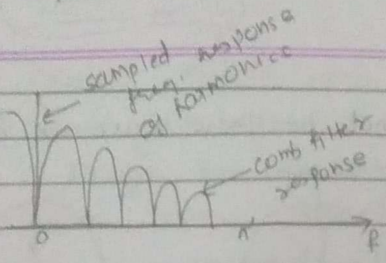
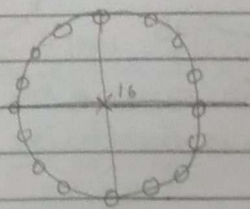
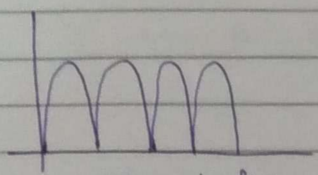


← To remove these harmonics we use comb. filter as shown.

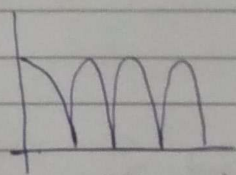
z-Plane (for comb filter)



In comb filter, the zeros are placed in harmonical manner to remove the harmonics. → (ω, 2ω, 4ω, 6ω, ...)



HPF comb filter



LPF Comb. filter

To get comb filter equation, replace z by z^L in filter equation.

where L is no. zeros required in the filter.

14 - Sep

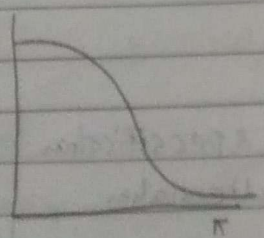
IIR filter Design:

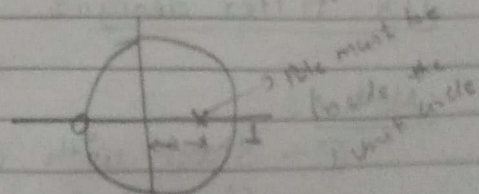
① Simple LPF IIR:

↓

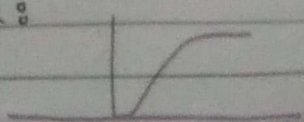
(first order → den. Power → Poles)

$$\alpha = \frac{1-\sin\omega c}{\cos\omega c}$$





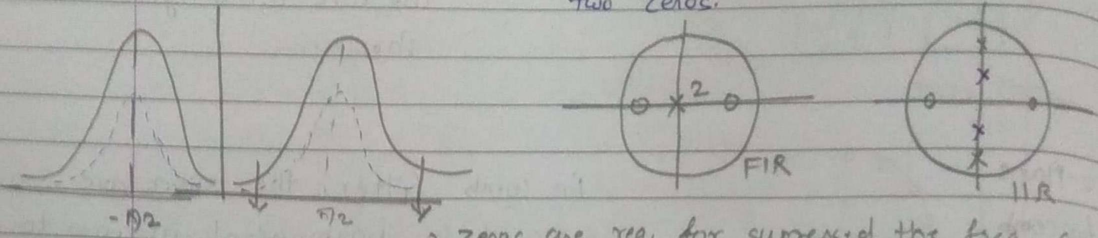
② Simple HPF IIR:



Date 26-Sep

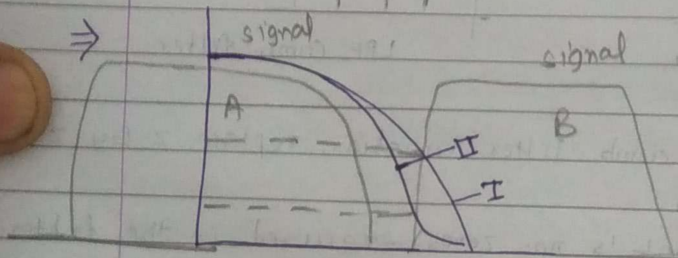
③ Simple BPF IIR:

Single order BPF ~~may~~ not be there becuz we required two zeros.



2 zeros are req. for supressed the freq. so there should be two or more poles are in sis to becomes the sis causal.

– α, β



II ~~filter~~ LPF is (steeper response)

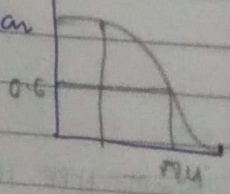
we have to design the filter type of II. for attenuate or remove the signal B

– Very steeper response are required in sophasticated application like in mobile. So we have to design that kind of filter that's IIR filters.

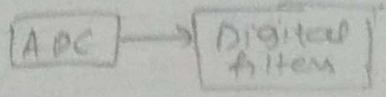
IIR filter design:

Method:

① – Start with digital specsification (for each freq. what attenuation required)



Z → digital domain



Date
Page

(2) Convert digital specs into analog specs using some kind of transformation e.g. BLT

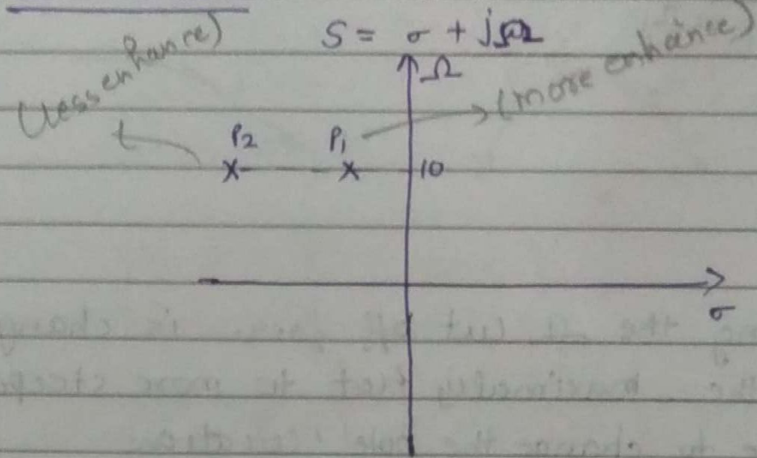
(3) Design Analog filter i.e. $H(s)$

(4) Use inverse transformation to convert $H(s)$ into $H(z)$.

Analog filter Design:

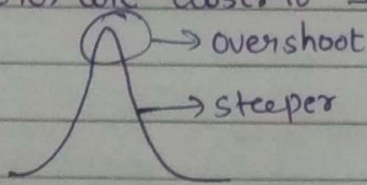
s-domain:

$s = \sigma + j\Omega$

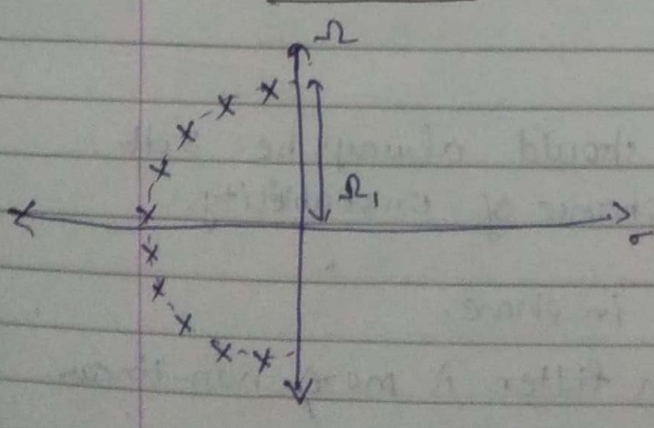


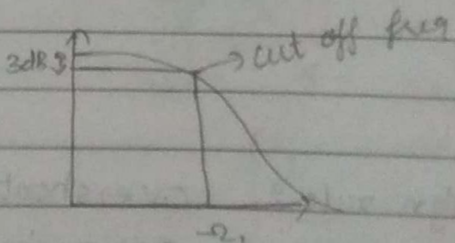
- poles at LHS ⇒ S/S is stable

- poles are closer to Ω axis, more dominating. more steeper sharp overshoot



1) Butterworth:

