1.(a)what are the difference between DIT and DIF?

X(K) computing an n points DFT requires multiplication & addition.

Q. Distinguish between DIT & DIF.

The full meaning of DIT & DIF is decimation in time and decimation in frequency respectively.

For DIT it is considered the individual samples (actually pair of samples) in time domain. And find their frequency parts and further to find FFT.

For DIF it is considered the indevidual samples in frequency domain. And combined them to come out with the actual FFT.

Butterfly Operation: The basic computational unit of the FFT shown in fig 7.5(a) is called

smile

(b)Write down the Dirichlet conditions.

Pg-430

domain.

Q. Write down Dirichlet condition.

soln: There are some conditions for the existence of fourier series and fourier transform of the periodic and non-periodic wave form respectively and those are called the Dirichlet condition.

Dirichlet condition of periodic wave form:

① If it is discontinou there are only a finite number of discontinities in the period T.

② It has a finite average value over the period T.

③ It has a finite number of positive and negative maxima in the period T.

Pg-430 Dirichlet Conditions of Non-periodic.

① $x(t)$ is absoulutely integrable

$$\int_{-\infty}^{+\infty} |x(t)|\, dt < \infty$$

② The number of maxima, minim a and the number of discontinuities of $x(t)$ in every finite interval is finite. $\boxed{15\text{-}02\text{-}2017}$

smile

(c)State and explain properties of fourier transform

22-02-2017

Properties of fourier transform:

1. linearity:

if $x_1(n) \xleftrightarrow{F} X_1(\omega)$ and $x_2(n) \xleftrightarrow{F} X_2(\omega)$

Then $a_1x_1(n) + a_2x_2(n) \xleftrightarrow{F} a_1X_1(\omega) + a_2X_2(\omega)$

2. Time shifting:

if $x_1(n) \xleftrightarrow{F} X(\omega)$ then $x(n-k) \xleftrightarrow{F} e^{-j\omega k} X(\omega)$

3. Time reversal

if $x(n) \xleftrightarrow{F} X(\omega)$ then $x(-n) \xleftrightarrow{F} X(-\omega)$

4. Convolution theorem,

if $x_1(n) \xleftrightarrow{F} X_1(\omega)$ and $X_2(n) \xleftrightarrow{F} X_2(\omega)$

then, $x(n) = x_1(n) * x_2(n) \xleftrightarrow{F} X(\omega)$

The convolution formula is,

$$x(n) = x_1(n) * x_2(n) = \sum_{k=-2}^{\infty} x_1(k)\, x_2(n-k)$$

smile

Co-relation theorem:

(V) If $x_1(n) \xleftrightarrow{f} X_1(\omega)$ and $x_2(n) \xleftrightarrow{f} X_2(\omega)$

Then $r_{x_1 x_2}(n) \xleftrightarrow{f} S_{x_1 x_2}(\omega) = X_1(\omega) X_2(-\omega)$

In that case

$$r_{x_1 x_2}(n) = \sum_{k=-\infty}^{\infty} x_1(k)\, x_2(k-n)$$

(VI) **Winer khintching theorem**

Let $x(n)$ be a real integer then,

$r_{xx}(l) \xleftrightarrow{f} S_{xx}(\omega)$

(VII) **Frequency shifting**

If $x(n) \xleftrightarrow{f} X(\omega)$ then

$e^{i\omega_0 n} x(n) \xleftrightarrow{f} X(\omega-\omega_0)$

(VIII) **Modulation theorem:**

If $x(n) \xleftrightarrow{f} X(\omega)$

Then $x(n) \cos\omega_0 n \xleftrightarrow{f} \frac{1}{2}\left[X(\omega+\omega_0) + X(\omega-\omega_0)\right]$

smile

(IX) Parseval's theorem:

If $x_1(n) \xleftrightarrow{F} X_1(\omega)$ and $x_2(n) \xleftrightarrow{F} x_2(\omega)$

then, $$\sum_{n=-\infty}^{\infty} x_1(n) * x_2^{*}(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} X_1(\omega) X_2^{*}(\omega)\, d\omega$$

(X) Multiplication of two sequences:

If $x_1(n) \xleftrightarrow{F} X_1(\omega)$ and $x_2(n) \xleftrightarrow{F} X_2(\omega)$

Then $x_3(n) = x_1(n)\, x_2(n) \xleftrightarrow{F} X_3(\omega)$

$$= \frac{1}{2\pi} \int_{-\pi}^{\pi} X_1(\lambda)\, x_2(\omega\,\lambda)\, d\lambda$$

Properties of DFD

smile

(d)Define Butterfly operation and explain a radix-2 decimation in time FFT.

Question: How many multiplications and additions are required to compute N-point DFT using radix-2 FFT? [NU: 2011]

Answer:

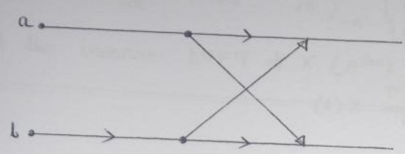


Fig: Butterfly Operation

Infig - an 8 point decimation-in-time FFT. This is so called because the alternate time sample are decimated by the process. We can see from this figure that an N-point FFT contains N/2 butterflies per stage with $\log_2 N$ stages giving a total of $(N/2)\log_2 N$ butterflies.

This method of FFT calculation involves $(N/2)\log_2 N$ complex multiplications and $N\log_2 N$ complex multiplications and $N\log_2 N$ complex additions providing a reduction in computations.

We observe that N-point DFT is computed using two length (N/2) DFTs and some extra operations called butterfly operations.

Following figure shows one of the butterfly operation. Each of those butterflies is a length 2-DFT breaking a length N DFT up into a two-dimentional length $(2 \times N/2)$ DFT

It involves one complex multiplication and two complex additions.

smile

2(a)Describe different process of inverse Z transform.

1- The inverse z-transform By contour integration

The inverse z-transform can be directly be determined from ~~cot~~ contour integral by using cauchy-theorem.

