University of Mumbai
Program: **Electronics and Telecommunication Engineering**
Curriculum Scheme: Rev2019
Examination: TE Semester V
Course Code: ECC-504 and Course Name: Discrete Time Signal Processing

Time: 2 hour 30 minutes Max. Marks: 80

===

Q1. Choose the correct option for following questions. All the Questions are compulsory and carry equal marks

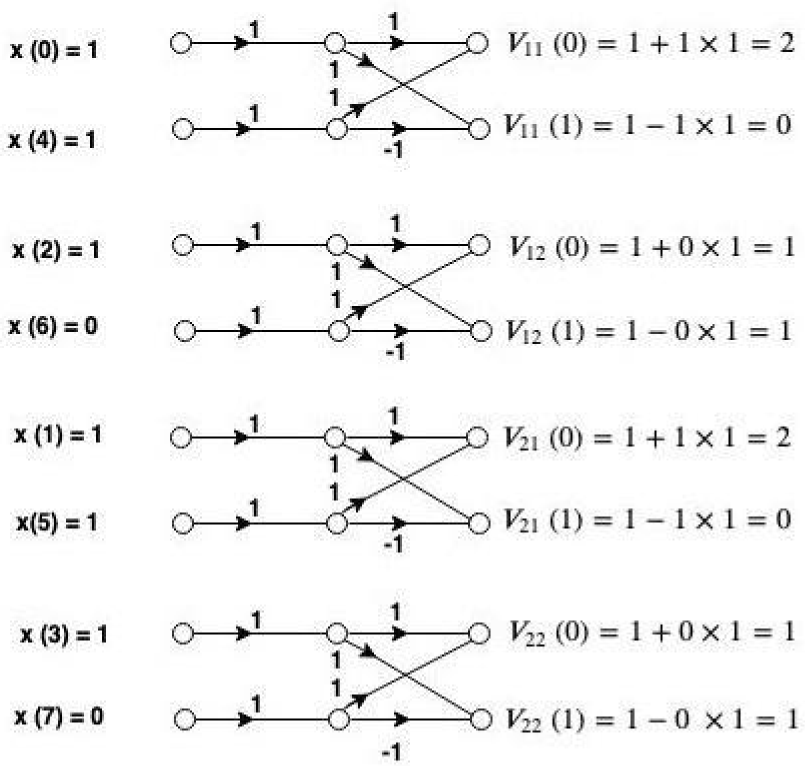
Question Number	**Correct Option (Enter either 'A' or 'B' or 'C' or 'D')**
Q1.	**C.**
Q2.	**A.**
Q3.	**A.**
Q4	**A.**
Q5	**C.**
Q6	**A.**
Q7	**B.**
Q8.	**A.**
Q9.	**B.**
Q10.	**A.**

Q.2

A. Find DFT of the following sequence using DIT FFT algorithm. x(n)={1,1,1,1,1,1,1,0}

Expected Solution:

N=8=23N=8=23
Given, x(n)={1,1,1,1,1,1,1,0}x(n)={1,1,1,1,1,1,1,0}
Step 1: x(n)x(n) is written in bit reversed order i.e. {1,1,1,1,1,1,1,0}{1,1,1,1,1,1,1,0} is the input for step 1.1..(2 Marks)
The phase factor for step 1 is W02=e0=1W20=e0=1
The butterfly computations for step 1 are:

