) = \{1, 4, 8, 9, 7, 6, 5, 3, 1\}$

Circular Convolution :

$$x(n) = \{1, 2, 3, 4\}$$
$$h(n) = \{2, 1, 3, 0\}$$
$$y(n) = x(n) * h(n)$$

$$\begin{pmatrix} x_1(0) & x_1(3) & x_1(2) & x_1(1) \\ x_1(1) & x_1(0) & x_1(3) & x_1(2) \\ x_1(2) & x_1(1) & x_1(0) & x_1(3) \\ x_1(3) & x_1(2) & x_1(1) & x_1(0) \end{pmatrix} \begin{pmatrix} x_2(0) \\ x_2(1) \\ x_2(2) \\ x_2(3) \end{pmatrix} = \begin{pmatrix} y(0) \\ y(1) \\ y(2) \\ y(3) \end{pmatrix}$$

$$\begin{pmatrix} 1 & 4 & 3 & 2 \\ 2 & 1 & 4 & 3 \\ 3 & 2 & 1 & 4 \\ 4 & 3 & 2 & 1 \end{pmatrix} \begin{pmatrix} 2 \\ 1 \\ 3 \\ 0 \end{pmatrix} = \begin{pmatrix} 15 \\ 17 \\ 11 \\ 17 \end{pmatrix}$$

convolved signal is

$$y(n) = \{15, 17, 11, 17\}$$

Model output signals:

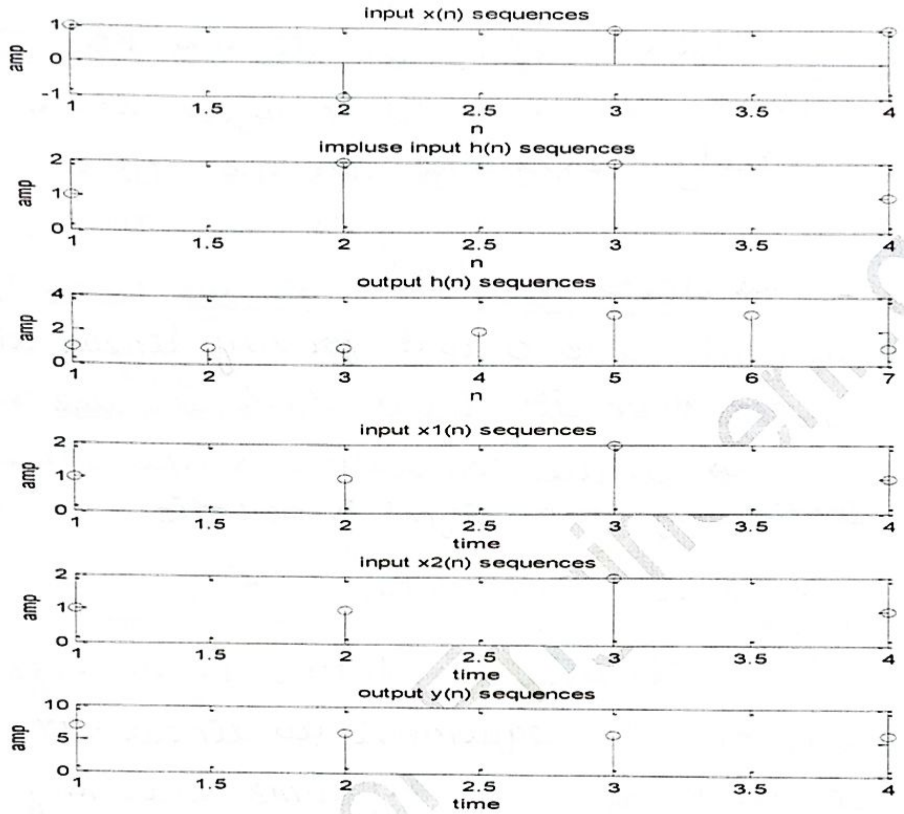


Figure 2.1.Linear and Circular convolution of sequences

10

26/8/2019

RESULT:
Thus the linear convolution and circular convolution of the given sequence was performed and the result was verified.

VIVA QUESTIONS:

1. What is meant by causality?

Influence by which one event, process, state or object contributes to the production of another event, process, state or object where the cause is partly responsible for the effect.

2. Differentiate linear convolution with circular convolution.

→ linear convolution is the basic operation to calculate the output for any linear time invariant system given its input and impulse response.

→ circular convolution is the same thing but considering the signal is periodic.

3. What is the length of linear and circular convolutions if the two sequences are having the length n1 and n2?

Circular convolution of two-finite length sequence is equivalent to linear convolution of two sequence, followed by time aliasing.

4. How to perform linear convolution using circular convolution?

Circular convolution of x and y to be equivalent, you must pad the vectors with zeros to length at least N+L-1 before you take the DFT. After you invert the product of DFT's. retain N+L-1 elements.

5. What is the use of function command 'deconv'?

This MATLAB function deconvolves a vector v out of vector u using long division, and returns the quotient q and remainder r such that u = conv(v, q) + r.

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

Practical Record Note

Subject Title: DIGITAL SIGNAL PROCESSING LABORATORY

Subject Code: EC8562

Name : SM. SWETHAA

Reg. No. : 821119106046

Degree : B.E.

Branch : ECE

Year/ Sem : III / V

Academic Year: 2021 – 2022

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DEPARTMENT OF
ELECTRONICS & COMMUNICATION ENGINEERING

BONAFIDE CERTIFICATE

Register Number: 821119106046

This is to certify that, the record work was done by the candidate Mr./Ms. SM. SWETHAA *of III Year, V Semester, B.E. Electronics and Communication Engineering for EC8562–DIGITAL SIGNAL PROCESSING LABORATORY during the academic year 2021-2022(Odd).*

Staff in-Charge 7/12/21

Head of the Department 7/12/21

H.O.D.
ELECTRONICS AND COMMUNICATION ENGINEERING
KINGS COLLEGE OF ENGINEERING
PUNALKULAM - 613 303.
GANDARVAKOTTAI TALUK, PUDUKOTTAI DISTRICT

This record is submitted for Anna University, Chennai, practical examination held on 11.03.2022 at Kings College of Engineering, Punalkulam.

Internal Examiner 11/3/22

External Examiner 11/3/22

CONTENTS

SIGN OF STAFF INCHARGE

Completed

SIGN OF HOD

EX. NO. : 1 GENERATION OF ELEMNTARY DISCRETE

DATE : 19.08.2021 TIME SEQUENCES

AIM :

To generate elemantary time sequences of unit impulse, unit step, delayed unit step exponential sine wave, cosine wave, Ramp and Random function using MATLAB.

APPARATUS REQUIRED :

PC with MATLAB 9.0

FUNCTION USED :

Plot (x, y);
stem (x,y)
Supplot (2, 1, 1);

PROGRAM :