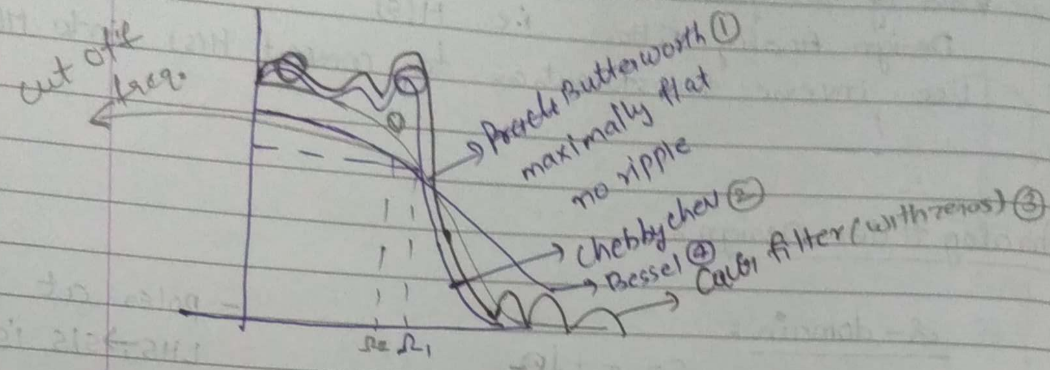




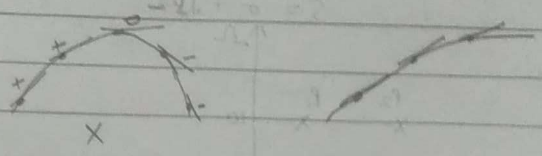
- butterworth is all pole and poles are on circle
- conjugate pair of poles are present because of real S/S.

- overshoot mag. is less than 3db, for stable s/s
-

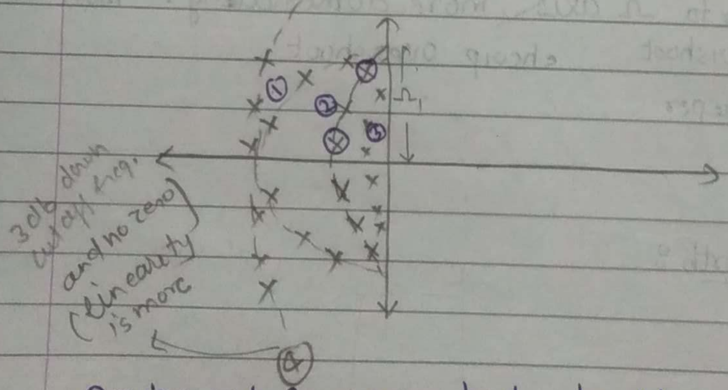
$$H(s) = \frac{1}{\text{eq}^n \text{ of } \odot \text{ circle}}$$



maximally flat



- By changing the Ω, cut off freq. is changes
  to make the maximally flat to more steeper
  we have to change the pole location
  circle $\longrightarrow$ ellipse (Pole nearer to Ω axis)



Design rule: overshoot should always be 3 db
otherwise chance of unstability.

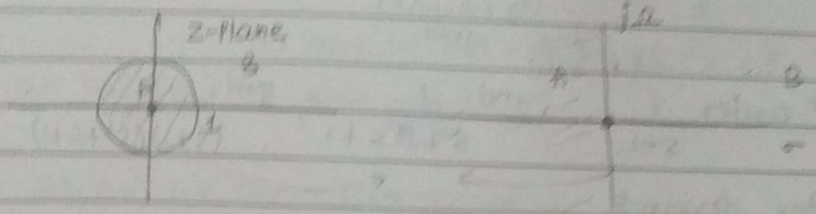
- ripple cause non-linearity in phase.
- The phase responce of caur filter is more non-linear as compared to cheby-chav
- To get more linear response $\Rightarrow$ poles away from circle
  IIR cannot be LINEAR $\boxed{\text{Bessel}}$

steep comes with ripple
linear

Date: 27-Sep

Digital IIR filter:

$z \longrightarrow s$



Eg: $s = z^2 + z - 2$

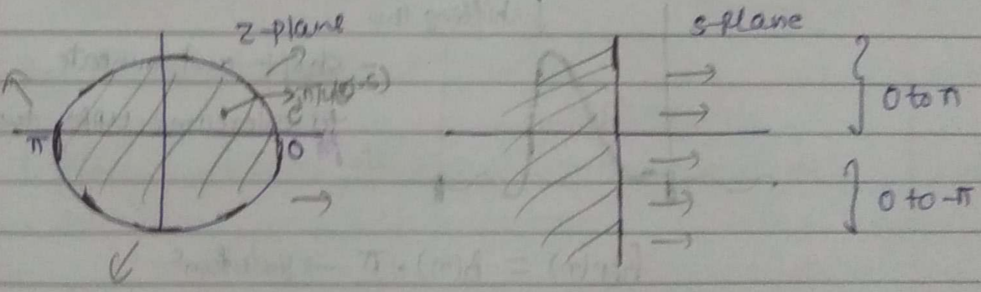
its not the valid transformation
bcuz if we put $z = 0$; $s = -2$, which is not valid.

$$\boxed{s = \frac{2}{T_s}\left(\frac{z-1}{z+1}\right)}$$

$T_s \rightarrow$ constant

$$\boxed{s = \frac{z-1}{z+1}} \quad \text{(A)} \quad \begin{pmatrix} z \to e^{j\omega} \to \text{digital freq} \\ s \to j\Omega \to \text{analog freq} \end{pmatrix}$$

its mapped



in eqn (A)

$z = e^{j\omega}$

& $s = j\Omega$

$H(\Omega) = H(s)\vert_{s=j\omega,\ \sigma=0}$
$H(s)$
$H(\Omega) \longrightarrow h(t)$ FT $\to$ S-do
$H(e^{j\omega}) = H(z)\vert_{\vert z\vert=1}$
DTFT $\to$ Z

then

$$\boxed{\Omega = \frac{2}{T_s}\tan\frac{\omega}{2}} \qquad \omega = \frac{2\pi F}{F_s}$$

discrete freq. is transform into analog freq.
that's Known as Bilinear transformation.

for Butterworth

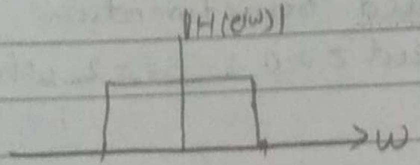
- $N \geq \frac{1}{2} \frac{\log\left(\frac{1}{\delta_2^2}-1\right) - \log\left(\frac{1}{\delta_1^2}-1\right)}{\log(\Omega_s/\Omega_p)}$

$\Omega_c = \Omega_p / \left(\frac{1}{\delta_1^2}-1\right)^{1/2N}$

1st order $\frac{1}{s+1}$; 2nd $\frac{1}{s^2+\sqrt{2}s+1}$; 3rd $\frac{1}{(s+1)(s^2+s+1)}$

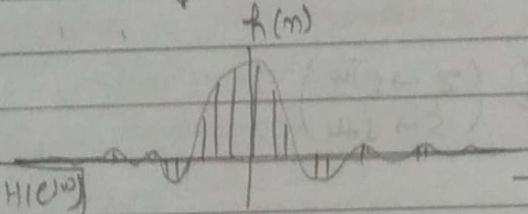
FIR filter design :

- window method or F.T. method.