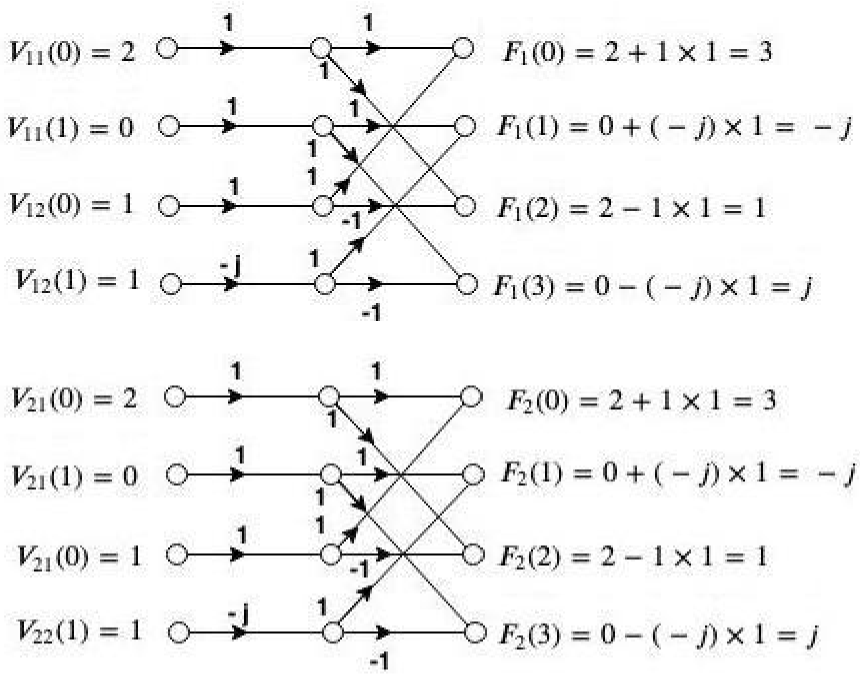


Output of step 1 is V(k)={2,0,2,0,2,0,1,1}V(k)={2,0,2,0,2,0,1,1}

Step 2: Output of step 1 forms the input for second step.(2 Marks)

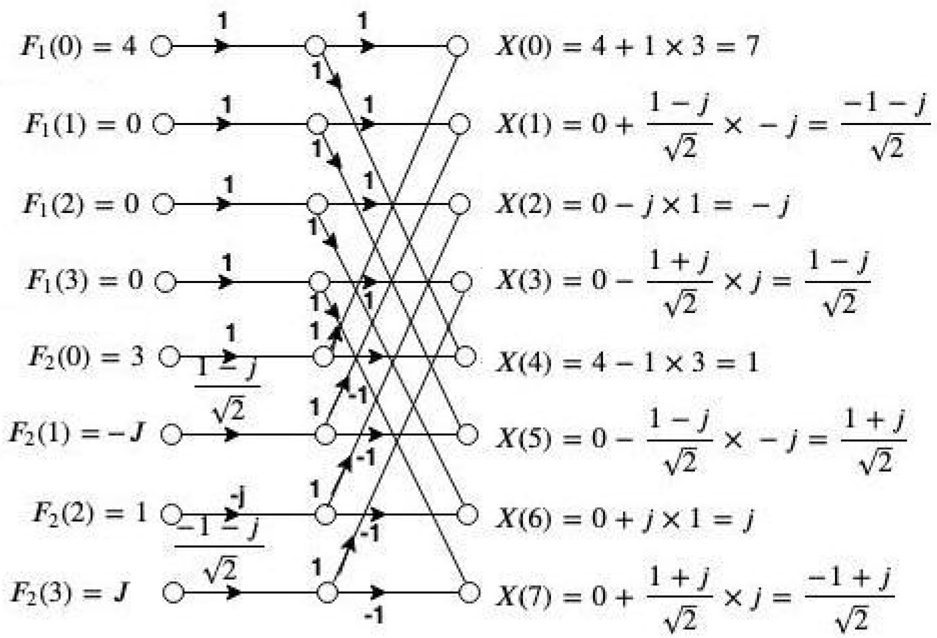
The phase factor for step 2W04=e0=12W40=e0=1 and W14=e−j2π/4=cos2π4−jsin2π4=0−j=−jW41=e−j2π/4=cos2π4−j sin2π4=0−j=−j

The butterfly computations for step 2 are:



Output of step 2 is F(k)={4,0,0,0,3,−j,1,j}F(k)={4,0,0,0,3,−j,1,j}
Step 3: Output of step 2 forms the input for third step.(2 Marks)