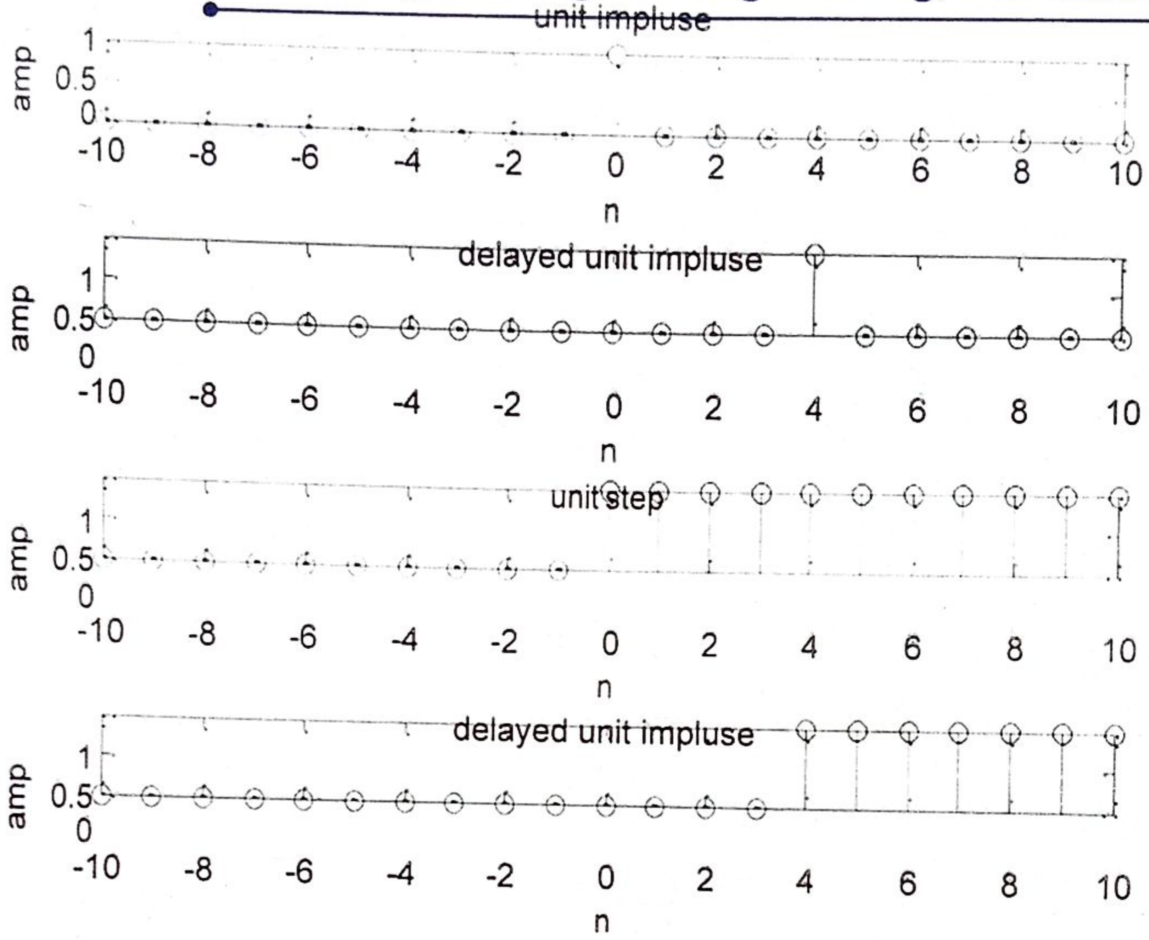
```
% GENERATION OF INPUT SIGNALS
clc ;
% UNIT IMPULSE
X = -10 : 1 :10 ;
Y = ((X-0) ==0) ;
subplot (7, 1, 1);
stem (x, y) ;
xlabel ('n');
ylabel ('amp');
title ('unit impulse');
```

```
% DELAYED UNIT IMPULSE
X =-10:1: 10;
Y = ((X - 4) = =0);
Subplot (7,1,3);
stem (x, y);
xlabel ('n');
ylabel ('amp');
title ('delayed unit impulse');
% UNIT STEP
X = -10:1:10;
Y = ((X - 0) >=0);
Subplot (7,1,5);
stem (x,y);
Xlabel ('n');
ylabel ('amp');
title ('unit step');
% DELAYED UNIT STEP
X =-10:1:10
Y = ((X-4) >=0);
subplot (7,1,7);
stem (x, y);
xlabel ('n');
Ylabel ('amp');
title ('delayed unit impulse');
```

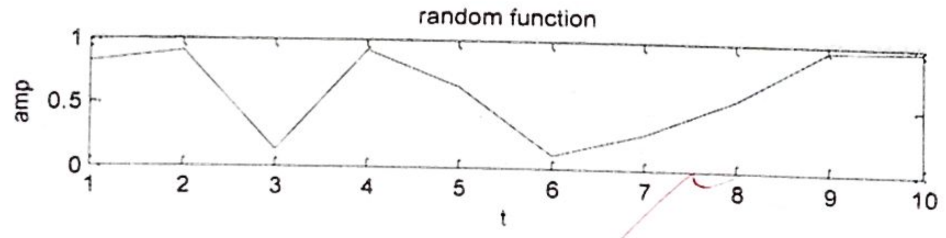
```
% EXPONENTIAL FUNCTION
x = 0:0.1:10;
subplot (7,1,1);
xlabel ('t');
ylabel ('amp');
title ('exponential function');

% RAMP FUNCTION
x = 0:1:20;
y = 0:1:20;
plot (x,y);
xlabel ("t');
ylabel ('amp');

% RANDOM FUNCTION
x = rand (1,10);
% y = rand (1,10);
plot (x);
xlabel ('t');
ylabel ('amp');
title ('random function');

% SINUSOIDAL FUNCTION
t = linespace (0, 2* pi, 1000);
disp ('INPUT');
disp (' ');
a = input ('enter the amplitude :');
f = input ('enter the frequency :');
th = input ('enter the angle :');
```

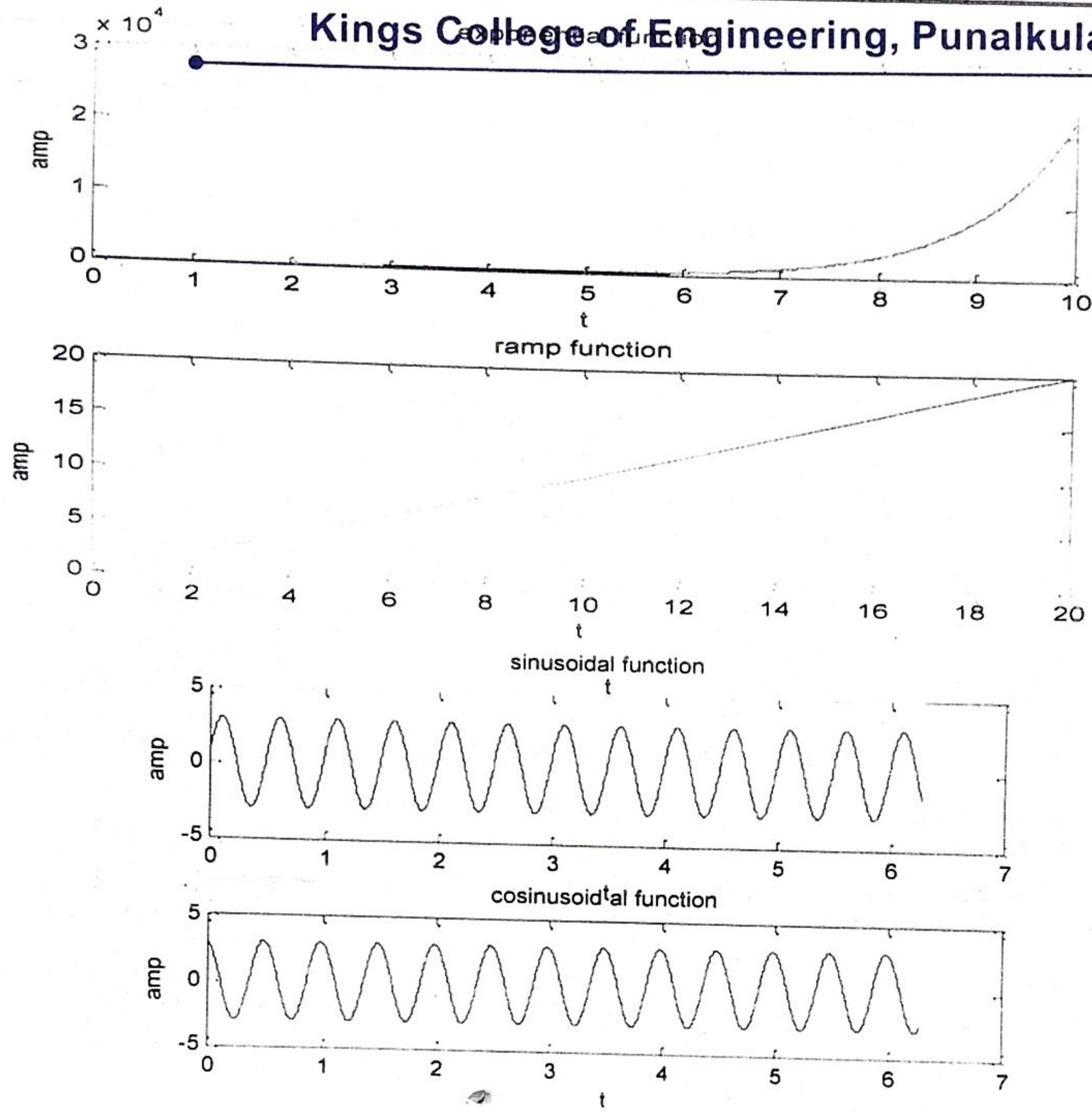
```
theta = th* pi/180;
y = a* sin ((2*pi * f *t) + theta);
xlabel ('t');
ylabel ('amp');
title ('sinusoidal function');
% COSINUSOIDAL FUNCTION
t= linespace (0, 2 * pi, 1000);
disp ('INPUT');
a= input ('enter the amplitude :');
f = input ('enter the frequency :');
th = input ('enter the angle :');
theta = th * pi/180;
y= a*cos ((2 * pi * f * t) + theta);
subplot (7, 1, 1);
xlabel ('t');
Ylabel ('amp');
title ('cosinusoidal function');
```