In this method z-transform we have,

$$x(n) = \frac{1}{2\pi j} \oint_c X(z)\, z^{n-1}\, dz$$

$$= \sum_{\text{all poles } \{z_i\} \text{ inside } c} [\text{residue of } X(z)\, z^{n-1} \text{ at } z = z_i]$$

$$= \sum_i (z - z_i)\, X(z) \cdot z^{n-1} \Big|_{z = z_i}$$

We get the output form contour integration by using cauchy's theorem.

smile

(2) Inverse z-transform by power series expansion:

let a z-transform z(z) with its corresponding ROC, we can expand x(z) into a power series of the form,

$$x(z) = \sum_{n=-\infty}^{\infty} c_n z^{-n}$$

With converts in the given ROC, then by the uniqueness of z-transform,

$$x(n) = c_n \text{ for all } n$$

when x(z) is rational the expansion can be perform by long divisor.

smile

3.4.3 ③ The inverse z-transform by partial fraction expansion: / Table lookup method

In the table lookup method we attain to express the function $X(z)$ as a linear combination.

$$X(z) = \alpha_1 X_1(z) + \alpha_2 X_2(z) + \cdots + \alpha_k X_k(z)$$

where $X_1(z), X_2(z), \ldots, X_k(z)$ are expression with inverse z transform $x_1(n), x_2(n) \ldots \ldots x_k(n)$ respectively available in a table of z-transform pair if such a decomposition is possible then $x(n)$, the inverse z-transform of $X(z)$, can easily be found using the linear property as,

$$x(n) = \alpha_1 x_1(n) + \alpha_1 x_2(n) + \cdots + \alpha_k x_k(n)$$

This approach is particularly useful if $X(z)$ is rational function.

smile

(b)

Define the inverse Z transform of 6

$$X(z) = \frac{1}{1 - 1.5z^{-1} + 0.5z^{-2}}$$

(a) ROC : $|z| > 1$

(b) ROC : $|z| < 0.5$

Answer: a) Since the ROC is the exterior of a circle, we expect $x(n)$ to be a causal signal. Thus we seek a power series expansion in negative power of z. By dividing the numerator of $X(z)$ by its denominator, we obtain the power series,

$$1 - \frac{3}{2}z^{-1} + \frac{1}{2}z^{-2} \,\Big)\, 1 \,\Big(\, 1 + \frac{3}{2}z^{-1} + \frac{7}{4}z^{-2} + \cdots$$

$$\underline{1 - \frac{3}{2}z^{-1} + \frac{1}{2}z^{-2}}$$
$$(-)\quad(+)\quad(-)$$

$$\frac{3}{2}z^{-1} - \frac{1}{2}z^{-2}$$

$$\underline{\frac{3}{2}z^{-1} - \frac{9}{4}z^{-2} + \frac{3}{4}z^{-3}}$$
$$(-)\quad(+)\quad(-)$$

$$\frac{7}{4}z^{-2} - \frac{3}{4}z^{-3}$$

$$\underline{\frac{7}{4}z^{-2} - \frac{21}{8}z^{-3} + \frac{7}{8}z^{-4}}$$
$$(-)\quad(+)\quad(-)$$

$$\frac{15}{8}z^{-3} - \frac{7}{8}z^{-4}$$

$$- - - - -$$

$$- - - - -$$

$$\therefore\ x(n) = \left\{1, \frac{3}{2}, \frac{7}{4}, \frac{15}{8}, \frac{31}{16}, \ldots\right\}$$

b) In this case ROC is the interior of a circle. Consequently, the signal $x(n)$ is anticausal. To obtain a power series expansion in positive powers of z, we perform the long division in the following way.

smile

$$
\begin{array}{r}
2z^2+6z^3+14z^4+30z^5+62z^6+\cdots \\
\tfrac{1}{2}z^{-2}-\tfrac{3}{2}z^{-1}+1 \;\big)\; 1 \qquad\qquad\qquad\qquad \\
1-3z+2z^2 \qquad\qquad\qquad \\
\hline
3z-2z^2 \qquad\qquad\qquad \\
3z-9z^2+6z^3 \qquad\qquad \\
\hline
7z^2-6z^3 \qquad\qquad \\
7z^2-21z^3+14z^4 \qquad \\
\hline
15z^3-14z^4 \qquad \\
15z^3-45z^4+30z^5 \\
\hline
31z^4-30z^5 \\
-\;-\;-\;-\;- \\
-\;-\;-\;-\;-
\end{array}
$$

Thus,

$$X(z) = \frac{1}{1-\frac{3}{2}z^{-1}+\frac{1}{2}z^{-2}} = 2z^2+6z^3+14z^4+30z^5+62z^6+\cdots$$

In this case $x(n)=0$ for $n\geq 0$.