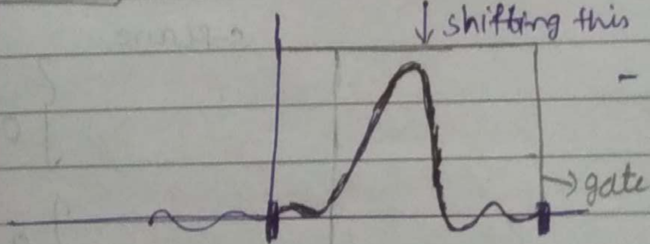
→ ideal low pass filter

$\updownarrow F^{-1}$



problem:
- h(n) becomes infinite
- its not causal
  $h(n) = 0 \; ; \; n < 0$

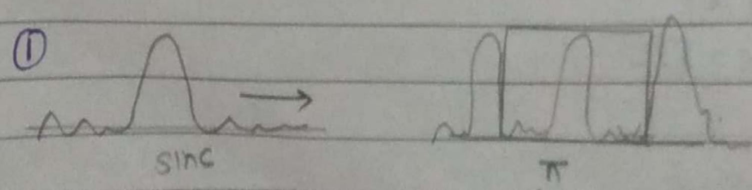
$\boxed{h(n-n_0) \leftrightarrow e^{-jn_0\omega} H(e^{j\omega})}$

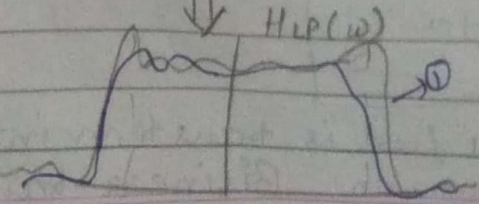


- shift and truncate to overcome upper two problem

$h_{LP}(n) = h(n) \cdot \Pi \rightarrow$ gate func^c

$H_{LP}(e^{j\omega}) = \Pi * \text{sinc}$





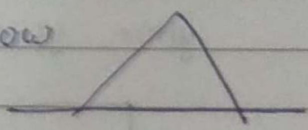
Gibbs effect

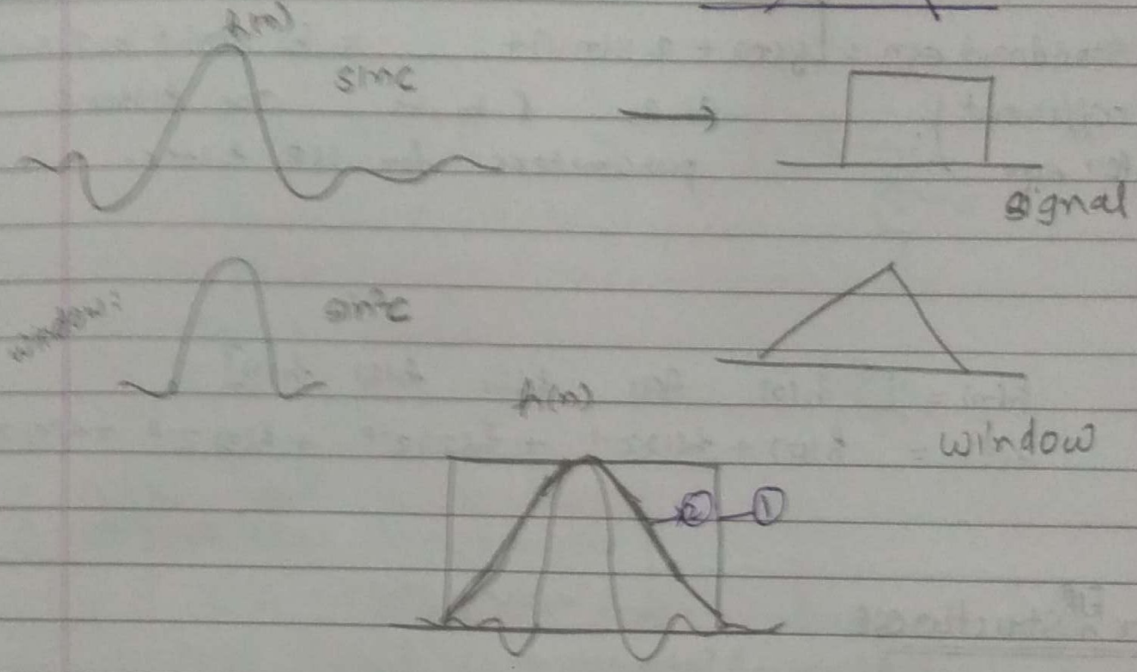
we have to reduce the ripples

↓

its bcuz we are using window, in which we are restricted the sinc wave

So now we are using window





by 2 ripples will reduced.

- Different shapes of windows are used for reduced the ripples.

Handout: FIR linear phase Digital filter Design using FT and Window method.

Date 18-Oct-

for IIR:

$$H(z) = \frac{Y(z)}{X(z)} = \frac{(\ldots) \rightarrow b}{(\ldots) \rightarrow a}$$ (Designing)

$$Y(z)[\ldots] = X(z)[\ldots]$$

↓ in time domain

$$y(n) + \_\, y(n-1) + \_\, y(n-2) = \_\_\_$$

Standard eqn (constant coefficient linear differ eqn): $\boxed{y(n) + a_1 y(n-1) + \ldots = b_1 x(n) + b_2 x(n-1) + \ldots}$

$a_1, a_2 \ldots$ & $b_1, b_2 \ldots$ are filter's parameters for IIR filter.

for FIR:

$$h(n) = [\, h(0) \quad h(1) \quad h(2) \quad h(3) \quad h(4) \,]$$

$$H(z) = h(0) + h(1)z^{-1} + h(2)z^{-2} + h(3)z^{-3} + h(4)z^{-4}$$

↓ what we do next, that's called as implementation

filter FIR Structures:

↳ (Pictorial Representation of filters)

for FIR: $H(z) = \frac{Y(z)}{X(z)}$ (Equalizer, moving avg. are using FIR filters)

↓ in time domain

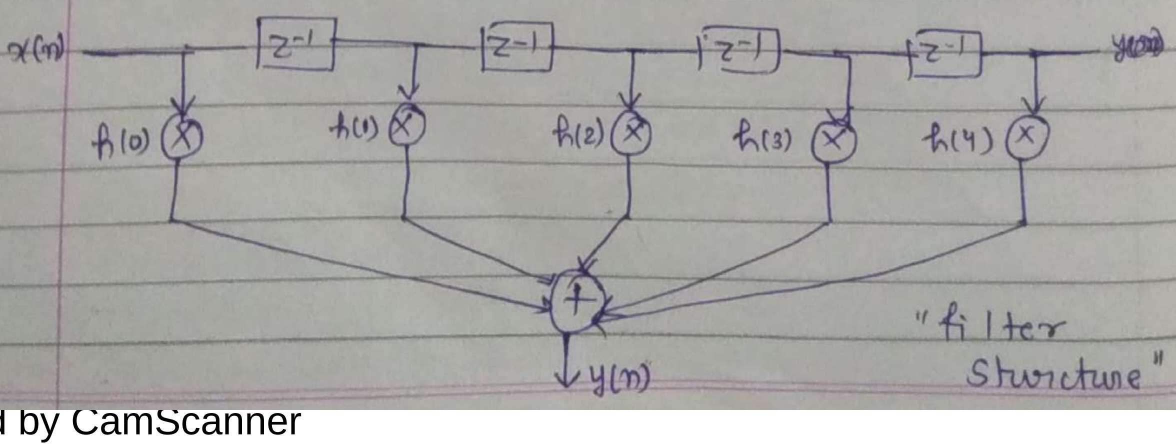
$$y(n) = h(0)x(n) + h(1)x(n-1) + h(2)x(n-2) + h(3)x(n-3) + h(4)x(n-4) \quad \text{— (A)}$$

O/P depends on only previous I/Ps.

Ex:- Digital x-ray, radar, audio signal

↓ $f = 8K$

$t = 1/8K$



"filter Structure"

gn using FT and

Date 18-Octo

(Designing)

]

ain

$= b_1 x(n) + b_2 x(n-1) + \dots$

are filter's

IIR filter.

$h(4)]$

$+ h(3) z^{-3} + h(4) z^{-4}$

hat's called as

alizer, moving avg.

using FIR filters)

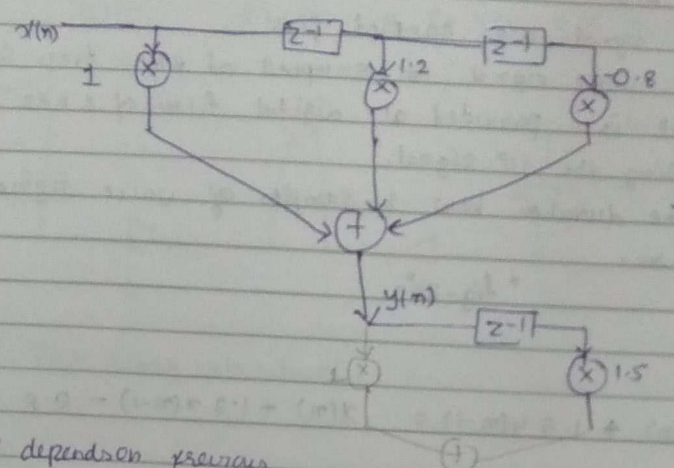
$(n-2) + h(3)\, x(n-3)$

filter

Structure"

Canonic: A digital filter structure is said to be canonic if the no. of delay = order of the transfer fun.

for IIR: $y(n) + 1.5\, y(n-1) = x(n) + 1.2\, x(n-1) - 0.8\, x(n-2)$



"filter structure" this taken to lab for implementation)

- O/P depends on previous O/P & i/ps.