Output :



Generation of Signals





Generation of Signals

VIVA QUESTIONS:

1. Give the classification of signals.

Continuous Time and Discrete Time Signals. Deterministic and Non-deterministic Signals. ... Real and Imaginary Signals.

2. What is meant by continuous and discrete time signal?

A signal is considered to be a continuous time signal if **it is defined over a continuum of the independent variable**. A signal is considered to be discrete time if the independent variable only has discrete values.

3. What are the types of systems?

Types of Systems

Physical or abstract systems.

Open or closed systems.

Deterministic or probabilistic systems.

Man-made information systems.

4. What is a energy signal? How to calculate energy of a signal?

If your signal is said to be power signal definitely that convey energy signal. So a signal can be both energy and power signal at a time.

Then. We have 1 by 2 n plus 1 then we have summation - n - and then we have a square. Because the inside we write X of n square. Correct so mod of X of n square is a square

5. What is a power signal? How to calculate power of a signal?

The power of a signal is **the sum of the absolute squares of its time-domain samples divided by the signal length**, or, equivalently, the square of its RMS level.

calculate the power for one period. For the analog sine we have Pa=11∫10sin2(2πt)dt=12.

RESULT :

Thus the graphical representation of unit impulse. unit step, delayed unit step, Exponential, sine wave, Cosine wave, Ramp and Random function was generated and verified.

EX. NO: 2 LINEAR AND CIRCULAR CONVOLUTION

DATE: 23.8.2021
26.8.2021

AIM:

To write the program for finding the linear and circular convolution of two signals using MATLAB.

APPARATUS REQUIRED:

PC with MATLAB 9.0

ALGORITHM:

Step 1: Get the input x(n) and h(n) sequence.

Step 2: Use conv() function.

Step 3: Plot the output sequence.

PROGRAM:

```
% LINEAR CONVOLUTION
clc;
clear all;
close all;
x = input ('Enter the x(n) sequences:');
y = input ( 'Enter the y(n) sequences:');
z = conv (x, y);
x label ('time');
y label ('amp');
title ('Impulse input h(n) sequences');
x label ('time');
y label ('amp');
title ('output h(n) sequences');
```

Various values of linear and circular convolution

S. NO	x(n)	h(n)	y(n)
Linear convolution			
1)	{1, 2, 3, 1, 2, 1, 1}	{1, 2, 1}	{1, 4, 8, 9, 7, 6, 5, 3, 1}
2)	{1, 1, 1}	{2, 2}	{2, 4, 4, 4, 2}
Circular convolution			
1)	{1, 2, 3, 4}	{2, 1, 3, 0}	{15, 17, 11, 17}
2)	{1, 2, 3, 1}	{4, 3, 2, 1}	{15, 16, 21, 18}

Linear convolution :

$x(n) = \{1, 2, 3, 1, 2, 1, 1\}$; $h(n) = \{1, 2, 1\}$; $L = 7$;

$M = 3$; $N = L + M - 1 = 9$

h(n) \ x(n)	1	2	3	1	2	1	1
1	1	2	3	1	2	1	1
2	2	4	6	2	4	2	2
1	1	2	3	1	2	1	1

```
% CIRCULAR CONVOLUTION
clc;
x = input ('enter the x1(n) sequence');
h = input (' enter the x2(n) sequence ');
N1 = length (x);
N2 = max (N1, N2);
N = length (h);
if (N3<0)
x [ x, zeros (1, -N3)];
else
x = h [zeros (1, N3)];
for n=1; N
    y(n)=0
end
y(n) = y(n) + x(i) * h(j);
end
subplot (3, 1, 1);
stem (x);
xlable ('time');
ylable ('amp');
subplot (3, 1, 2);
title (' input x2(n) sequences');
subplot (3, 1, 3);
stem (y);
xlabel ('time');
ylabel ('amp');
title ('output y(n) sequences');
```

Add values :-

1, 2+2, 1+4+3, 2+6+1, 3+2+2, 1+4+1, 6+2+1,
1+2,

$x(n) * h(n) = y(n) = \{1, 4, 8, 9, 7, 6, 5, 3, 1\}$

circular convolution :-

$x(n) = \{1, 2, 3, 4\}$

$h(n) = \{2, 1, 3, 0\}$

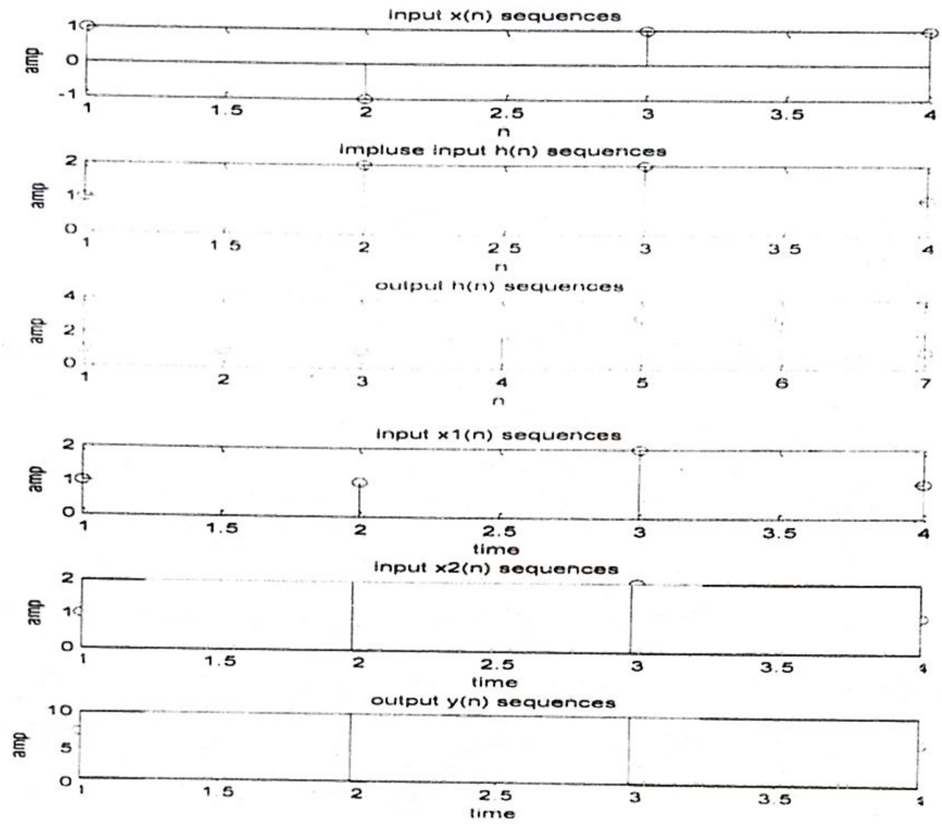
$y(n) = x(n) * h(n)$

$$\begin{pmatrix} x_1(0) & x_1(2) & x_1(2) & x_1(1) \\ x_1(1) & x_1(0) & x_1(3) & x_1(2) \\ x_1(2) & x_1(1) & x_1(0) & x_1(3) \\ x_1(3) & x_1(2) & x_1(1) & x_1(0) \end{pmatrix} \begin{pmatrix} x_2(0) \\ x_2(1) \\ x_2(2) \\ x_2(3) \end{pmatrix} = \begin{pmatrix} y(0) \\ y_1(1) \\ y(2) \\ y(3 \end{pmatrix}$$