$$x(n) = \{\cdots,62,30,14,6,2,0,\underset{\uparrow}{0}\}$$

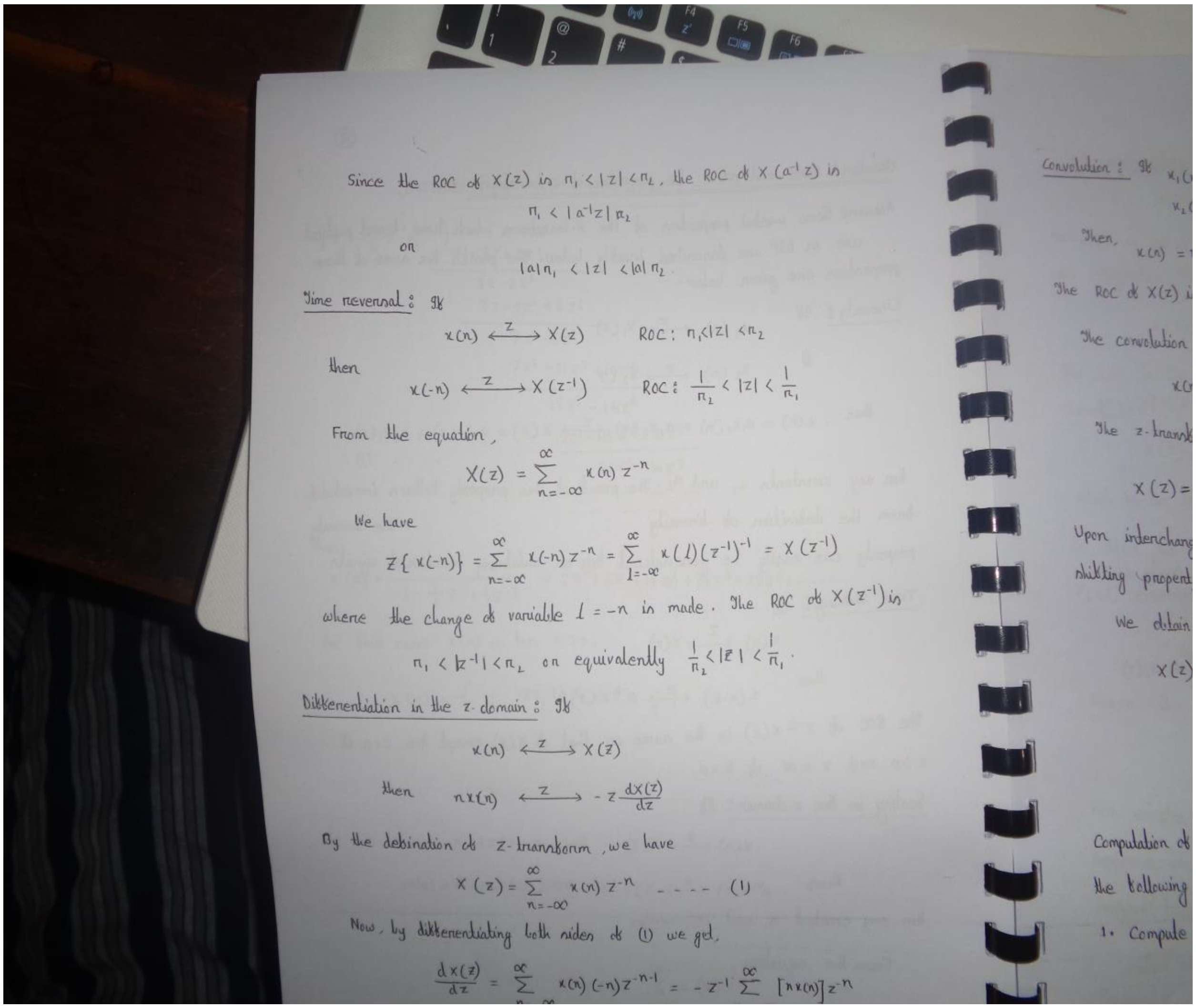
smile

(c)Prove the following properties of z transform

i)The reversal property

ii)The convolution property

iii)The scalling property



Since the ROC of $X(z)$ is $r_1 < |z| < r_2$, the ROC of $X(a^{-1}z)$ is

$$r_1 < |a^{-1}z| r_2$$

or

$$|a| r_1 < |z| < |a| r_2$$

Time reversal : If

$$x(n) \xleftrightarrow{z} X(z) \qquad ROC : r_1 < |z| < r_2$$

then

$$x(-n) \xleftrightarrow{z} X(z^{-1}) \qquad ROC : \frac{1}{r_2} < |z| < \frac{1}{r_1}$$

From the equation,

$$X(z) = \sum_{n=-\infty}^{\infty} x(n) z^{-n}$$

We have

$$Z\{x(-n)\} = \sum_{n=-\infty}^{\infty} x(-n) z^{-n} = \sum_{l=-\infty}^{\infty} x(l)(z^{-1})^{-l} = X(z^{-1})$$

where the change of variable $l = -n$ is made. The ROC of $X(z^{-1})$ is

$$r_1 < |z^{-1}| < r_2 \text{ or equivalently } \frac{1}{r_2} < |z| < \frac{1}{r_1}.$$

Differentiation in the z-domain : If

$$x(n) \xleftrightarrow{z} X(z)$$

then

$$n x(n) \xleftrightarrow{z} -z \frac{dX(z)}{dz}$$

By the definition of z-transform, we have

$$X(z) = \sum_{n=-\infty}^{\infty} x(n) z^{-n} \quad \text{- - - - (1)}$$

Now, by differentiating both sides of (1) we get,

$$\frac{dX(z)}{dz} = \sum x(n)(-n) z^{-n-1} = -z^{-1} \sum^{\infty} [n x(n)] z^{-n}$$

smile

Convolution: If $x_1(n) \xleftrightarrow{z} X_1(z)$

$$x_2(n) \xleftrightarrow{z} X_2(z)$$

Then,

$$x(n) = x_1(n) * x_2(n) \xleftrightarrow{z} X(z) = X_1(z)\, X_2(z)$$

The ROC of $X(z)$ is at least, the intersection of that for $X_1(z)$ and $X_2(z)$.

The convolution of $x_1(n)$ and $x_2(n)$ is defined as,

$$x(n) = \sum_{k=-\infty}^{\infty} x_1(k)\, x_2(n-k)$$

The z-transform of $x(n)$ is

$$X(z) = \sum_{n=-\infty}^{\infty} x(n)\, z^{-n} = \sum_{n=-\infty}^{\infty} \left[\sum_{k=-\infty}^{\infty} x_1(k)\, x_2(n-k) \right] z^{-n}$$

Upon interchanging the order of the summations and applying the time-shifting property in $x(n-k) \xleftrightarrow{z} z^{-k} X(z)$

We obtain,

$$X(z) = \sum_{k=-\infty}^{\infty} x_1(k) \left[\sum_{n=-\infty}^{\infty} x_2(n-k)\, z^{-n} \right]$$

$$= X_2(z) \sum_{k=-\infty}^{\infty} x_1(k)\, z^{-k}$$

$$= X_2(z)\, X_1(z).$$

Computation of the convolution of two signals using the z-transform requires the following steps:

1. Compute the z-transforms of the signals to be convolved,

smile

property can easily be generalized for an arbitrary number ...

Time Shifting: If

$$x(n) \xleftrightarrow{z} X(z)$$

then $$x(n-k) \xleftrightarrow{z} z^{-k}X(z)$$

The ROC of $z^{-k}X(z)$ is the same as that of $X(z)$ except for $z=0$ if $k>0$ and $z=\infty$ if $k<0$.

Scaling in the z-domain: If

$$x(n) \xleftrightarrow{z} X(z), \quad ROC: r_1 < |z| < r_2$$

then $$a^n x(n) \xleftrightarrow{z} X(a^{-1}z), \quad ROC: |a|r_1 < |z| < |a|r_2$$

for any constant a, real or complex.