Phase factors for step 3 are W08=e0=1W80=e0=1
W18=e−j2π/8=cos2π8−jsin2π8=12√−j12√W81=e−j2π/8=cos2π8−jsin2π8=12−j12
W28=e−j4π/8=cos4π8−jsin4π8=0−j×1=−jW82=e−j4π/8=cos4π8−jsin4π8=0−j×1=−j
W38=e−j6π/8=cos6π8−jsin6π8=−12√−j12√W83=e−j6π/8=cos6π8−jsin6π8=−12−j12
The butterfly computations for step 3 are:



Output of step (2 Marks)
3 ={7,−12√−12√j,−j,12√−12√j,1,12√+12√j,j,−12√+12√j}={7,−12−12j,−j,12−12j,1,12+12j,j,−12+12j}
Hence, the DFT of given sequence x(n)x(n) is (2 Marks)
X(k)=DFT{x(n)}={7,−12√−12√j,−j,12√−12√j,1,12√+12√j,j,−12√+12√j}X(k)=DFT{x(n)}={7,−12−12j,−j,12−12j,1,12+12j,j,−12+12j}

Q.2
B. Differentiate between Butterworth and chebyshev filter

Butterworth Filter	**Chebyshev Filter**
For a particular desired specification of a digital filter the order of butterworth filter will be higher than chebyshev filter	For a particular desired specification of a digital filter the order of chebyshev filter will be lower as compared to butterworth filter

For a particular specification butterworth filter requires more hardware	For a particular specification chebyshev filter requires less hardware
This filter has cutoff frequency not equal to passband frequency $\Omega c=\Omega p/(1A2p-1)12N\Omega c=\Omega p/(1Ap2-1)12N$	This filter has cutoff frequency equal to passband frequency $\Omega c=\Omega p\Omega c=\Omega p$
The transition band of butterworth filter is wider as compared to chebyshev filter	The transition band of chebyshev filter is narrow as compared to butterworth filter
All poles of filter will always lie on circle having radius $r=\Omega cr=\Omega c$	All the poles of a filter will lie on ellipse having major axis 'R', 'ζζ', minor axis 'r'
Butterworth filter has no ripples either in passband or stopband	Chebyshev filter will have ripples either in stop band or passband
Butterworth filter does not have any type	It has two type i.e Type 1= Ripple in passband and Type 2=Ripple in stopband

Q. 2

C. Explain application of DSP for ECG Signal Analysis

- Most of the biomedical signals are in audio range and DSP is widely used in many biomedical applications.
- The signals from the body, are captured using electrodes. But the major problem is addition of noise in such signals.
- To remove this noise signals; digital filters are used.

Fetal ECG Monitoring

- Electrical activity of a heart is called as electrocardiogram (ECG).
- Fetal ECG represents electrical activity of the baby's heart. It is similar to the adult ECG waveform.
- The Fetal ECG (FECG) contains five peaks namely PQRST. This waveform is shown in

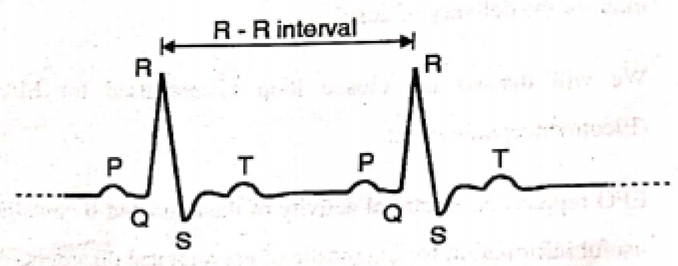


Fig. 7.1.1

Fig.7.1 .1

- The Fetal Heart Rate (FHR) is obtained from R-to-R interval of the waveform shown in Fig. 7.1 .1 .
- ECGECG is having PP -wave, QRS complex wave and T wave.

- Instantaneous heart rate is obtained by multiplying time interval between R-to-R (in miliseconds) by 60,00060,000 .
- A suitable DSP algorithm is used successive QRS complex waveform and from this, R-to-R interval is calculated. It gives value of corresponding FHR.
- The shape of QRS complex waveform changes from patient to patient.
- The obtained ECG signal is compared with a known standard ECG and then the locations of QRS complex in the obtained ECG can be determined on the basis of similarity.