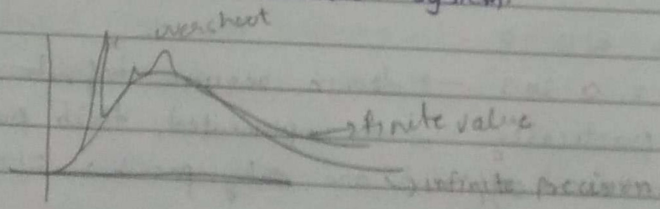
fig-2

finite word length effect:

- if we are using the finite values of a & b then it may occur overshoot and unstablize the s/s

$a = 0.3164$ → we are using $a = 0.3$

then it will may shift the pole position, then it may occur that pole at the unit circle, which makes the unstable system.



Quantization:

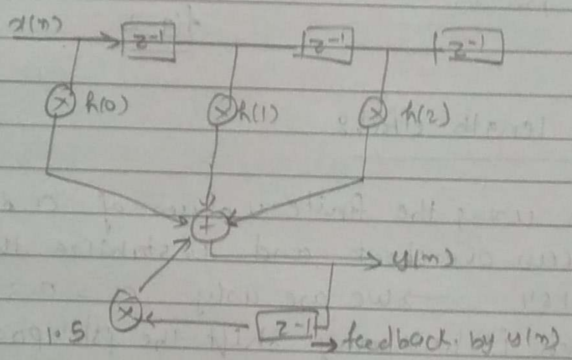
for digital filter design, when we quantization our signal then we do some approximation that's we can say that we finite the values.

Date 19-Octo
Page

- ~~approximation of window (~~
in eq^n Ⓐ - $x(n)$ here can be audio signal, radar signals, X-ray etc.
- analog signals are sampled first.
  Ex: voice signal is sampled at 8kHz freq. So, a clock signal is provided at digital filter of 8 kHz for sampling the i/P signal.
- The time duration b/w 2 samples of voice signal is 1/8K sec.

"fig-2"

$$y(n) + 1.5\, y(n-1) = x(n) + 1.2\, x(n-1) - 0.8\, x(n-2)$$



problems:

① $h(0) = h(4) = 0$

② $h(2) = 0.3183$ → design requires infinite precision but practically when implemented, finite precision parameters / coefficients are only possible ⇒ error occurs.

"finite word length concept" (on Previous Page)

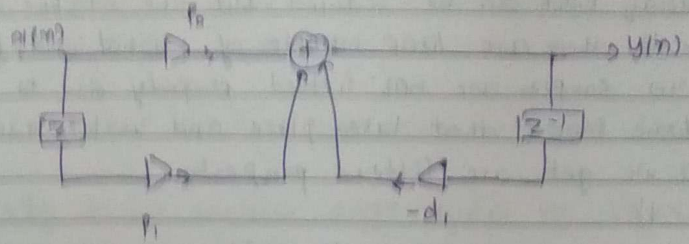
implementation techniques:

① hardware ③ FPGA

② Software → DSP processors & program it

Equivalent structure :

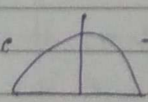
$$y(n) = -d_1\, y(n-1) + p_0\, x(n) + p_1\, x(n-1)$$



" Tap delay structure"

19-Oct

- In digital design, approximation occurs due to quantization taking place on signals.
- A finite length of signal is considered practically, i.e. approximation takes place → done by rectangular window

But practically we take a window which is like  → again approximation takes place.

$$y(n) = d_1\, y(n-1) + p_0\, x(n) + p_1\, x(n-1)$$

general structure → similar to done previously

(Transpose operation) — another structure :

- ① Reverse all path
- ② Replace pick-offs node by adder and vice versa (node ↔ adder)
- ③ interchange i/p and o/p nodes.

- LPC contains both FIR & IIR filter.

Basic FIR Structure :

- used in moving average filter.
- used in equalizer
- Known as "Tap delay structure" ↓ multiplication of coefficient
- Also known as "direct form"
- delay of this filter = $\frac{1}{f_s \text{ (sampling freq)}}$

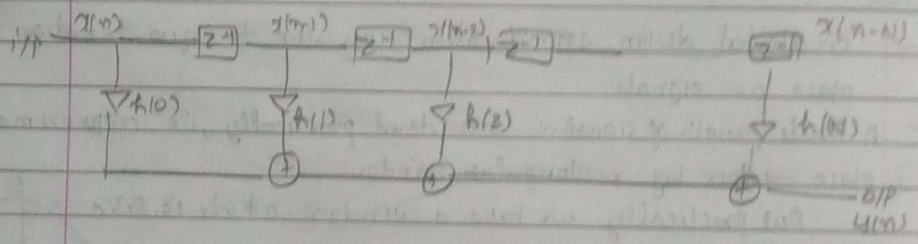
Output of filter is at the same instant when I/P is provided. for Initial few samples, data is not filtered properly → This is a limitation.

⇓

So delay block can be reduced, but the steepness ↓ses.
Ex:- if there are 1000 samples of signal and from them 1st 70 samples are not filtered properly due to the hardware process that takes place and rest 930 samples that we get are filtered properly

⇓

To reduce the no. of samples which are not properly, the processor design should contain less multiplication & addition blocks.



20-Oct.

Slide topics

linear phase FIR filter structures
Cascade FIR structures
IIR Filter structures

- We can change the order of multiplier and delay block, only becuz the s/s is linear.
  - similar thing was done in
    Σ-Δ modulation & Δ-Σ modulation
    spread spectrum (spreading & modulation or vice-versa
    ↳ done for analysis purpose.

- advantage of factorizing H(z) & cascading the structures

The coefficients of H(z) changes ⇒ finite precision taken ~~we~~ get it on the coeff. may ↓ses ⇒ o/p ~~that~~ ~~we~~ get is the required one.

— Direct form → normal factorization of H(z) takes
— parallel form   place.
↓
Partial fraction of H(z) is taken here.

(?) why do we require diff^r types of structure.
— look at 2 slides —
① Stable o/p changes to instability due to finite precision
② perfect low pass filter changes to filter with ripples.

This happens because the poles position varies due to finite precision.
so diff^r str^e provide with diff^r o/ps with small variations ⇒ the one providing the best o/p is considered

Date 20-Octo

Multirate Digital Signal Processing

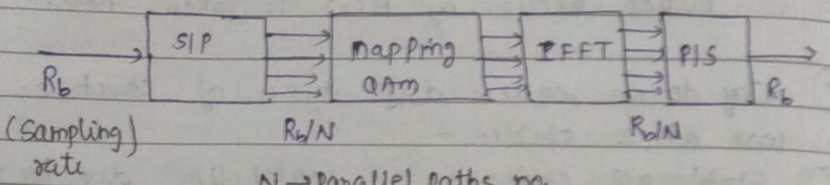
Basic sampling rate alternation devices:

- Up & down sampler

multirate : processing at different sampling rate

- A s/s which having diff sampling rate for diff s/s.

for example in OFDM



N → parallel paths no.