From the equation,

$$X(z) = \sum_{n=-\infty}^{\infty} x(n) z^{-n}$$

we get,

$$Z\{a^n x(n)\} = \sum_{n=-\infty}^{\infty} a^n x(n) z^{-n} = \sum_{n=-\infty}^{\infty} x(n)(a^{-1}z)^{-n}$$

$$= X(a^{-1}z)$$

smile

3(a)

Determine the response of the following system to the input signal

$$x(n)=\begin{cases}|n|, & -3 \le n \le 3\\ 0, & \text{otherwise}\end{cases}$$

$y(n)=1/3[x(n+1)+x(n)+x(n-1)]$

SAPO exam2.2.1 Determine the response of the following system in the input signal

$$x(n)=\begin{cases}|n| & -3 \le n \le 3\\ 0 & \text{otherwise}\end{cases}$$

$$y(n)=\tfrac{1}{3}[x(n+1)+x(n)+x(n-1)]$$

$$y(0)=\tfrac{1}{3}[x(-1)+x(0)+x(1)]$$

$$=\tfrac{1}{3}[1+0+1]=\tfrac{2}{3}$$

$$y(1)=\tfrac{1}{3}[x(2)+x(1)+x(0)]=\tfrac{1}{3}[2+1+0]=1$$

$\vdots$

$$\therefore\ y(n)=\{\cdots 0, 1, \tfrac{5}{3}, 2, 1, \tfrac{2}{3}, 1, 2, \tfrac{5}{3}, 1, 0, \cdots\}$$

There are three methods that are often used for evalution of the inversion of z-transform.

(1) Direct evalution by contour integration.

smile

(b)Discuss the classification of discrete time system.

Combindly as

$$x(n) = x_e(n) + x_o(n)$$

Classification of discrete time ~~signal~~ System:

Static ~~vs~~ (memoryless) vs Dynamic (with memory) system:

A Discrete time signal is called static or memoryless if its output at any instant n depends at most on the input ~~sim~~ sample at the same time but not on past or future samples of the input.

smile

$y(n) = a x(n),$
$y(n) = n x(n) + b x^3(n)$

If the output of the system depends not only the present sample of the input but also the past sample of the input is called dynamic or with memory system. Its output at time n is completely determined by the input sample in the interval, ~~determined by the input~~

$$n-N \text{ to } n \ (N \geq 0)$$

$$[n-(n-N) = N]$$

The system is said to have memory of duration N.

Example: $y(n) = x(n) + 3x(n-1)$

$$y(n) = \sum_{k=0}^{n} x(n-k)$$

smile

Time variant and time invariant systems

A system is called time invariant if its input-output characteristics output change with time. A relaxed system $\mathcal{T}$ is time invariant or shift invariant if and only if

$$x(n) \xrightarrow{\mathcal{T}} y(n)$$

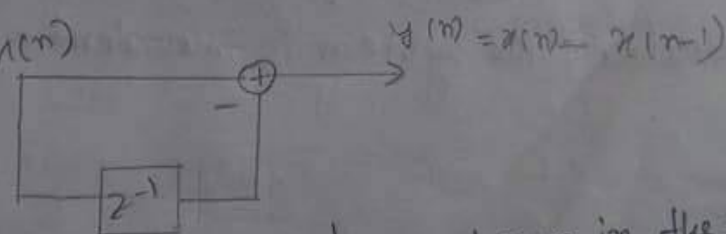
$$x(n-k) \xrightarrow{\mathcal{T}} y(n-k)$$

for every input signal $x(n)$ and every time shift k.

If the output $y(n,k) \neq y(n-k)$ at least for one value of k, is called time variant.

30-11-16

Q.



Determine if the system shown in the figure is time variant or invariant?

smile

(c)Define the following term
signal,system,Nyquist rate

Signal:

Signal is Defined as physical quantity that varies with time, space or any other independent variable or variables. Mathmatically, It could be describe as a function of one or more independent variables.

$$S_1(t) = 5t$$

$$S_2(t) = 20t^2$$

$$S(x,y) = 3x + 2xy + 10y^2$$

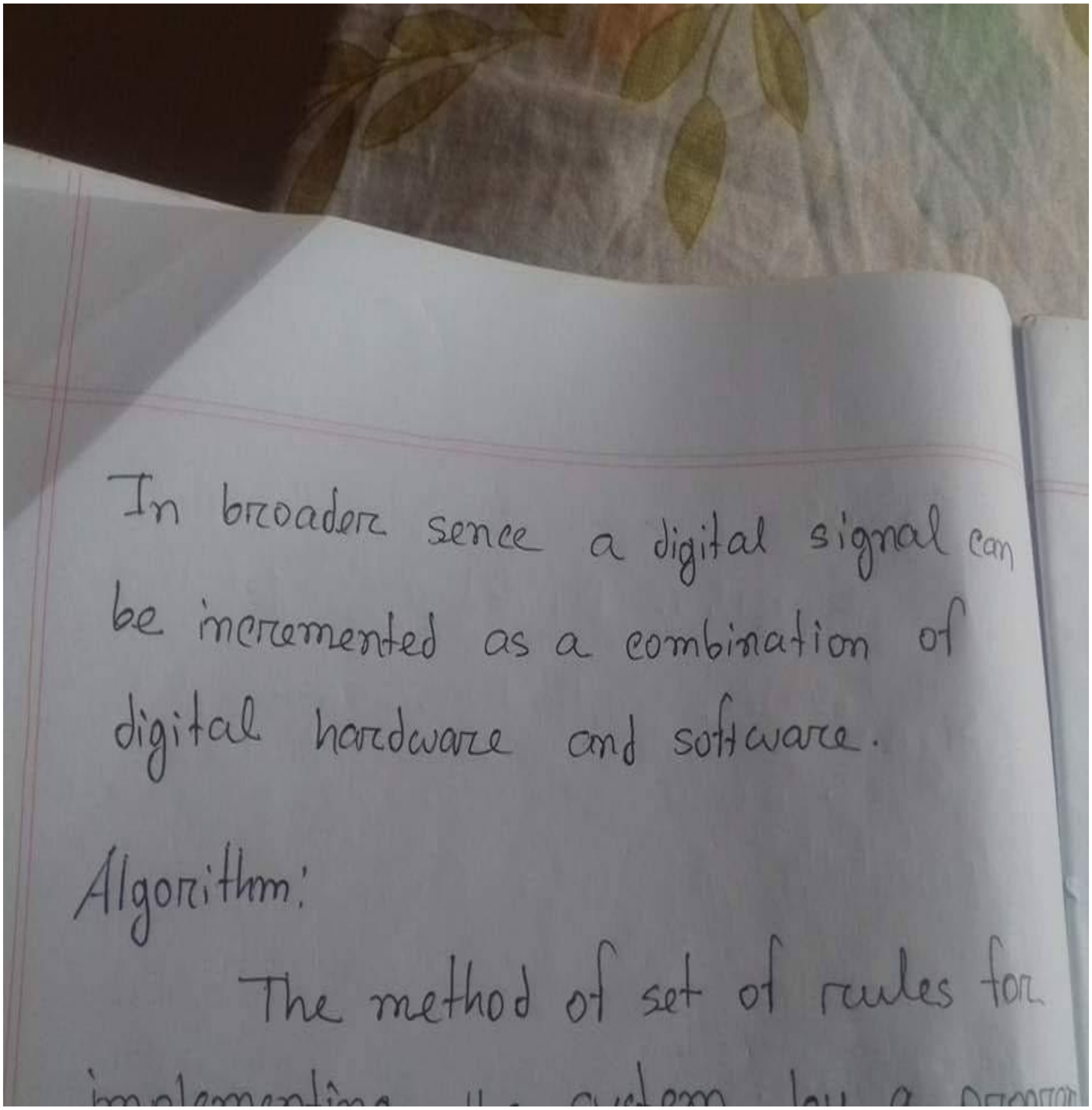
$$\sum_{i=1}^{n} \underbrace{A_i(t)}_{\text{amplitude}} \cdot \sin[\underbrace{2\pi f_i(t)}_{\text{frequency}}\, t + \underbrace{i(t)}_{\text{phase}}]$$