$$\begin{pmatrix} 1 & 4 & 3 & 2 \\ 2 & 1 & 4 & 3 \\ 3 & 2 & 1 & 4 \\ 4 & 3 & 2 & 1 \end{pmatrix} \begin{pmatrix} 2 \\ 1 \\ 3 \\ 0 \end{pmatrix} = \begin{pmatrix} 15 \\ 17 \\ 11 \\ 17 \end{pmatrix}$$

convolved signal is $y(n) = \{15, 17, 11, 17\}$

output signals:



Linear and Circular convolution of sequences

VIVA QUESTIONS:

1. What is meant by causality?

Influence by which one event, process, state or object (a cause) contributes to the production of another event, process, state or object (an effect) where the cause is partly responsible for the effect,

2. Differentiate linear convolution with circular convolution.

Linear convolution is the basic operation to calculate the output for any linear time invariant system given its input and its impulse response.

Circular convolution is the same thing but considering that the support of the signal is periodic (as in a circle, hence the name).

3. What is the length of linear and circular convolutions if the two sequences are having the length n1 and n2?

Thus, the circular convolution of two finite-length sequences is **equivalent to the linear convolution of the two sequences, followed by time aliasing**. Example: Consider two identical sequences x1[n] and x2[n] of length L in OSB

4. How to perform linear convolution using circular convolution?

For the circular convolution of x and y to be equivalent, you must **pad the vectors with zeros to length at least N + L - 1 before you take** the DFT. After you invert the product of the DFTs, retain only the first N + L - 1 elements. Create two vectors, x and y , and compute the linear convolution of the two vectors.

5. What is the use of function command 'deconv'?

This MATLAB **function** deconvolves a vector v out of a vector u using long division, and returns the quotient q and remainder r such that u = conv(v,q) + r.

RESULT :

Thus the linear convolution and circular convolution of given sequence and performed and result was verified.

B.E / B.Tech. PRACTICAL END SEMESTER EXAMINATIONS, NOVEMBER / DECEMBER 2019

Fifth Semester

EC8562 - DIGITAL SIGNAL PROCESSING LABORATORY

(Regulations 2017)

Time : 3 Hours | Answer any one Question | Max. Marks 100

(To be filled by the question paper setter)

Aim/Principle/Apparatus required/Procedure	Tabulation/Circuit/ Program/Drawing	Calculation & Results	Viva-Voce	Record	Total
25	35	20	10	10	100

1.	(a) Generate standard discrete time sequences namely unit impulse, unit step, unit ramp signals by running a suitable MATLAB program. (b) Write and execute a MATLAB program to generate the frequency spectrum of these standard signals.
2.	Write and execute a MATLAB program to perform periodic convolution of 2 discrete time sequences: $x_1(n) = [1,2,3,5,6,-7]$ and $x_2(n) = [4,5,6,7,8]$. Also verify the results manually.
3.	Write and execute a MATLAB program to perform convolution of the 2 discrete time sequences along linear time axis: $x_1(n) = 2$, $2<n<7$ and $x_2(n) = 1$, $1<n<8$; Also verify the results manually.
4.	Write and execute a MATLAB program to perform cross correlation of the sequences $x(n) = [1,1,2,2]$ and $y(n) = [1,0,5,1]$. Also verify the results manually.
5.	By executing a suitable MATLAB program, determine the auto correlation of $x(n) = [1,2,3,4]$. Also verify the results manually.
6.	Write and execute a suitable MATLAB program to compute 8 point DFT of the discrete time sequence $x(n) = [2,1,2,1,1,2,1,2]$. Also plot the magnitude and phase spectrum.
7.	Write and execute a MATLAB program to find the response of an LTI system for the input $x(n) = [1,2,3,4]$ and the impulse response $h(n) = [1,2,2,1]$ by performing circular convolution of suitable length. Verify the system response through linear convolution.
8.	Write and execute a MATLAB program to determine the impulse response of FIR lowpass filter using rectangular window and hence plot the frequency response. Verify the response by working out the design manually.
9.	Write and execute a MATLAB program to determine the impulse response of FIR highpass filter using Hamming window and hence plot the frequency response. Verify the response by working out the design manually.
10.	By executing a suitable MATLAB program, determine the impulse response of FIR bandpass filter using Hanning window and hence plot the frequency response. Verify the response by working out the design manually.
11.	By executing a suitable MATLAB program, determine the impulse response of FIR bandstop filter using Fourier series method and hence plot the frequency response. Verify the response by working out the design manually.
12.	Write and execute a MATLAB program to design a Butterworth digital IIR low pass filter using Bilinear transformation to satisfy the following specifications: T=0.1 sec ; $0.6 \leq \lvert H(e^{j\omega}) \rvert \leq 1.0$; for $0 \leq \omega \leq .35\pi$ $\lvert H(e^{j\omega}) \rvert \leq 0.1$; for $0.7\pi \leq \omega \leq \pi$
13.	Write and execute a MATLAB program to design a Butterworth digital IIR high pass filter using Bilinear transformation to satisfy the following specifications: T=0.1 sec ; $0.6 \leq \lvert H(e^{j\omega}) \rvert \leq 1.0$; for $0.7\pi \leq \omega \leq \pi$ $\lvert H(e^{j\omega}) \rvert \leq 0.1$; for $0 \leq \omega \leq .35\pi$
14.	Write and execute a MATLAB program to design a Butterworth digital IIR low pass filter using Impulse invariant transformation to satisfy the following specifications: T=1 sec ; $0.707 \leq \lvert H(e^{j\omega}) \rvert \leq 1.0$; for $0 \leq \omega \leq .3\pi$ $\lvert H(e^{j\omega}) \rvert \leq 0.2$; for for $0.75\pi \leq \omega \leq \pi$
15.	Write and execute a MATLAB program to design a Chebyshev digital IIR low pass filter using Impulse invariant transformation to satisfy the following specifications: T=1 sec ; $0.9 \leq \lvert H(e^{j\omega}) \rvert \leq 1.0$; for $0 \leq \omega \leq .25\pi$ $\lvert H(e^{j\omega}) \rvert \leq 0.24$; for for $0.5\pi \leq \omega \leq \pi$

16.	Write and execute a MATLAB program to design a Chebyshev digital IIR low pass filter using Bilinear transformation to satisfy the following specifications: T=1 sec ; $0.8 \leq \lvert H(e^{jw}) \rvert \leq 1.0$; for $0 \leq \omega \leq 0.2\pi$ $\lvert H(e^{jw}) \rvert \leq 0.2$; for for $0.32\pi \leq \omega \leq \pi$
17.	Generate sinusoidal and square waveforms using TMS3205416 DSP processor.
18.	Implement the generation of triangular waveform and random noise using TMS3205416.
19.	Perform up sampling (by a factor of 3) and down sampling (by a factor of 2) on a discrete time signal using TMS3205416.
20.	Design and implement a FIR low pass filter with a cut off frequency of 1 KHz with TMS3205416. Plot the frequency response.
21.	Design and implement an IIR Butterworth low pass filter with a cut off frequency of 1 KHz using TMS3205416. Verify the frequency response.
22.	Demonstrate the indirect, direct and immediate addressing modes of TMS3205416 using programming examples

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

ACADEMIC YEAR 2021-2021(ODD SEMESTER)

Sub. Code : EC8562
Sub.Name : Digital Signal Processing Lab
Staff Name :Mr. R. Balakrishnan

Branch / Year / Sem : B.E ECE / III/V
Batch : 2019-2023
Academic Year : 2021-22 (ODD)
Date :02.12.2021

Model Lab –I Schedule

Sl.No	Batch number	Roll Number	Date	Time
1.	One	1-21	07.12.2021	9.15a.m. To 12.15a.m
2.	Two	22-42	07.12.2021	12.45p.m. To 3.00p.m.