R_b → data rate

- filter structure → have digital components & they work on a single clock.
- clocks are req. bcuz i/ps are changing.
- when digital components work on diff clock frequencies → called multirate
- used in OFDM

- conversion of CD files (audio) to mp3 format
- Basestation

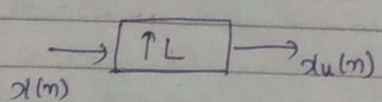
25-Octo

sampling rate
discrete freq
spectrum

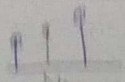
UP-sampler → increase the sampling rate
Down-sampler → decrease the sampling rate

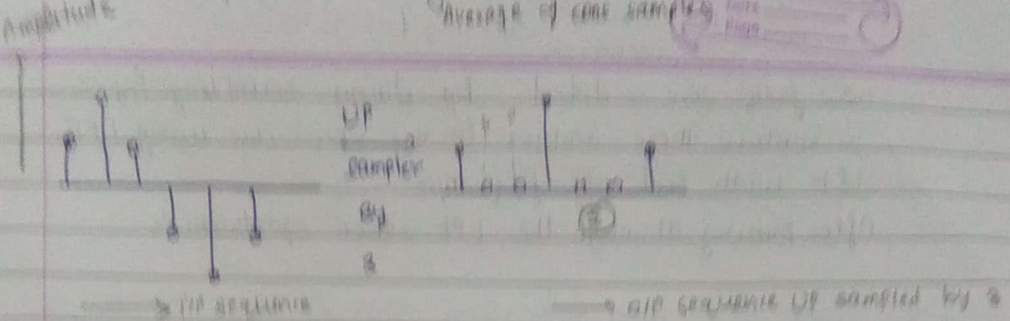
UP sampler :

$y(n) = x(n/L)$



(L-1) zero is added between two consecutive samples.

interpolation:  Average of cone samples



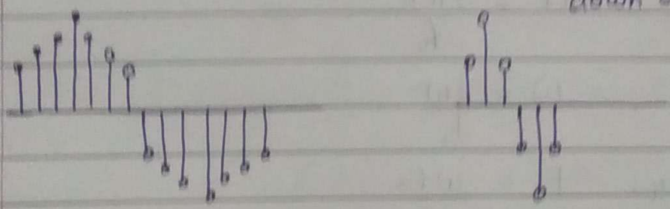
- spectrum will string after the up sampler (freq is decreased) - discrete freq. is lsen.
- By increasing the sampling rate means samples in a cycle is increases.
- passing through ② this to LPF. and (averaging is done)

$$x_u(n) = \begin{cases} x(n/2) & n = 0, \pm 2, \pm 4, \ldots \\ 0 \end{cases}$$

→ up sampling factor is 2

Down sampler:

down sampled by 3



$x(nM) = y(n)$

- spectrum will spreaded (becuz freq is lsen)(expand)
- By decreasing the sampling rate means samples in a cycle is decreases.
- discrete freq. is lsen.

Digital Analog

$f = F/F_s$

will decreases, will increa

UP sampler: (we dont changing the analog freq.

→ $X(e^{j\omega})$ → i/p

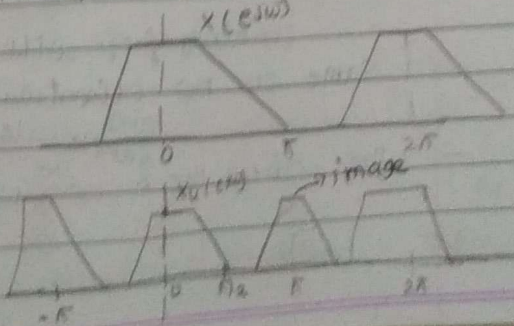
$X_u(z) = X(z^2)$

$X(e^{j\omega 2})$ → o/p up sampled by 2

$\omega = 0$; $X(e^{j0})$

$\omega = \pi/2$; $X(e^{j\pi})$

$\omega = \pi$; $X(e^{j2\pi})$

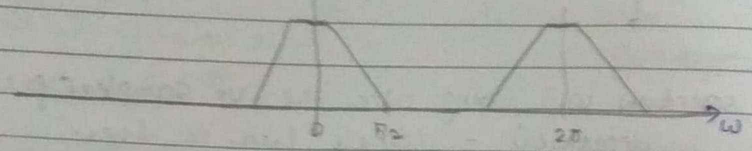


$f = F / F_s$ — discrete freq. is change.

F ↓ will remain same; F_s ↓ sampling freq.

- new freq. are added by doing sampling by 2.
- To remove these images in the spec., we using LPF.
- LPF with cut off freq. $\pi/2$
- After passing through the LPF, then spectrum is

(Discrete spectrum)



because discrete spectrum is repeatative.

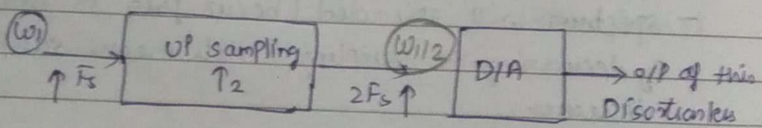
- information loss is not occur as at $\omega = \pi/2$ to π → ① data & $\pi/2$ to π → ② no data

Digital to Analog convertor

$$2\pi f = \frac{2\pi F}{F_s}$$

$$\omega = \frac{2\pi F}{F_s}$$

analog freq: $$F = \frac{\omega F_s}{2\pi}$$



$$F_1 = \frac{\omega_1 F_s}{2\pi}$$

$$F_1' = \frac{\omega_1 \times 2F_s}{2(2\pi)} = F_1$$

analog freq. will same & spectum will remain same so no distortion will occurs.

- LPF gives the avg sample value to zero valued samples.

- complete graph is string.

change.

y 2.

LPF.

$\pi \rightarrow$ ① do

P of this

rtionless

will remain

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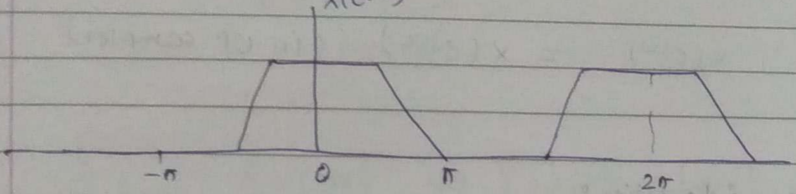
Down Sampler: ($F = f \cdot F_s$)

will increase

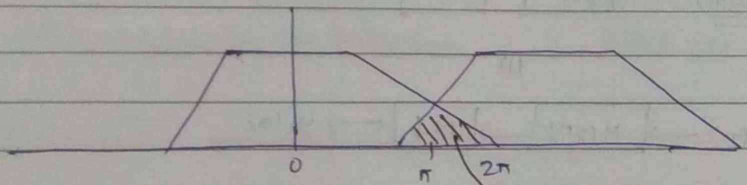
will decrease

nyquist criteria $f_s = 2 f_m$

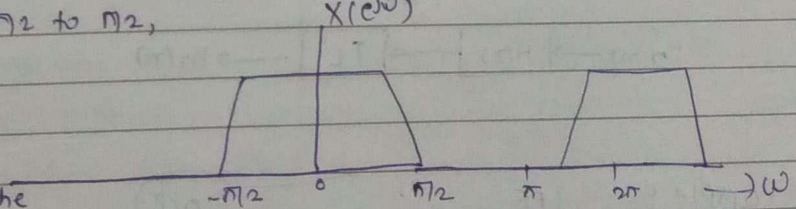
- if sampling rate is 8 KHz & $f_m = 4$ KHz, then we cant do down sampled this signal because distortion may occur.
- if over sampled $\rightarrow$ then and then only we can do down sampling.



down sampler factor by 2,

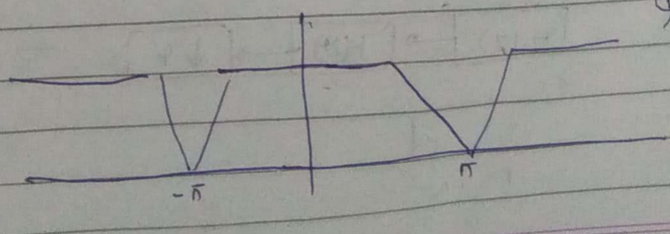


- with the y-axis graphs are expanded.
- Condition for, aliasing is not there, if down sampler factor is 2, so signal should be in between $-\pi/2$ to $\pi/2$,



- if we down sampler by M, then for getting the without any distortion signal, signal must be in the range of π/M.