09-11-2016

System:

A system can be define as a physical design that performs on operation on a signal or software realization of operation on a signal.

smile


In broader sence a digital signal can
be incremented as a combination of
digital hardware and software.

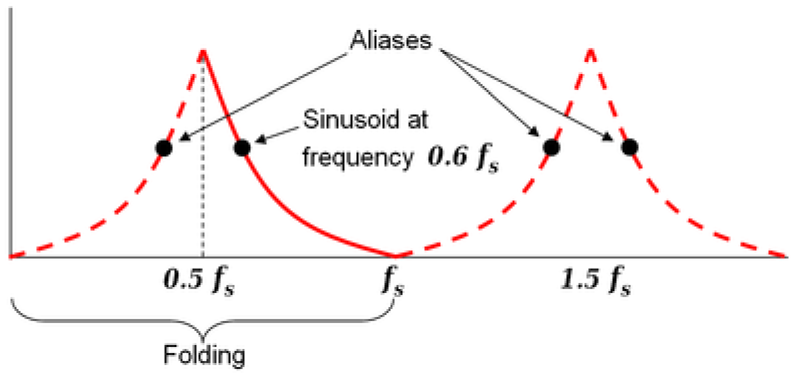
Algorithm:

The method of set of rules for

Nyquist rate:

The dashed red lines are the corresponding paths of the aliases. The **Nyquist frequency**, named after electronic engineer Harry **Nyquist**, is half of the sampling **rate** of a **discrete signal processing** system. It is sometimes known as the folding **frequency** of a sampling system.

smile



smile

(d) Consider the analog signal

$X_a(t) = 3\cos 2000\pi t + 5\sin 6000\pi t + 10\cos 12000\pi t$

(a)What is the Nyquist rate for this signal?
(b)Assume now that we sample this signal using a sampling rate $F_s = 5000$ sample/s.
(c)What is the analog signal $y_a(t)$ that we can reconstruct from the samples if we use ideal interpolation?

smile

repeated by sampling the same discrete time signal.

$F_{s/2}$ is called the folding frequency.

math

$\boxed{14.4}$

g-29

(a) $x_a(t) = 3\cos 2000\pi t + 5\sin 6000\pi t + 10\cos 12000\pi t$

$f_N = 2f_{max} = 2\times 6000$
$= 12000$ kHz

$\boxed{x_a(t) = A\cos 2\pi f t}$

Q. What is nyquest rate?

The frequency existing in the analog signal are:

$f_1 = 1000,\ f_2 = 3000,\ f_3 = 6000$

$f_n = 2f_{max}$

~~$= 2\times 12000$ kHz~~

~~$= 12000$ kHz Ans.~~

$= 2\times 6000$ Hz

$= 12000$ Hz ans

$= 12$ kHz

smile

(b) f_s = 5000 Hz sample/second what is the discrete time signal obtain after sampling?

$$x(n) = x_a(nT) = a\cos 2\pi f nT$$
$$= A\cos 2\pi \left(\frac{f_1}{f_s}\right) n$$

$f_s = 5000\,Hz$

$= 5\,kHz$

$\therefore \frac{f_s}{2} = \frac{5}{2} = 2.5\,kHz$

$$x(n) = x_a(nT) = 3\cos 2\pi \left(\tfrac{1}{5}\right)n + 5\sin 2\pi \left(\tfrac{3}{5}\right)n + 10\cos 2\pi \left(\tfrac{6}{5}\right)n$$

$$= 3\cos 2\pi \left(\tfrac{1}{5}\right)n + 5\sin 2\pi \left(1 - \tfrac{2}{5}\right)n + 10\cos 2\pi \left(1 + \tfrac{1}{5}\right)n$$

$$= 3\cos 2\pi \left(\tfrac{1}{5}\right)n + 5\sin 2\pi \left(-\tfrac{2}{5}\right) + 10\cos 2\pi \left(\tfrac{1}{5}\right)n$$

$$= 13\cos 2\pi \left(\tfrac{1}{5}\right)n - 5\sin 2\pi \left(\tfrac{2}{5}\right)n$$

smile

23-11-2016

What is the analog signal? $y_a(t)$ from the sample if we use ideal interpolation formula.

Here, the frequency components are 1KHz and 2kHz

So, the recovery signal $y_a(t)$

$$= 13 \cos 2\pi f_1 t - 5 \sin 2\pi f_2 t$$

$$= 13 \cos (2\times1000)\pi t - 5 \sin (2\times2000)\pi t$$

$$= 13 \cos 2000\pi f - 5 \sin 4000\pi t$$

7) Block diagram Representation of discrete

smile

4(a)what do you mean by DSP?Write down some application of DSP.

Sht

Digital Signal Processing (DSP-417)

Q-1/ What do you mean by DSP? List the application of DSP.