2/12/21
Staff in charge
(Mr. R. Balakrishnan)

2/12/2021
HOD/ECE

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Date :02.12.2021

MARK ALLOCATION for MODEL LAB I

Sl.No	Contents	Marks
1.	AiM/Principle/Apparatus required/Procedure	25
2.	Program/ / Circuit Diagram/Block Diagram	35
3.	Calculation/Results	20
4.	Vivavoce	10
5.	Record	10
	Total	100

02/12/21

Staff in charge
(Mr. R. Balakrishnan)

1/12/2021

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING ACADEMIC YEAR 2021-2021(ODD SEMESTER)

Sub. Code : **EC8562**
Sub.Name : **Digital Signal Processing Lab**
Staff Name :Mr. R. Balakrishnan

Branch / Year / Sem : B.E ECE / III/V
Batch : 2019-2023
Academic Year : 2021-22 (ODD)
Date :02.12.2021

ATTENDANCE SHEET

R.NO	REG.NO. OF THE STUDENT	NAME OF THE STUDENT	SIGNATURE
1	821119106001	ABIMANEU S	
2	821119106002	AGALYA P	P Akyi.
3	821119106004	BLESSON MANUEL J	
4	821119106005	DHARMADURAI A	
5	821119106006	DHARSHINI C	C. Dharshini
6	821119106007	DURGA SRI R	R. Durgasri.
7	821119106008	GANGA L	L. Ganga
8	821119106009	GANGA R	R. Ganga
9	821119106010	GAYATHRI K	K. Gayathri
10	821119106011	GAYATHRI S	S. Gayathri
11	821119106012	ISHWARYA K	K. Ishwarya
12	821119106013	JAYAKUMAR A	A. Jayakumar.
13	821119106015	JOTHIKA R	R. Jothika
14	821119106016	KABILAN R	R. Kabilan
15	821119106017	KABISHENA P	P. K
16	821119106019	KARIKALAN G	Karikalan. G
17	821119106020	KARTHICK N	AB.
18	821119106021	KARTHIKA DEVI M	M Karthika Devi
19	821119106022	KIRUBADHARSHINI S	S. Kirubadharshini
20	821119106023	KRISHNADEVI G	AB – Long Absent
21	821119106024	LOGESHWARAN P	P. Logeshwaran

22	821119106025	MADHUMITHA G	G. Madhumitha
23	821119106026	MAHESWARI V	[illegible]
24	821119106027	MATHIVANAN K	K. mathivanan
25	821119106028	NITHITHA U	U. Nitthitha
26	821119106029	NIVETHITHA S	S. Nivethitha
27	821119106030	PAVITHRA P	[illegible]
28	821119106031	PRAKASH A	[illegible]
29	821119106032	PRETHIYA B	B. Prethiya
30	821119106033	PRIYANKA K	K. Priyanka.
31	821119106034	RAMANA BHARATHI S	[illegible]
32	821119106035	RENUKA K	Renuka. K
33	821119106036	RUTHRA R	R. Ruthra
34	821119106037	SABARINATHAN S	[illegible]
35	821119106039	SARASWATHI K	K. Sathy
36	821119106040	SATHYA G	G. Sathya
37	821119106042	SHATHANA B	B. Shathana
38	821119106043	SOUNDHARYA R	R. Soundharya
39	821119106044	SURIYA C	Suriya. c
40	821119106045	SUSIKUMAR T	[illegible]
41	821119106046	SWETHAA S M	SM. Swethaa
42	821119106048	VAISHNAVI G	G. Vaishnavi

2/12/21

Staff in charge
(Mr. R. Balakrishnan)

2/12/202

HOD/ECE

Reg. No. : 821119106046

Roll. No. : 19EC41

Class : III year ECE

EC8562 - DIGITAL SIGNAL PROCESSING LAB

MARK ALLOCATION

CONTENTS	MARKS ALLOCATED	MARKS OBTAINED
Aim / principle / Apparatus required / procedure	25 marks	25
Tabulation / circuit / program / Drawing	35 marks	35
Calculation / Result	20 marks	20
Viva voce	10 marks	10
Record.	10 marks	10
TOTAL	100 Marks.	100

One zero zero.

2. Write and execute a MATLAB program to perform periodic convolution of 2 discrete sequences: $x_1(n) = [1,2,3,5,6,-7]$ and $x_2(n) = [4,5,6,7,8]$. Also verify the results manually.

AIM:

To write the program to the linear convolution of two signals using MATLAB.

APPARATUS REQUIRED:

PC with MATLAB 9.0

ALGORITHM:

Step 1 : Get the input $x_1(n)$ and $x_2(n)$.

Step 2 : Use conv() function

Step 3 : Plot the output sequence.

CALCULATION:

$x_1(n) = [1, 2, 3, 5, 6, -7]$ and $x_2(n) = [4, 5, 6, 7, 8]$

$x_2(n)$ \ $x_1(n)$	1	2	3	5	6	-7
4	4	8	12	20	24	-28
5	5	10	15	25	30	-35
6	6	12	18	30	36	-42
7	7	14	21	35	42	-49
8	8	16	24	40	48	-56