$\Downarrow$ Down sampled



Fmax.

discrete

$f = \frac{F}{F_s}$

$\pi/4 = \frac{F_{max}}{2F_s}$

If we want to change the sampling rate by L/M then it means that L → up sampling factor & M → down sampling factor

$x(n) \to \boxed{\uparrow L} \to \boxed{h(z)} \xrightarrow{w(n)} \boxed{H(z)} \to \boxed{\downarrow M} \to y(n)$

* By using LPF cut off freq at π/2 (factor 2)

↓

Down sampler

* Up sampler → LPF

* LPF → down sampler

– $X(e^{j\omega}) = X(e^{jL\omega})$ (in up sampler)

Noble identities:

Cascade eq #1 $x(n) \to \boxed{\downarrow M} \to \boxed{H(z)} \to y_1(n)$ → lower freq

|||

$x(n) \to \boxed{H(z^M)} \to \boxed{M\downarrow} \to y_1(n)$ → Higher freq.

Cascade eq #2

$x(n) \to \boxed{\uparrow L} \to \boxed{H(z^L)} \to y_2(n)$ → Higher

|||

$x(n) \to \boxed{H(z)} \to \boxed{\uparrow L} \to y_2(n)$ → lower

For example :

① LPF $\boxed{H(z)} \to \boxed{\downarrow 4} \to$ DSP $\boxed{H(z)}$ → we req two clocks Fs & Fs/4

Fs   Fs/4

$\underbrace{\boxed{H_1(z)} \to \boxed{H(z^4)}}_{Fs} \to \boxed{\downarrow 4}$ → in this we req. only single clock

- High freq. designed is very difficult

(2) $\boxed{H(z)} \rightarrow \boxed{\uparrow 4} \rightarrow \boxed{H_1(z)}$ LPF

$x(n) \rightarrow \boxed{H(z^4)} \rightarrow \boxed{\uparrow L}$

$\boxed{\uparrow 4} \rightarrow \boxed{H(z^4)} \rightarrow \boxed{H_1(z)}$

Ex:- $H(z) = 1 + z^{-1} + z^{-2} + 1.5z^{-5} + 2z^{-6}$ $\} \rightarrow$

$= H(z^0) + H(z^3) + \ldots$

$\Rightarrow$ $\boxed{H(z^m)} \rightarrow \boxed{\downarrow M}$ $\Rightarrow$ $\boxed{\downarrow M} \rightarrow \boxed{H(z)}$

$\boxed{\uparrow L} \rightarrow \boxed{H(z^L)}$ $\Rightarrow$ $\boxed{H(z)} \rightarrow \boxed{\uparrow L}$

Up sampler — filtering — upsampler

$\Downarrow$ — filter operates on high freq. | $\Downarrow$ filter operates on lower freq.

~~polyphase decomposition~~ :

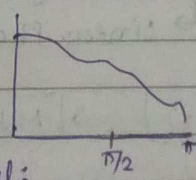
$\Rightarrow$ (Digital BANK) speech signal (Analog $\rightarrow$ Digital) $F = 4K$; $f_s = 8K$

$\omega = \frac{2\pi 4}{8} = \pi$

$\downarrow$

- Bit rate & sampling rate decrease for compress the Actual signal.

Quantization (bits provided)



8 bits/sample

- for compress the signal:

bit rate $= 64$ Kbps

- down sampler factor $= 2$, but for that after the $\uparrow \pi/2$ signal will not there. Hence we can't do down sampling here

for $\downarrow$ bit rate — for compress the signal if we $\downarrow$ ses the bits/sample then quantization noise will $\uparrow$se and if we can't $\downarrow$se the f_s Because of Down sampler concept.

- single rate → IIR filter
- multirate → FIR filter

Date 15-Nov

Polyphase Decomposition :

Basic structure :

L/M → Sampling rate change by the
L → UP sampler factor
M → Down "

$x(n) \rightarrow \boxed{\uparrow L} \rightarrow \underbrace{\boxed{H_u(z)} \xrightarrow{w(n)} \boxed{H_d(z)}} \rightarrow \boxed{\downarrow M} \rightarrow y(n)$

⇓

$\boxed{H_v(z)}$

↳ we can combine both the filter but cut off freq. is decided by the Higher cut off freq. of the $H_u(z)$ & $H_d(z)$.

- in this we can also do the down sampler first, but we can't combine the both filters.

Computation Requirements: computational efficiency is higher in IIR because for the same application order of IIR < order of FIR, so power is less in IIR, in single rate

IIR → stability
↳ Linear Phase

$x(n) \rightarrow \boxed{H_d(z)} \rightarrow \boxed{M\downarrow} \rightarrow y(n)$

(Refer slides) FIR structure

- FIR is used for multirate because computation efficiency is higher in that. As we can seen in the FIR structure, in FIR O/P depends on only previous i/p so we can disable the multiplier and adder for particular samples which we don't requires

$\boxed{\text{FIR}} \rightarrow \boxed{\downarrow M}$

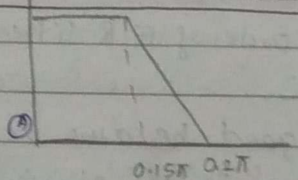
Structure

- in FIR ~~strcture~~ structure (N+1) Multiplier is required.

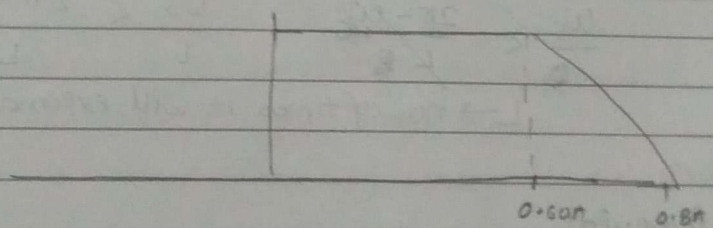
FIR	IIR
- length of N	- ~~order~~ K
$R_{M \cdot FIR} = N \times F_T$	$R_{M\,IIR} = (2K+1) F_T$
- down sampler by M.	- down sampler by M
$= N \frac{F_T}{M}$	$= K \times F_T + (K+1) \frac{F_T}{M}$

~~- How we used noble identities in system~~

Interpolated FIR Filter 8



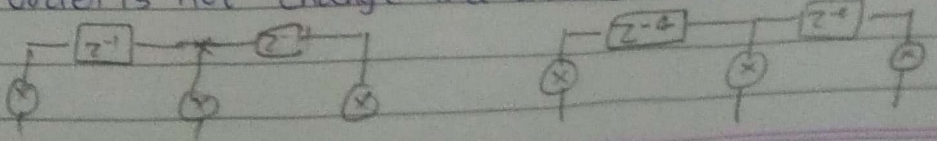
#1 Design a filter which cut off freq. is 4 times of (B)

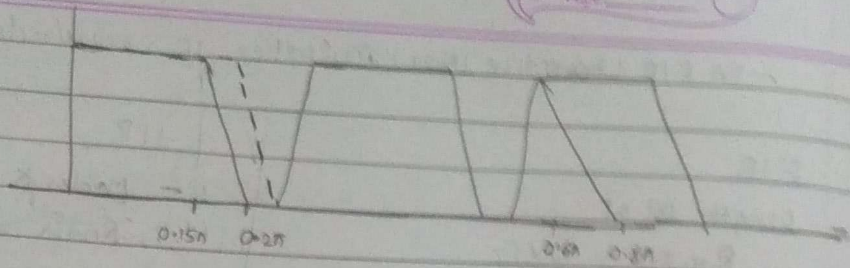


#2 in this we have to play with system, not with signal. if signal is there when we done up sampling.
- By using comb filter, we placed zeros $z \rightarrow z^L$
- $H(z) \rightarrow H(z^L)$ if we replace every delay with L times delay must be done in comb filter. and we get the response is 4 time ~~stn~~ compres of the H(z).

Replace every z^{-1} with z^{-4}

(multiplier is same) order is not change but delay is change





#3 use another LPF

- Which remove the unwanted parts and but cut off freq. is Higher than previous one ~~can we~~
  - Power ~~loss~~ is less
  - Order = Order of 2nd LPF filter + order of FIR filter
    [which is same for $F(z)$ & $I(z)$]

$\boxed{z^4}$

multirate is concen then it is good because we can use the noble identities for computation efficiency.

condition for

$$\frac{\omega_s}{\cancel{L}} < \frac{2\pi - \omega_s}{\cancel{L}} \qquad \frac{\omega s}{L} < \frac{2\pi - \omega s}{L}$$

$L \rightarrow$ no. of times it will expand

Kaiser's formula:

$$N = -20 \log$$

- in IFIR, we can use the multiplier and we $\overset{\text{can}}{}$ used here nobel identities.