Ans: Digital Signal Processing (DSP)

Digital signal Processing is concerned with the digital representation of signals and the use of digital Processors to analyse, modify or extract information from signals. most signals in nature are analogue in form, often meaning, that they vary continuously with time and represent the variations of Physical quantities such as sound waves.

The specific reason for Processing a digital signal may be, for example - To remove interference or noise from the signal, to obtain the spectrum of the data, or to transform the signal into a more suitable form.

The classification of digital signal Processing has two types. There are-

1. Analog Signal Processing (ASP)
2. Digital Signal Processing (DSP)

smile

Application of Digital Signal Processing

- Image Processing
  - Pattern recognition
  - Robitic vision
  - image enhancement
  - Fascimile
  - animation
- Instrumentation / control
  - spectrum analysis
  - Position reduction
  - noise reduction
  - Data compression
- Speech / Audio
  - speech recognition
  - speech synthesis
  - text to speech
  - Digital audio
- Military
  - secure Communication
  - radar Processing
  - sonar Processing
  - missle Processing

smile

- Telecommunication
  - echo cancellation
  - adaptive equalization
  - ADPCM transcoders
  - video confenenceing
- Biomedical
  - Patient monitoring
  - scanners
  - ECG brain mappers
  - ECG analysis
- Consumen application
  - Digital cellunan mobile Phones
  - Digital television
  - Digital cameras
  - Intennet Phones, music & video
  - voice mail systems.

(b)Define recursive system,non-recursive system,signal,system.

smile

(c) Consider a system whose output y(n) is related to the input x(n) by

$$y(n) = \sum_{k=-\infty}^{\infty} x(k)\, x(n+k)$$

Determine whether or not the system is (a)Linear (b)Shift invariant (c)Stable (d)Causal.

smile

Question: Consider a system whose output $y(n)$ is related to the input $x(n)$ by $y(n) = \sum_{k=-\infty}^{\infty} x(k)\, x(n+k)$ — DC – 4 (e)

Determine whether or not the system is-

i) Linear
ii) Shift- invariant
iii) Stable
iv) Causal.

Answer:

Given, $y(n) = \sum_{k=-\infty}^{\infty} x(k)\, x(n+k)$

i) $y(0) = \sum_{k=-\infty}^{\infty} x^2(k)$

$y_1(0) = \sum_{k=-\infty}^{\infty} x_1^2(k)$

$y_2(0) = \sum_{k=-\infty}^{\infty} x_2^2(k)$

$\therefore y_3(0) = \tau[a_1 x_1(n) + a_2 x_2(n)]$

$= [a_1 x_1(k) + a_2 x_2(k)]^2$

$= a_1^2 x_1^2(k) + a_2^2 x_2^2(k) + 2a_1 a_2 x_1(k)\, x_2(k).$

on the other hand,

$a_1 y(0) + a_2 y_2(0) = a_1 x^2(k) + a_2 x_2^2(k)$

$\therefore$ The system is non-linear.

ii) Given, $y(n) = \sum_{k=-\infty}^{\infty} x(k)\, x(n+k)$

Output change:

$y(n-k) = \sum_{k=-\infty}^{\infty} x(k)\, x(n-k+k) = \sum_{k=-\infty}^{\infty} x(k)\, x(k).$

smile

output change :

$$y(n,k) = \sum_{k=-\infty}^{\infty} x(n-k)\,x(n+k-k)$$

$$= \sum_{k=-\infty}^{\infty} x(0)\,x(n).$$

$\therefore\ y(n,k) \neq y(n-k).$

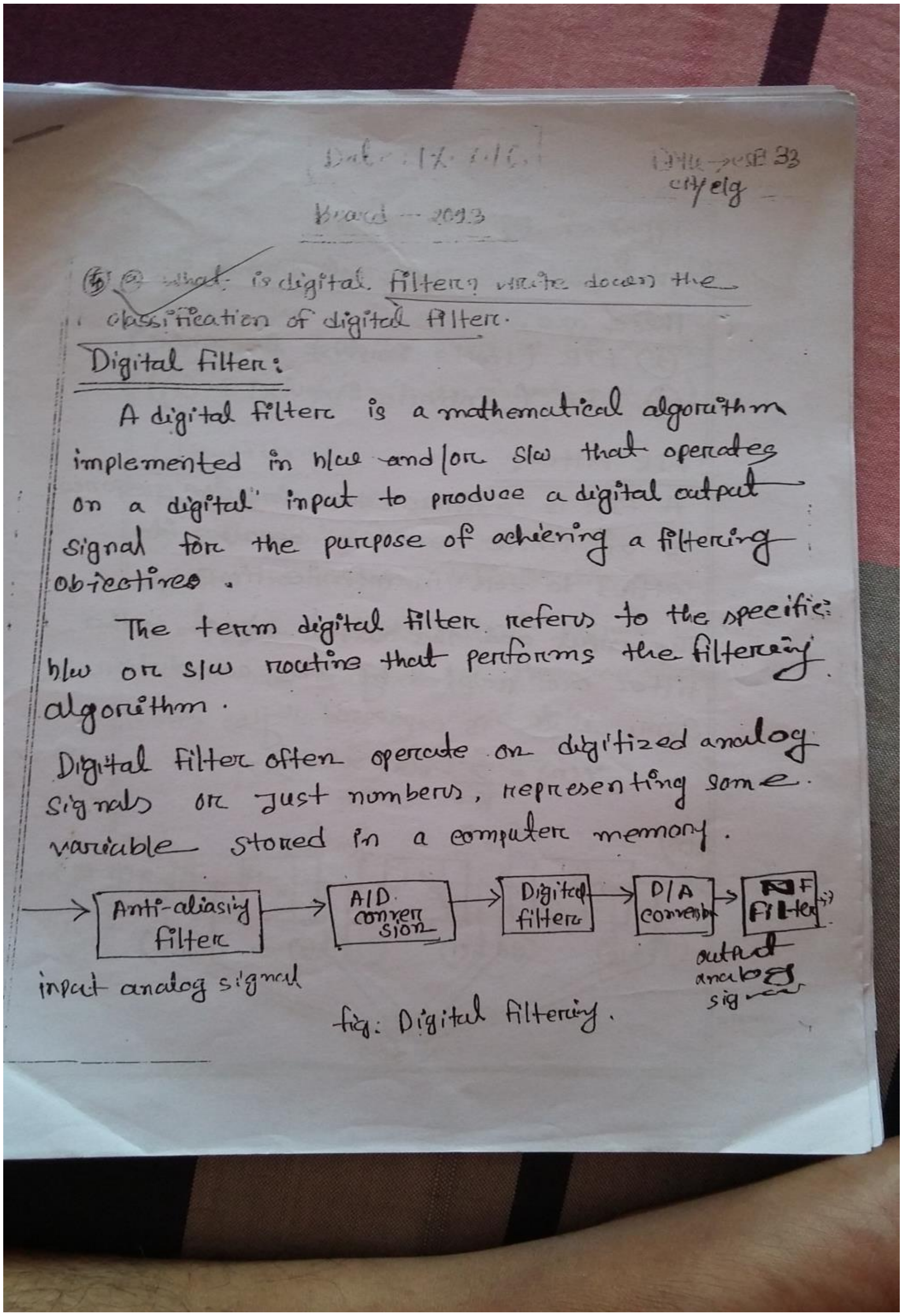
So the system is not shift.