DSP based Closed Loop Controlled Anaesthesia

- In case of surgery; anaesthesia is injected to the patient body.
- A proper amount of drug to induce anaesthesia is required to inject in the patient body.
- Extra amount of drug produces side effects, as well as less amount of drug can produce psychological consequences for a long term.
- So it is required to make necessary changes in the dose of anaesthetic drugs and to control the anaesthesia.
- Using DSP, a closed loop system can be designed to monitor the does of anaesthesia and it is an automatic system.
- But, this system requires an accurate measurement of the depth of anaesthesia and it requires a feedback signal to monitor the delivery of dose.
- We will discuss the closed loop system used for EEG (Electroencephalogram).
- EFG represents, electrical activity of the brain and it contains useful information for diagnostic of neurological disorders.
- From the EEG signals; the features like bispectral index Audiotory Evoked Response (AER) are extracted and then the depth of anaesthesia is determined.
- Basically AER signal gives information from consciousness to unconsciousness condition of the patient.
- But these AER signals are usually small and they are mixed up with EEG.
- So a proper system is required to extract these AER signals and then process it.
- A bispectral index is obtained by performing spectrum analysis of EEG.
- The signal I (k) operates infusion pump which controls the flow rate of drug.
- It gives information about the changes in frequency components of EEG at different consciousness levels.
- A block diagram of DSP based closed controlled anaesthesia system is shown in Fig. 7.1 .2 .
- EEG electrodes are placed on the scalp of the patient; which generates EEG signals.
- Using suitable signal processing method; noise contained in EEG is reduced.
- EEG monitor produces EEG index (bispectral index) and it gives the measurement of drug induced in patient body.
- This EEG index is used as a feedback signal and it is compared with the target EEG index.
- The target EEG index is a standard value which is determined by considering many biomedical factors.
- The difference between target EEG index and measured EEG produces an error signal, e (k).
- This signal is applied to PID (Proportional-Integral- Derivative) control; to generate the required control signal, C(k).
- The signal C(k)C(k) is applied to pharmacokinetic // pharamaco- dynamic model and this model generates a signal I (k) to determine the rate at which, drug is injected in to the

patient body.

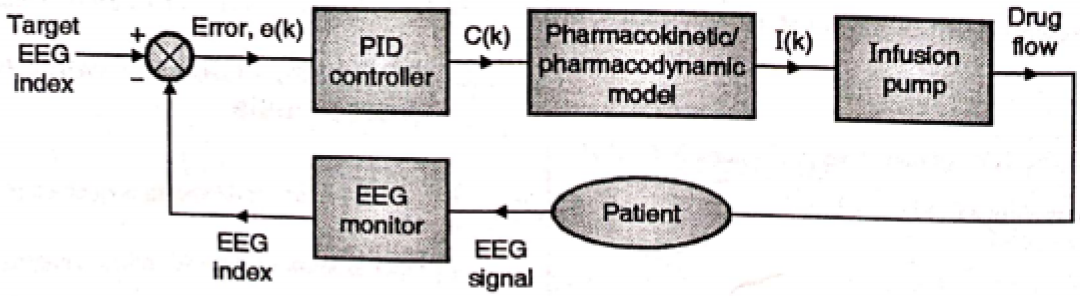


Fig. 7.1.2

Q.3

A. Differentiate IIR and FIR systems.

Sr. No.	**IIR systems**	**FIR systems**
1.	IIR stands for infinite impulse response systems	FIR stands for finite impulse response systems
2.	IIR filters are less powerful than FIR filters, & require less processing power and less work to set up the filters	FIR filters are more powerful than IIR filters, but also require more processing power and more work to set up the filters
3.	They are more easy to change "on the fly”.	They are also less easy to change "on the fly" as you can by tweaking (say) the frequency setting of a parametric (IIR) filter
4.	These are less flexible.	Their,greater power means more flexibility and ability to finely adjust the response of your active loudspeaker.
5.	It cannot implement linear-phase filtering.	It can implement linear-phase filtering.
6.	It cannot be used to correct frequency-response errors in a loudspeaker	It can be used to correct frequency-response errors in a loudspeaker to a finer degree of precision than using IIRs