Polypha

Problems Typica

H(z) o
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for ex
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Down sam

Date 15-Nov
Page ______

Polyphase Decomposition :

problem: Typical down sampler is

$$\rightarrow \boxed{H(z)} \rightarrow \boxed{\downarrow M} \rightarrow$$

→ here sampling is very high so time is very less. for that we required very accurate processor which is very costly.

H(z) operates only 1 at of M sampling times
H(z) operates on sampling freq. that is much higher
How do we shift $\boxed{\downarrow M}$ block ahead of $\boxed{H(z)}$

(Analogy to in 1 us complete process is done
5 us " " " "] – which one is

preferable for low price of processor.)
So here how can we get the second one block.

(P) What is polyphase decomposition ?

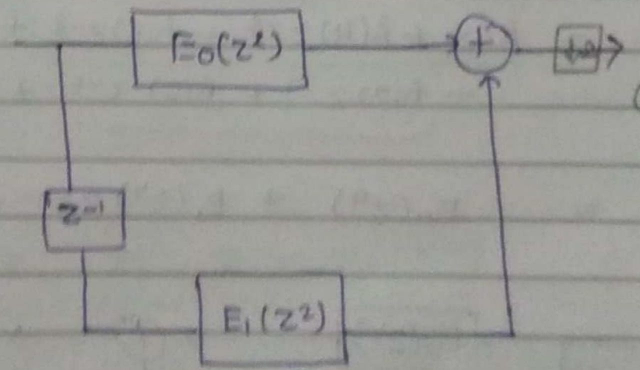
for ex:

$$H(z) = 1 + 2z^{-1} + 3z^{-2} + 4z^{-3}$$
$$= (1 + 3z^{-2}) + z^{-1}(2 + 4z^{-2})$$
$$= E_0(z^2) + z^{-1} E_1(z^2)$$

where $E_0(z) = 1 + 3z^{-1}$
$E_1(z) = 2 + 4z^{-1}$

order is not change here becuz here we can't added the (z^{-1}) any term.

H(z) ≡



Down sampler:

$$\rightarrow \boxed{H(z)} \rightarrow \boxed{\downarrow 2} \rightarrow$$

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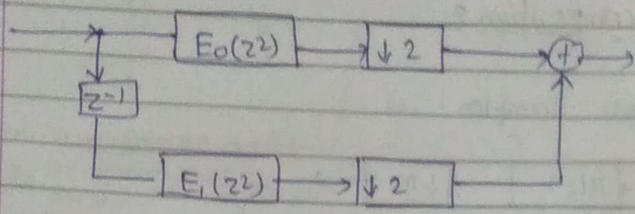
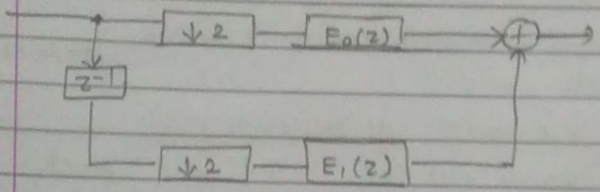


$\Downarrow$ Noble identities

(filter operates at low frequency)



its known as "polyphase decomposition"

↳ delay

↳ one or more

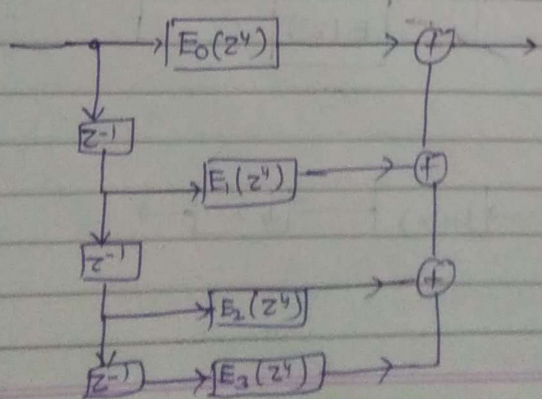
Example: $H(z) = h(0) + h(1)z^{-1} + h(2)z^{-2} + h(3) \ldots + h(4) + h(7)z^{-7} + h(8)z^{-8}$

4

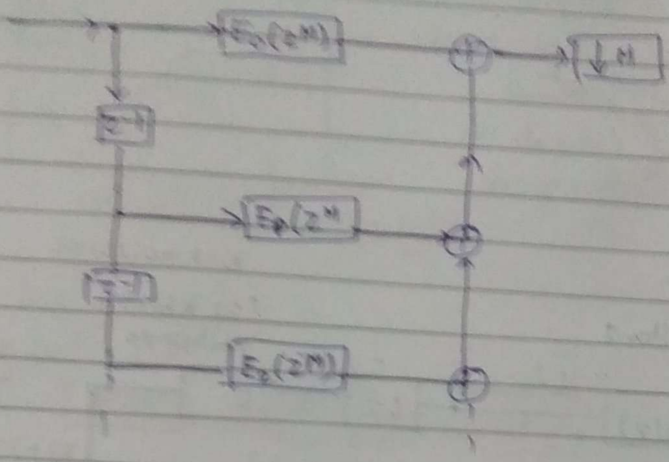
polyphase decomposition by factor 4

$$= \underbrace{h(0) + h(4)z^{-4} + h(8)z^{-8}} + \underbrace{h(1)z^{-1} + h(5)z^{-5}} + \underbrace{h(2)z^{-2} + h(6)z^{-6}} + \underbrace{h(3)z^{-3} + h(7)z^{-7}}$$

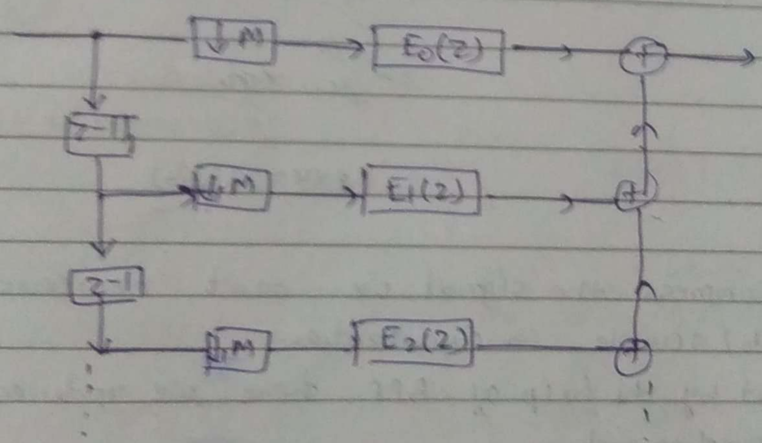
$$= E_0(z^4) + E_1(z^4)z^{-1} + z^{-2}E_2(z^4) + z^{-3}E_3(z^4)$$



generalized form :



III Noble identities



$$H(z) = \sum_{l=0}^{M-1} z^{-l} \; E_l(z^M)$$

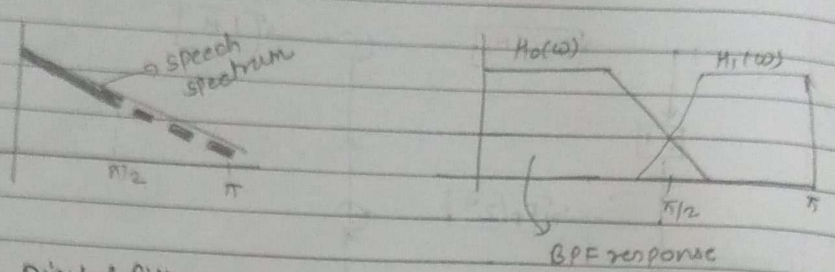
where $E_l(z) = \sum_{n=0}^{\infty} h(l+nM) \, z^{-n}$