iii) The system is not bounded, so the system is not stable or the system is unstable.

iv) Future input, not causal.

smile

5(a)Define digital fliter with simplified block diagram.



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5 (a) What is digital filter? Write down the classification of digital filter.

Digital Filter:

A digital filter is a mathematical algorithm implemented in h/w and/or s/w that operates on a digital input to produce a digital output signal for the purpose of achieving a filtering objectives.

The term digital filter refers to the specific h/w or s/w routine that performs the filtering algorithm.

Digital filter often operate on digitized analog signals or just numbers, representing some variable stored in a computer memory.

fig: Digital filtering.

(b)Advantages of direct and cascade diagram.

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(c)Advantages of normalized frequance in degining fliter.

DC-5(c)

Question: (a) What are the advantage of using normalized frequency in designing filter? [... 2011,

(b) Write the various frequence transformation in analog domain. [NUE 201...

Answer: (a) Advantages of using normalized frequency in designing filter:-

The normalized frequency of a periodic signal is its frequency expressed in units of cycles (or radians) per sample, rather than in the usual SI units of hertz (cycles per second). The cycles-per-sample frequency is computed by dividing the cycles-per-second frequency by the sampling rate (samples per second); symbolically the "per second" units cancel:

(Cycles/ second) / (Samples/second) = cycles/ sample.

The abstract reason for using normalized frequency is that, from the point of view of signal processing, a second is an arbitrary unit of time, while the sampling interval is a meaningful quantity: the frequency of a signal with respect to 1 second does not tell we about the behavior of the signal, but the frequency of a signal with respect to the sampling interval tells we the effect of sampling on the signal, via the sampling theorem. Stated alternatively, this process is called "normalization", and the sampling frequency is a normalizing constant.

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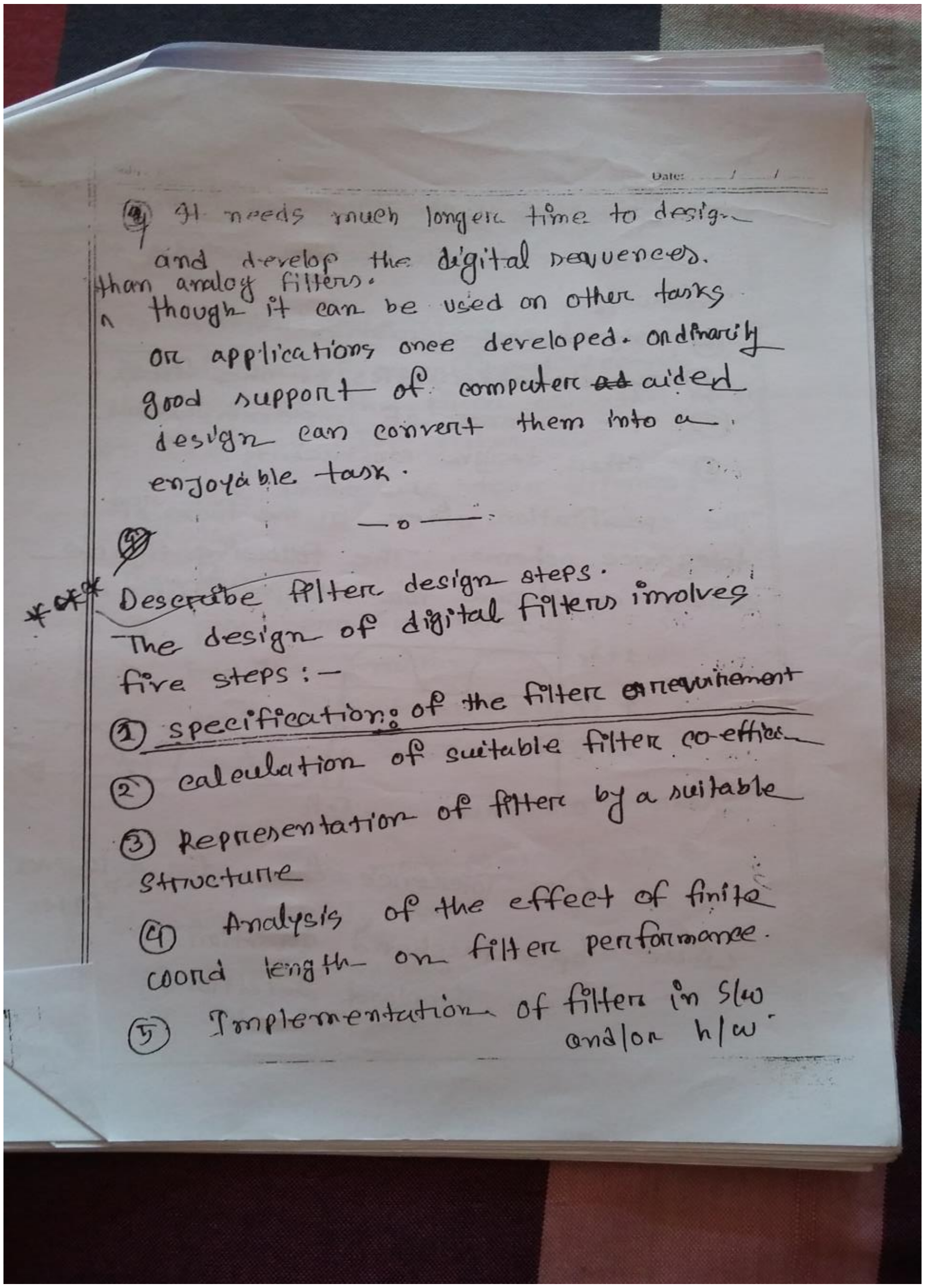
(d)Describe fliter design step.