7.	IIRs can provide good resolution even at low frequencies.	FIRs can be limited in resolution at low frequencies, and the success of applying FIR filters depends greatly on the program that is used to generate the filter coefficients
8.	Usage is generally more easier than FIR filters.	Usage is generally more complicated and time-consuming than IIR filters
9.	Finite Non-Zero Samples ← IIR Filter → Infinite Non-zerp Samples	Finite Non-Zero Samples ← FIR Filter → Finite Non-zero Samples
10.	IIR filter uses current input sample value, past input and output samples to obtain current output sample value.	FIR filter uses only current and past input digital samples to obtain a current output sample value. It does not utilize past output samples.
11.	Simple IIR equation is mention below.,$y(n)= b(0)x(n) + b(1)x(n-1) + b(2)x(n-2) + b(3)x(n-3) + a(1)y(n-1) + a(2)y(n-2) + a(3)y(n-3)$	Simple FIR equation is mention below. $y(n)= h(0)x(n) + h(1)x(n-1) + h(2)x(n-2) + h(3)x(n-3) + h(4)x(n-4)$
12.	Transfer function of IIR filter will have both zeros and poles and will require less memory than FIR counterpart	Transfer function of FIR filter will have only zeros, need more memory
13.	IIR filters are not stable as they are recursive in nature and feedback is also involved in the process of calculating output sample values.	FIR filters are preferred due to its linear phase response and also they are non-recursive. Feedback is not involved in FIR, hence they are stable
14.	IIR filter need more power due to more coefficients in the design.	FIR filter consume low power
15.	IIR filters have analog equivalent	FIR have no analog equivalent.
16.	IIR filters are more efficient	FIR filters are less efficient
17.	IIR filters are used as notch(band stop),band pass functions.	FIR filters are used as anti-aliasing,low pass and baseband filters

18.	IIR filter need lower order than FIR filter to achieve same performance	FIR filter need higher order than IIR filter to achieve same performance.
19	Delay is less than FIR filter.	Delay is more than IIR filter.
20.	It has higher sensitivity than FIR filter	It has lower sensitivity than IIR filter