$y(1) = 4 \quad = 4$

$y(2) = 5+8 \quad = 13$

$y(3) = 6+10+12 \quad = 28$

$y(4) = 7+12+15+20 \quad = 54$

$y(5) = 8+14+18+25+24 \quad = 89$

$y(6) = 16+21+30+30-28 \quad = 69$

$y(7) = 24+35+36-35 \quad = 60$

$y(8) = 40+42-42 \quad = 40$

$y(9) = 48-49 \quad = -1$

$y(10) = -56 \quad = -56$

$y(n) = [4, 13, 28, 54, 89, 69, 60, 40, -1, -56]$

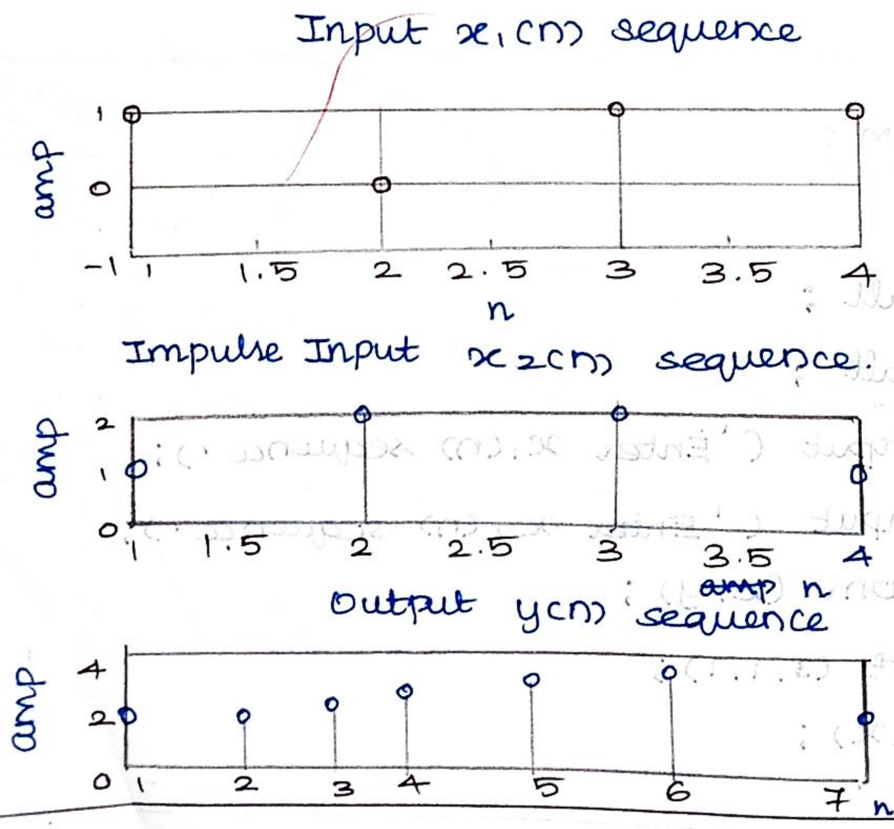
$x_1(n)$	$x_2(n)$	$y(n)$
[1, 2, 3, 5, 6, 7]	[4, 5, 6, 7, 8]	[4, 13, 28, 54, 89, 69, 60, 40, -1, -56]

PROGRAM :

```
clc;
clear all;
close all;
x = input ('Enter x1(n) sequence');
y = input ('Enter x2(n) sequence');
z = conv (x, y);
subplot (3,1,1);
stem (x);
```

```
xlabel ('time');
ylabel ('amp');
title ('Input x1(n) sequence');
Subplot (3, 1, 2);
stem (y);
xlabel ('time');
ylabel ('amp');
title ('Impulse input x2(n) sequence');
Subplot (3, 1, 3);
stem (z);
xlabel ('time');
ylabel ('amp');
title ('output y(n) sequence');
```

OUTPUT SIGNAL :



MATLAB OUTPUT :

Enter $x_1(n)$ sequence : 1, 2, 3, 5, 6, -7

Enter $x_2(n)$ sequence : 4, 5, 6, 7, 8

After convolution

Output $y(n)$ sequence : 4, 13, 28, 54, 89, 69, 60, 40, -1, -56

o/p verified

RESULT :

Thus the linear convolution for the given sequence is proceded and the result is verified successfully.

EC8562- Digital Signal Processing Lab

Regno : 821119106033 Class : III - ECE

ROLL NO : 19EC30

Allocation of Marks :

Contents	Marks allocated	Marks obtained
Aim / priciple / apparatus required Procedure	25 marks	25
Tabulation / circuit / Program Drawing	35 marks	35
Calculation / Result	20 marks	20
Viva voce	10 marks	2
Record	10 marks	10
Total.	100 -	92

Nice Tried

5.	a) Write a Mat Lab program to calculate Autocorrelation and energy of sequ x(n)={1,3,5,2,1,5} and verification of its properties. b) Write a Mat Lab program to calculate Cross-correlation of a given sequences and verifica of its any four properties if x(n){1,1,2,2} and y(n)={3,3,4,4}. Plot the results.

30

(a) AIM :

To write a program for auto correlation and cross correlation signals using MATLAB.

APPARATUS REQUIRED :

PC with MATLAB 9.0

ALGORITHM :

Step 1 : Get input x(n) and h(n) sequence

Step 2 : Use xcorr ( ) function

Step 3 : Plot the output sequence.

Calculation :

(a) $x(n) = \{1, 3, 5, 2, 1, 5\}$

$$\begin{pmatrix} 1 & 3 & 5 & 2 & 1 & 5 \\ 3 & 5 & 2 & 1 & 5 & 1 \\ 5 & 2 & 1 & 5 & 1 & 3 \\ 2 & 1 & 5 & 1 & 3 & 5 \\ 1 & 5 & 1 & 3 & 5 & 2 \\ 5 & 1 & 3 & 5 & 2 & 1 \end{pmatrix} \begin{pmatrix} 5 \\ 1 \\ 2 \\ 5 \\ 3 \\ 1 \end{pmatrix}$$

$$= \begin{pmatrix} 36 \\ 45 \\ 60 \\ 40 \\ 44 \\ 64 \end{pmatrix}$$

(b) Cross correlation.

$x(n) = \{1, 1, 2, 2\}$ $y(n) = \{3, 3, 4, 4\}$

$$= \begin{pmatrix} 1 & 1 & 2 & 2 \\ 1 & 2 & 2 & 1 \\ 2 & 2 & 1 & 1 \\ 2 & 1 & 1 & 2 \end{pmatrix} \begin{pmatrix} 4 \\ 4 \\ 3 \\ 3 \end{pmatrix}$$

$$= \begin{pmatrix} 4+4+6+6 \\ 4+8+6+3 \\ 8+8+3+3 \\ 8+4+3+6 \end{pmatrix}$$

$$= \begin{pmatrix} 20 \\ 21 \\ 22 \\ 21 \end{pmatrix}$$

a) program % AUTO CORRELATION.

```
x=input('Enter any sequence');
Subplot(3,2,1);
Stem(x);
xlabel('Time Period');
ylabel('Amplitude');
title('Input sequence');
y=xcorr(x);
Subplot(3,2,2);
xlabel('Time period');
ylabel('Amplitude');
title('Auto correlation');
```

b) Program % CROSS CORRELATION

```
x=input('Enter any sequence');
Subplot(3,2,1);
Stem(x);
xlabel('Time period');
ylabel('Amplitude');
h=input('Enter any sequence');
Subplot(3,2,2);
xlabel('Time period')
ylabel('Amplitude')
title('Cross correlation');
```

Properties of auto correlation :

(i) The mean square value of a random process can be obtained from the auto-correlation function $R(\tau)$.

(ii) $R(\tau)$ is even function τ.

(iii) $R(\tau)$ is maximum at $\tau = 0$ $|R(\tau)| \leq R(0)$.

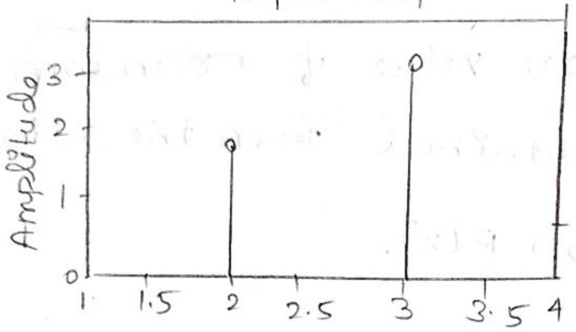
In other words, this means the maximum value $R(\tau)$ is attained at $\tau = 0$

(iv) If $R(\tau)$ is the auto correlation of a stationary random process $\{x(t)\}$ with no periodic components and with non zeros means ~~then~~

$$\lim_{\tau \to \infty} R(\tau) = [E(x)]^2$$

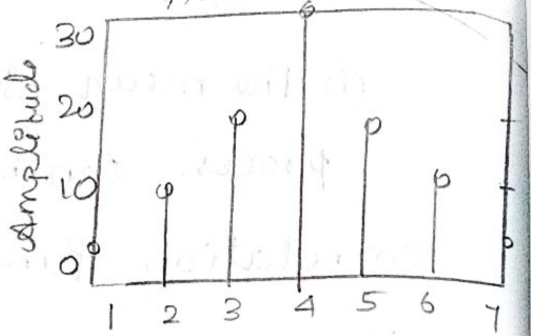
Output :