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(4) It needs much longer time to design and develop the digital sequences. than analog filters. though it can be used on other tasks or applications once developed. ordinarily good support of computer ~~ad~~ aided design can convert them into a enjoyable task.

— o —

(5)

*** Describe filter design steps.

The design of digital filters involves five steps :—

(1) specification of the filter requirement

(2) calculation of suitable filter co-effic

(3) Representation of filter by a suitable structure

(4) Analysis of the effect of finite coord length on filter performance.

(5) Implementation of filter in S/w and/or h/w.

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Specification of the filter requirement:

Requirement specification include ~~the~~ following.

ⓐ signal characteristics

ⓑ The characteristics of the filter.

ⓒ The manner of implementation.

ⓓ Other design constraints.

The specification often in the form of tolerance schema. The following figure shows a schema for a lowpass filter.

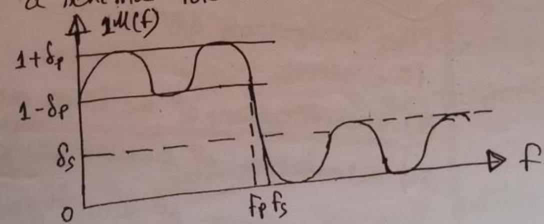

fig: Tolerence schema for a lowpass filter.

where δ_p = Passband deviation

δ_s = Stopband deviation.

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f_p = passband edge frequency.

f_s = stopband edge "

(2) Co-efficient calculation:

Calculation of co-efficient depends on the type of the filter wheather it is an IIR or FIR filter.

for IIR there are four different methods.

They are:—

(a) pole-zero placement.
(b) impulse invariant.
(c) Matched z-transform
(d) bilinear z-transform.

For FIR filter there are several methods but most commonly used three methods are.

(i) window method.
(ii) Optimal sampling.

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(iii) Frequency sampling. Realizati.

(3) Representation of a filter by suitable structures

Realization involves converting a given transfer function. Block diagrams are often used to depit filter structures and they show the computation procedure for implementing the DF.

For IIR filter three structure commonly used.

1. direct form
2. cascade "
3. parallel "

(4) Analysis of the effect of finit wordlength filter performance:

In this section the arithmatic operation indicated in differences equation are performed using finit precision arithmatic. It has some features which make analysis perfect. these are

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(1) A finite number of bits are to degrade the performance of the filter.

(2) It is ~~useable~~. unusable.

(3) The designer must analyse these effects and choose suitable wordlength.

The main source of performance degradation (wasting (ঙ্গ) error) in digital filters are as follows.

1. Input/output signal quantization.
2. co-efficient quantization.
3. Arithmetic roundoff error.
4. overflow.

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5 Implementation of filter in s/w and h/w:

After quantizing the co-efficients and filter variables to the selected wordlengths is acceptable, the differences equation must be implemented as a software routine or in hardware.

Implementation has some characteristics There are:-

1. calculate co-efficient
2. Suitable structure defines
3. verify filter degration
4. Select wordlength.
5. Analyze co-efficient.
6. Different equation analyze.

In Implementation time the designer will define which system are save where.

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(e)Describe the advantages and disadvantages of digital fliter over analog

Question: Describe (e) the advantage and disadvantage of digital filter over analog filter? DC

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Answer: Advantage of digital filter over analog filter:

- Digital filters can have characteristics which are not possible with analog filters
- The performance of digital filters does not vary with environment changes.
- The frequency response of a digital filter can be automatically adjusted if it is implement using a programmable processor, which is why they are widely used in adaptive filters.
- Several input signals or channels can be filtered by one digital filter without the need to replicate the hardware.
- Both filtered and unfiltered data can be saved for further use.
- Extremly stable with respect to both time and temperature.
- The performance of a digital filter is repeated from unit to unit.
- Digital filters can be used at very low frequencies.
- Digital filters are small in size, consume low power and keep the cost down.
- The precision of digital filter is limited only by the word length used.

Disadvantages of digital filter over analog filter:

Lower Bandwidth: The maximum bandwidth of signals that digital filters can handle, in real time, is much lower than for analog filters. In real-time situations, the analog-digital-analog conversion processes introduce a speed constraint on the digital filter performance.

Speed limitation: The speed of operation of a digital filter depends on the speed of the digital processor used and on the number of arithmetic operations that must be performed for the filtering algorithm, which increases as the filter response is made tighter.

fliter.

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Finite wordlength effects: Digital filters are subject to ADC noise resulting from quantizing a continuous signal and to roundoff noise incurred during computation. With higher order recursive filters, the accumulation of round-off noise could lead to instability.

Long design and development times: The design and development times for digital filters, especially hardware development, can be much longer than for analog filters.

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(f)Classify digital fliter.Describe the rationales of choosing FIR and IIR fliter.

De-5(f) 52

Question: Describe the rational of choosing between FIR and IIR filters.

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Answer: The choice between FIR and IIR filters depends largely on the relative advantages of the two filter types:

- FIR filters can have an exactly linear phase response. The implication of this is that no phase distortion is introduced into the signal by the filter.
- The phase response of IIR filters are non-linear, especially at the band edge.
- FIR filters realized nonrecursively, that is by direct evaluation of equation $y(n) = \sum_{k=0}^{N-1} h(k)x(n-k)$ are always stable. The stability of IIR filters cannot always be guranteed.
- The effects of using a limited number of bits to implement filters such as roundoff noise and coefficient quantization errors are much less severe in FIR than in IIR.
- FIR requires more coefficient for sharp cutoff filters than IIR.
- Analog filters can be readily transformed into equivalent IIR digital filters meeting similar specification. This is not possible with FIR filters as they have no analog counterpart.
- In general, FIR is algebraically more difficult to synthesize, if CAD support is not available.

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6(a)Explain --
i)Z transfrom and inverse transform.
ii)FIR and IIR fliter.

iii)Adaptive fliter as noise canceller.

iv)DFT nad IDFT

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