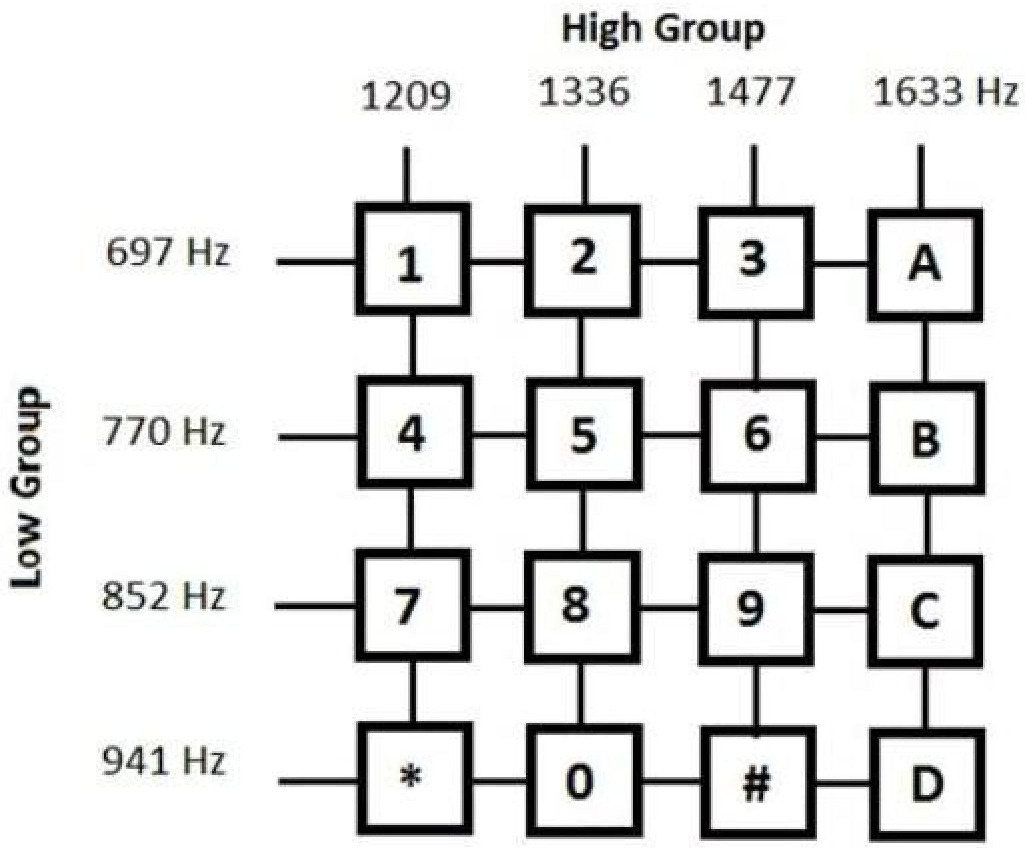
Q. 3

B. Write a short note on Dual Tone Multi-Frequency Signal Detection.

• Dual Tone Multi-Frequency or DTMF is a method for instructing a telephone switching system of the telephone number to be dialed, or to issue commands to switching systems or related telephony equipment.

• The DTMF dialing system follow the technique proposed by AT&T in the 1950s called MF (Multi-Frequency) which was deployed within the AT&T telephone network to direct calls between switching facilities using in-band signaling.

• The DTMF system uses eight different frequency signals transmitted in pairs to represent sixteen different numbers, symbols and letters. This table shows how the frequencies are organized:



• The frequencies used were chosen to prevent any harmonics from being incorrectly detected by the receiver as some other DTMF frequency. The transmitter of a DTMF signal simultaneously sends one frequency from the high-group and one frequency from the low-group.

• This pair of signals represents the digit or symbol shown at the intersection of row and column in the table. For example, sending 1209Hz and 770Hz indicates that the "4" digit is being sent.

• At the transmitter, the maximum signal strength of a pair of tones must not exceed +1 dBm, and the minimum strength is -10.5 dBm for the low-group frequencies and -8.5 dBm for the high-group frequencies.

Labeling of DTMF numeric digits

• The DTMF telephone keypad is laid out in a 4×4 matrix of push buttons in which each row represents the low frequency component and each column represents the high frequency component of the DTMF signal. Pressing a key sends a combination of the row and column frequencies.

• For example, the key 1 produces a superimposition of tones of 697 and 1209 hertz (Hz). Initial pushbutton designs employed levers, so that each button activated two contacts. The tones are decoded by the switching center to determine the keys pressed by the user.

• DTMF was originally decoded by tuned filter banks. By the end of the 20th century, digital signal processing became the predominant technology for decoding. DTMF decoding algorithms often use the Goertzel algorithm to detect tones.

Q.3

C. Design a FIR filter using window method for following specification. Use hamming window of length.
Step-1: Identify the specification of filter 2 Marks
Step-2: Calculate the Inverse Fourier Transform of H(ω) 2 Marks
Step-3: Calculation of window Response W(n) 1 Mark
Step-4: Calculation of window Response h(n) 1 Mark
Step-5: Calculate of filter Transfer Function 2 Marks
Step-6: Realization Structure 2 Marks

Q.4
A.What is multirate DSP? Where it is required?
Multi rate DSP :

• Multi rate simply means "multiple sampling rates". A multi rate DSP system uses multiple sampling rates within the system. Whenever a signal at one rate has to be used by a system that expects a different rate, the rate has to be increased or decreased, and some processing is required to do so. Therefore "Multi rate DSP" really refers to the art or science of changing sampling rates.

• Multi-rate processing finds use in signal processing systems where various sub-systems with differing sample or clock rates need to be interfaced together. At other times multi-rate processing is used to reduce computational overhead of a system. For example, an algorithm requires k operations to be completed per cycle. By reducing the sample rate of a signal or system by a factor of M, the arithmetic bandwidth requirements are reduced from kfs operations to kfs/M operations per second.