Input response



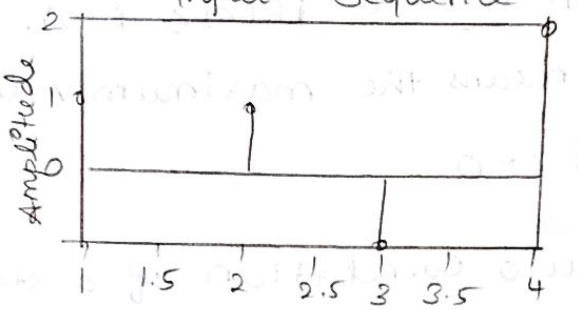
$x(n) = \{1, 2, 3, 4\}$

Auto correlation



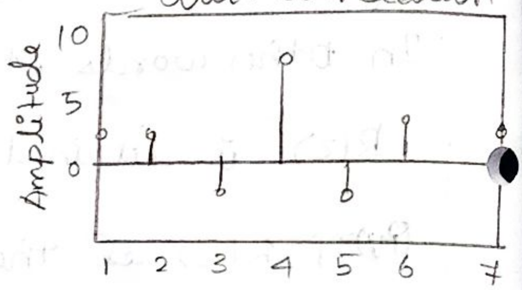
$R_{xx} = \{4, 11, 20, 30, 20, 11,$

Input sequence



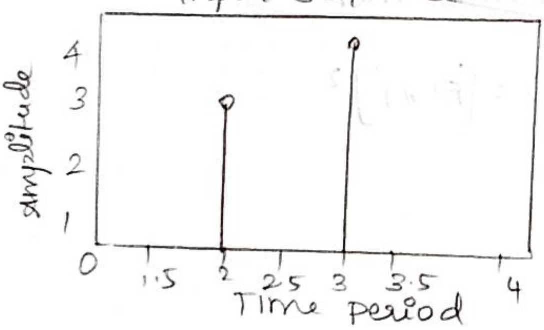
$x(n) = \{1, 2, 3, 4\}$

auto correlation

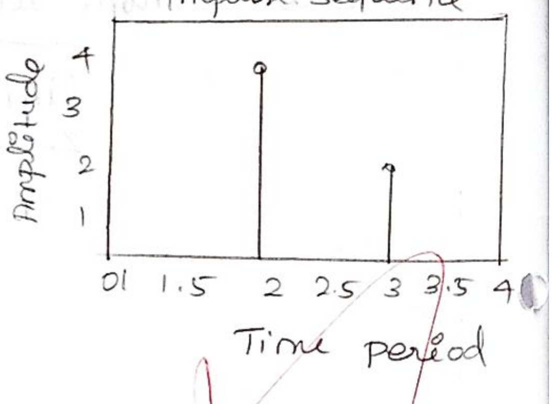


$R_{xx} = \{2, 1, -2, 7, 2, 1, -2\}$

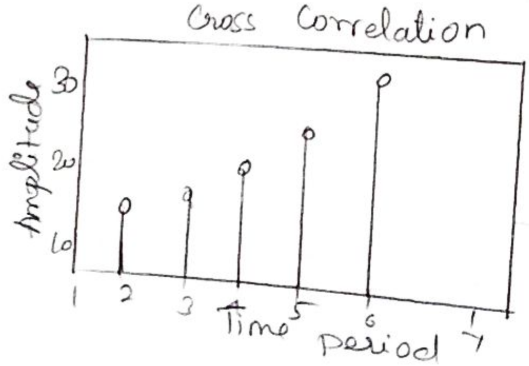
Input sequence



Impulse sequence



Cross Correlation



$R_{xy} = \{1, 4, 10, 20, 25, 24, 16\}$

Result :

Thus the auto correlation and cross correlation of the sequence performance and the result was verified.

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING ACADEMIC YEAR 2021-2021(ODD SEMESTER)

Sub. Code : EC8562
Sub.Name : Digital Signal Processing Lab
Staff Name :Mr. R. Balakrishnan

Branch / Year / Sem : B.E ECE / III/V
Batch : 2019-2023
Academic Year : 2021-22 (ODD)
Date : 02.12.2021

MODEL LAB MARKSHEET

R.NO	REG.NO. OF THE STUDENT	NAME OF THE STUDENT	MODEL MARK (100)	MARKS IN WORDS
1	821119106001	ABIMANEU S	92	NINE TWO
2	821119106002	AGALYA P	91	NINE ONE
3	821119106004	BLESSON MANUEL J	93	NINE THREE
4	821119106005	DHARMADURAI A	90	NINE ZERO
5	821119106006	DHARSHINI C	93	NINE THREE
6	821119106007	DURGA SRI R	93	NINE THREE
7	821119106008	GANGA L	98	NINE EIGHT
8	821119106009	GANGA R	99	NINE NINE
9	821119106010	GAYATHRI K	97	NINE SEVAN
10	821119106011	GAYATHRI S	90	NINE ZERO
11	821119106012	ISHWARYA K	91	NINE ONE
12	821119106013	JAYAKUMAR A	92	NINE TWO
13	821119106015	JOTHIKA R	100	ONE ZERO ZERO
14	821119106016	KABILAN R	85	EIGHT FIVE
15	821119106017	KABISHENA P	88	EIGHT EIGHT
16	821119106019	KARIKALAN G	89	EIGHT NINE
17	821119106020	KARTHICK N	AB	AB
18	821119106021	KARTHIKA DEVI M	94	NINE FOUR
19	821119106022	KIRUBADHARSHINI S	88	EIGHT EIGHT
20	821119106023	KRISHNADEVI G	AB	AB
21	821119106024	LOGESHWARAN P	91	NINE ONE

22	821119106025	MADHUMITHA G	93	NINE THREE
23	821119106026	MAHESWARI V	94	NINE FOUR
24	821119106027	MATHIVANAN K	96	NINE SIX
25	821119106028	NITHITHA U	98	NINE EIGHT
26	821119106029	NIVETHITHA S	99	NINE NINE
27	821119106030	PAVITHRA P	96	NINE SIX
28	821119106031	PRAKASH A	88	EIGHT EIGHT
29	821119106032	PRETHIYA B	89	EIGHT NINE
30	821119106033	PRIYANKA K	92	NINE TWO
31	821119106034	RAMANA BHARATHI S	93	NINE THREE
32	821119106035	RENUKA K	91	NINE ONE
33	821119106036	RUTHRA R	89	EIGHT NINE
34	821119106037	SABARINATHAN S	89	EIGHT NINE
35	821119106039	SARASWATHI K	89	EIGHT NINE
36	821119106040	SATHYA G	85	EIGHT FIVE
37	821119106042	SHATHANA B	98	NINE EIGHT
38	821119106043	SOUNDHARYA R	99	NINE NINE
39	821119106044	SURIYA C	95	NINE FIVE
40	821119106045	SUSIKUMAR T	91	NINE ONE
41	821119106046	SWETHAA S M	100	ONE ZERO ZERO
42	821119106048	VAISHNAVI G	100	ONE ZERO ZERO

2/12/21

Staff in charge
(Mr. R. Balakrishnan)

2/12/2021

HOD/ECE

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

ACADEMIC YEAR :________________(ODD/EVEN SEMESTER)

Course File

Format B

CONTENT BEYOND THE SYLLABUS

TITLE : Sampling Theorem, Filter Design.

OBJECTIVE : To understand the students in the virtual lab & theorems related to that

METHODOLOGY : BB / PPT / Virtual lab.

EVALUATION : Questioning

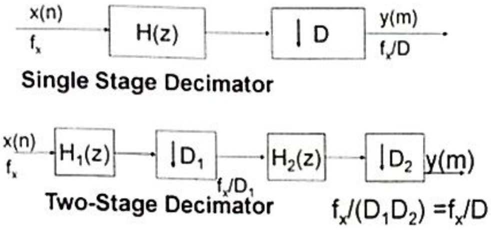
DATE OF COMPLETION: 1/11/21

7/12/21

STAFF INCHARGE SIGN

Sample Rate Conversion by Stages

Sometimes it is computationally more efficient to perform sample rate conversion in multiple stages.



Approach

- The signal is decimated by D_1 in the first stage and D_2 in the second stage, giving an overall decimation factor of $D=D_1D_2$.