• The most immediate reason is when you need to pass data between two systems which use incompatible sampling rates. For example, professional audio systems use 48 kHz rate, but consumer CD players use 44.1 kHz; when audio professionals transfer their recorded music to CDs, they need to do a rate conversion.

• But the most common reason is that multirate DSP can greatly increase processing efficiency (even by orders of magnitude!), which reduces DSP system cost..

Multirate consists of:

• Decimation: To decrease the sampling rate,

• Interpolation: To increase the sampling rate, or,

• Resampling: To combine decimation and interpolation in order to change the sampling rate by a fractional value that can be expressed as a ratio. For example, to resample by a factor of 1.5, you just interpolate by a factor of 3 then decimate by a factor of 2 (to change the sampling rate by a factor of 3/2=1.5.)

Applications:

1. Dual-Tone Multifrequency Signal Detection

Dual-tone multifrequency (DTMF) signaling, increasingly being employed worldwide with push-button telephone sets, offers a high dialing speed over the dial-pulse signaling used in conventional rotary tele- phone sets. In recent years, DTMF signaling has also found applications requiring interactive control, such as in voice mail, electronic mail (e-mail), telephone banking, and ATM machines

2. Spectral Analysis of Sinusoidal Signals

An important application of digital signal processing methods is in determining in the discrete-time do- main the frequency contents of a continuous-time signal,
more commonly known as spectral analysis. More specifically, it involves the determination of either t he energy spectrum or the power spectrum of the signal.

3. Musical Sound Processing

Almost all musical programs are produced in basically two stages. First, sound from each individual instrument is recorded in an acoustically inert studio on a single track of a multitrack tape recorder.

Then, the signals from each track are manipulated by the sound engineer to add special audio effects and are combined in a mix-down system to finally generate the stereo recording on a two-track tape recorder.

The audio effects are artificially generated using various signal processing circuits and devices, and they are increasingly being performed using digital signal processing techniques.

4. Signal Compression

As mentioned earlier, signals carry information, and the objective of signal processing is to preserve the information contained in the signal and extract and manipulate it when necessary. Most digital signals encountered in practice contain a huge amount of data.

Q.4

B. Write down the design steps for FIR filter using the window techniques. Compare windows.

The design steps for FIR filter using window techniques are

1. The specifications required for FIR filter design are-

a) Type of Filter: Low Pass, High Pass, band-pass and band-stop filters.

b) Order N (or length M) of the filter, N=M−1N=M−1
c) Desired Frequency Response

2. Desired Impulse Response hd(n)hd(n) is computed using above specifications by Inverse DTFT of Hd(ejw)Hd(ejw)
3. Desired Impulse Response hd(n)hd(n) is of infinite duration. To make if of finite duration, it is multiplied with a suitable window function
4. Transfer Function of the FIR Filter is obtained by taking Z Transform of Finite Impulse Response h(n).

Compare windows

Sr No	Type of window	Definition of Window Sequence W(n)= ⎧⎨⎩ A00≤n≤N−1 else , where W (n)={A0≤n≤N−10 else , where	Approximate Transition width of Main Lobe	Magnitude of the Peak of First Side Lobe	Minimum Stop-band Attenuation
1	Rectangular	A=1A=1	4πN4πN	−13dB−13dB	21dB21dB
2	Traingular	A=1−|n−α|αA=1−|n−α|α	8πN8πN	−25dB−25dB	25dB25dB
3	Hanning	A=0.5−0.5cosπnαA=0.5−0.5cosπnα	8πN8πN	−31dB−31dB	44dB44dB
4	Hamming	A=0.54−0.46cosπnαA=0.54−0.46cosπnα	8πN8πN	−41dB−41dB	53dB53dB
5	Blackman	A=0.42−0.5cosπnα+0.08cos2πnαA=0.42−0.5cosπnα+0.08cos2πnα	12πN12πN	−57dB−57dB	74dB