- The filters $H_1(z)$ and $H_2(z)$ can be designed with wider transition bandwidth specifications since the overall filter response is narrow.

Advantages/Disadvantages of Sample Rate Conversion by Stages

- Reduced Computation
- Fewer coefficients reduces finite word-length problems.
- Drawback: original decimation and interpolation ratios cannot always be easily factored into suitable numbers.
- More (likely acceptable) passband aliasing

Example Conversion by Stages

Input Signal:

Assume a bandlimited digital audio signal.

Bandwidth =4kHz, Sampling rate, F_s=8kHz, D=50

Decimation Lowpass Filter:

Passband: 0 to 75 Hz Peak passband ripple $\delta_p=10^{-2}$

Transition band: 75 to 80 Hz Peak stopband ripple $\delta_s=10^{-4}$

Example Conversion by Stages (2)

The approximate length of an FIR filter, according to Kaiser formula, is:

$$N = \frac{-20\log_{10}\sqrt{\delta_p\delta_s} - 13}{14.6\Delta f}$$

Where $\Delta f = (f_s-f_p)/F_s$ is the normalized transition BW of the filter; f_s and f_p are passband and stopband edge frequencies respectively.

Example Conversion Single Stage (3)

Substituting the numerical values,

$$N = \frac{-20\log_{10}\sqrt{10^{-6}} - 13}{14.6(5/8000)} \approx 5150$$

Lowpass FIR filter has a length of 5150

The number of multiplications per sec is:

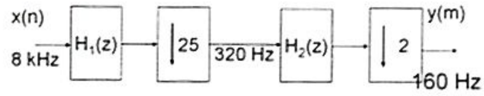
$$M_{sec} = 5150 * \frac{8000}{50} = 824000$$

Multistage Sample Rate Conversion Example (4)

The sample rate can be lowered to 160 Hz and still maintain the information. Hence the decimation factor is can be calculated as

$$D = \frac{f_s}{(2)(80)} = 50$$

Example Conversion by Stages (5)



Two-Stage Decimator

Let the decimation be achieved in two stages. Let D_1=25 and D_2=2.

Multistage Sample Rate Conversion Example (6)

- $H_1(z)$ and $H_2(z)$ are cascaded the overall linear passband specs can be halved between them.
- Stopband specs can remain same for both since the cascade filters can only increase the overall stopband attenuation.
- In choosing stopband edge, there is a tradeoff between the filter length and amount of tolerable aliasing in the passband

Multiple Stage Conversion Example (7)

Stage 1: Filter Specifications

Passband edge, f_p=75 Hz.

Peak passband ripple, δ_p= 0.005=(10^{-2}/2)

Stopband edge, f_s=240 Hz (=320-80),

Peak stopband ripple, $\delta_s = 10^{-4}$

Sampling Rate, F_s=8 kHz

Multistage Sample Rate Conversion Example (8)

The choice of passband edge frequency depends on the decimation factor. In this case, due to the attenuation by filter, aliasing energy in the band from 240 Hz to 320 Hz will be small compared to that in 160 to 240 Hz, i.e, the stopband aliases on top of the passband.

Multistage Sample Rate Conversion Example (9)

f_p= output sample rate of decimator stage - desired transition band upper frequency

By Kaiser Formula, the approximate FIR filter length of the first stage:

$$N_1 = \frac{-20\log_{10}\sqrt{(0.005)(10^{-4})} - 13}{14.6(240-75)/8000}$$

$$\Rightarrow N_1 \approx 166$$

Multistage Sample Rate Conversion Example (10)

Stage 2: Filter specifications

Passband edge, f_p: 75 Hz
Passband ripple, $\delta_p = 0.005(=10^{-2}/2)$

Stopband edge, f_s: 80 Hz
stopband ripple, $\delta_s = 10^{-4}$

Sampling frequency, $F_s = 320$ Hz

Multistage Sample Rate Conversion Example (11)

Approximate length of this second stage FIR

$$N_1 = \frac{-20\log_{10}\sqrt{(0.005)(10^{-4})}-13}{14.6(80-75)/320}$$

$$\Rightarrow N_1 \approx 219$$

Multiplication per sec for these two stage

$$M_{sec} = M_{1,sec} + M_{2,sec}$$

$$= 166 * \frac{8000}{25} + 219 * \frac{320}{2}$$

$$= 88160$$

Multistage Sample Rate Conversion Example (12)

Summary:

Implementation approach	Single-stage	Two-stage
Number of filter coefficients	5150	385
Number of multiplications per sec	824000	88160

Computation is reduced by a factor of 8 with the two stage implementation.

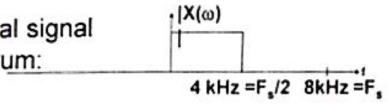
Multistage Sample Rate Conversion Example (13)

Key to computational advantage:

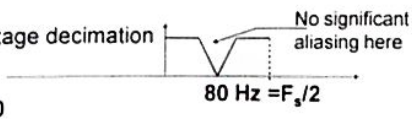
Aliasing is allowed, especially in the bands of no interest.

Multistage Sample Rate Conversion Example (14)

Original signal spectrum:



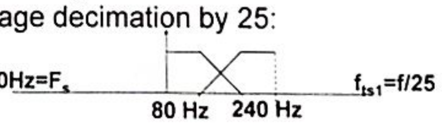
Single stage decimation by 50:



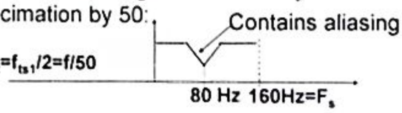
f_{ss}=f/50
160 Hz=F_s

Multistage Sample Rate Conversion Example (15)

Two-stage decimation: After first stage decimation by 25:



After second stage decimation by 2: Overall decimation by 50:



Digital Filter Banks

Applications

* Communications

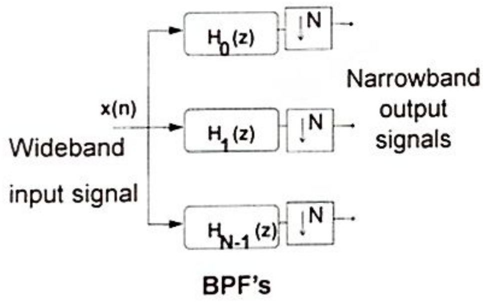
  Convert FDM to TDM signals
* Speech and Image Coding for Data Reduction

  More bits to represent the lower band where most of energy is located
* Computational/Structural Advantages for Wideband Processing

Implementation of Digital Filter Banks (DFBs)

- Two basic structures:
  - DFB Analyzer: Splits signals into subbands
  - DFB Synthesizer: Combines the subbands into a wideband signal

Analysis Filter Bank



Implementation of Digital Filter Bank Analyzer

- The input signal is decomposed into N channel signals band pass filters with center frequencies

$$\omega_k = \frac{2\pi k}{N} \qquad ; k = 0,1,2,\ldots,N-1$$

- Each of these filtered signals is decimated to reduce the sample rate for each channel

Implementation of Analysis Filter Bank

- Bandpass filtering can be viewed as down conversion and lowpass filtering. The frequency response of the bandpass filter is

$$H_k(\omega) = H_0\left(\omega - \frac{2\pi k}{N}\right) \qquad ,k = 0,1,2,\ldots,N-1$$

or

$$h_k(n) = h_0(n)\,e^{\frac{j2\pi nk}{N}} \qquad ,k = 0,1,2,\ldots,N-1$$

- $H_k(z)$, k=0,1, ..., N-1 is a uniform filter bank, $H_0(z)$ is a LPF.

Implementation of Analysis Filter Bank (2)

- Channel outputs are assumed to be critically sampled, i.e., the sample rate of the wideband signal is N times the sample rate of each narrow band signal.
- We will consider DFT and IDFT filter bank structures for the analyzer and the synthesizer respectively