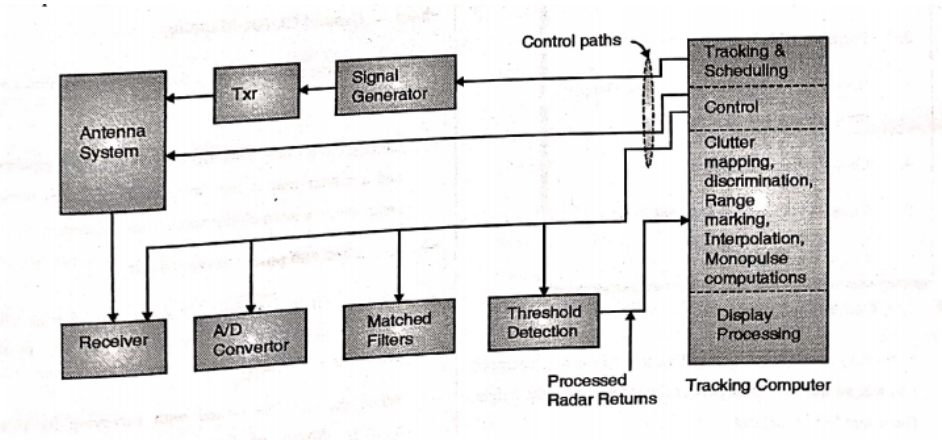
Q.4

C. Explain application of DSP for Radar signal processing

- RADAR transmits radio signals at distant objects and analyzes reflection.
- Data gathered can include the position and movement of the object, also radar can identify the object through its "signature" - the distinct reflection it generates.
- There are many forms of RADAR - such as continuous CW), Doppler, ground penetrating or synthetic aperture; and they're used in many applications, from air traffic control to weather prediction.
- In the moderm Radar systems digital ignal processing DSP is used extensively. At the transmitter end, it generates and shapes the transmission pulses, controls the antenna beam patter while at the receiver, DSP performs many complex tasks, including STAP (space time adaptive processing)- the removal of clutter, and beamforming (electronic guidance of direction).
- The front end of the receiver for RADA is still often analog due the high frequencies involved. With fast ADC convertors - often multiple channel, complex IF signals are digitized. However,

digital technology is coming closer to the antenna. We may also require fast digital interfaces to detect antenna position, or control other hardware.

- The main task of a radar's signal processor is to make decisions. After a signal has been transmitted, the receiver starts receiving return signals, with those originating from near objects arriving first because time of arrival translates into target range.
- The signal processor places a raster of range bins over the whole period of time, and now it has to make a decision for each of the range bins as to whether it contains an object or not.
- This decision-making is severely hampered by noise. Atmospheric noise enters into the system through the antenna, and all the electronics in the radar's signal path produces noise too.



- Fig. 7.3.1 : Block Diagram of a Modern Radar system 5 marks

Major blocks of Modern Radar System-5 marks

- The major components of modern radar are the antenna, the tracking computer and the signal generator.
- The tracking computer in the modern radar does all the functions. By scheduling the appropriate antenna positions and transmitted signals as a function of time, keeps track of targets and running the display system.
- Even if atmospheric attenuation can be neglected, the return from a distant object is incredibly weak. Target returns often are no stronger than twice the average noise level, sometimes even buried under it.
- It is quite difficult to define a threshold for the decishold whether a given peak is noise or a real target. If the threshold is too high then existing targets are suppressed, that is, the probability of detection (PD) will drop.
- If the threshold is too low then noise peaks will be reported as targets, that is, the probability of false alarms (PFA) will rise.
- A common compromise is to have some 90%% probability of detection and a false alarm rate of 10−610−6 . It maintains a given PFA known as CFAR, for Constant False Alarm Rate. Rather than keeping the threshold at a fixed point, CFAR circuitry inspects one range bin after the other and compares the signal level found there with the signal levels found in its neighboring bins. If the noise level is rather high in all of these (eg, because of precipitation) then the CFAR circuit will raise the threshold accordingly.