$0 \leq l \leq M-1$

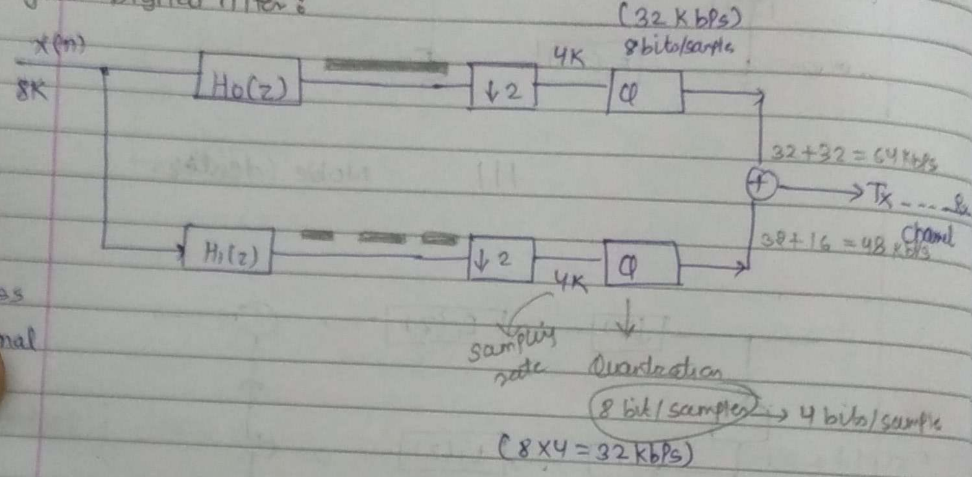
shifted impulse response

Also Refer slides

Digital filter Bank :

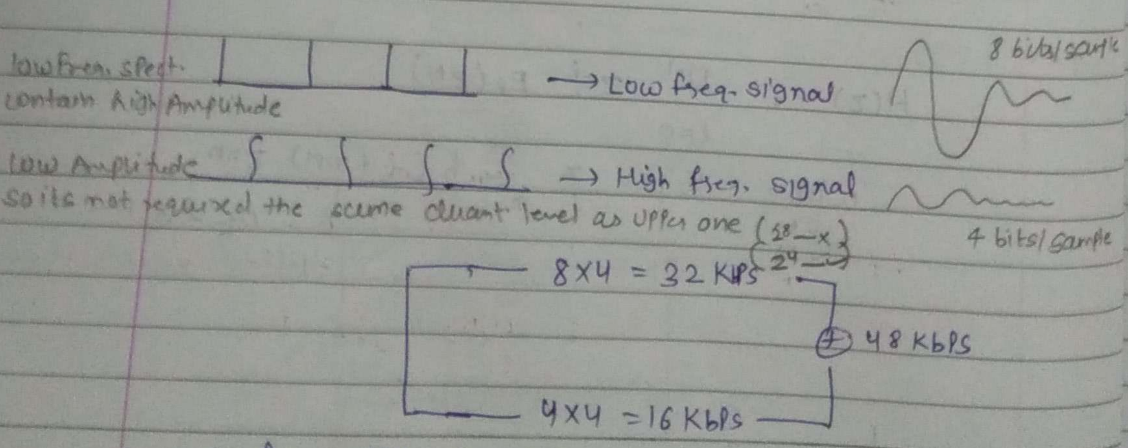


Analysis Digital filter :



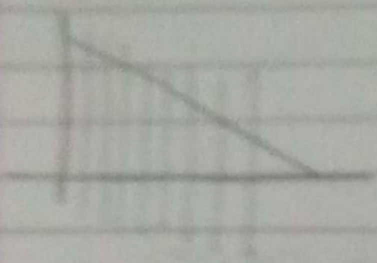
to compress signal

- To compress the signal, we can't decrease the bits / sample in quantization.
- But by the help of BPF, here we reduces the sampling rate.



So here bit rate will decreases

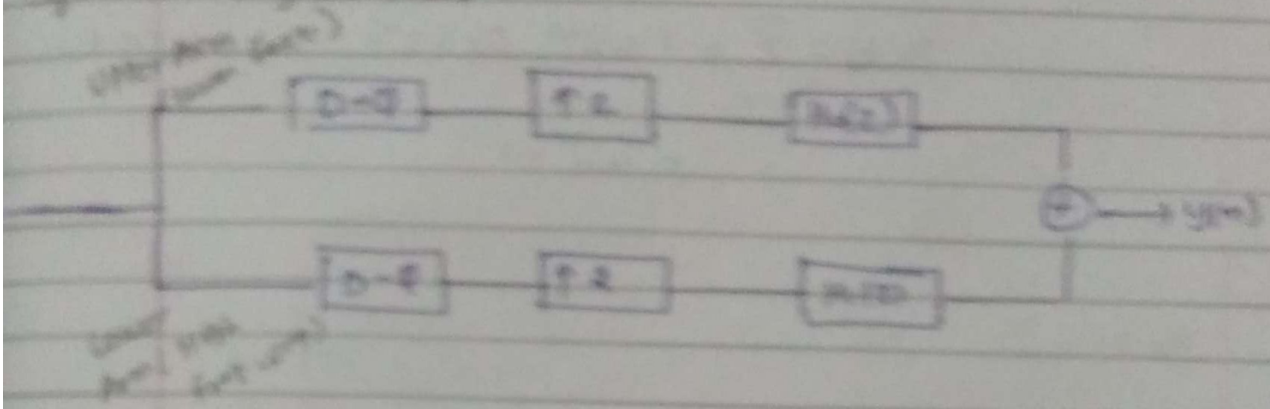
information spectrum will divided into the bands

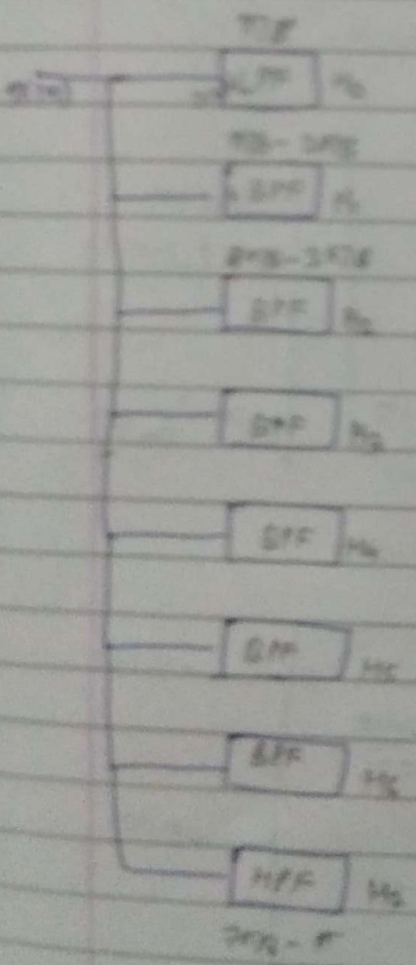


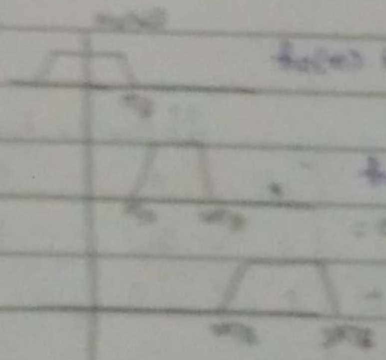
where multiple digital filter are required that known as Digital filter bank.

- in digital filter bank, multiple no of digital filter are there

Synthesis Digital filter:







$h_0(n) \longleftrightarrow H_0(\omega)$ ①

$h_1(n) \longleftrightarrow H_1(\omega)$ ②

How can we go ② to ① (modulation)

for that ② will be shifted

that the signal, not the [illegible]

so by doing fourier transform

$e^{j\omega_0 n}h(n) \longleftrightarrow H(\omega-\omega_0)$

so by doing that we have only design the proto LPF.

Uniform DFT filter Banks:

poly phase deco. $H_0(z) = \sum_{l=0}^{M-1} z^{-l} \, E_l(z^M)$

$z \rightarrow z \, W_M^{k}$

$H_k(z) = \sum_{l=0}^{M-1} z^{-l} \, W_M^{-kl} \, E_l(z^M W_M^{kM})$

$H_k(z) = \sum_{l=0}^{M-1} z^{-l} \, W_M^{-kl} \, E_l(z^M) \qquad 0 \le k \le M-1$

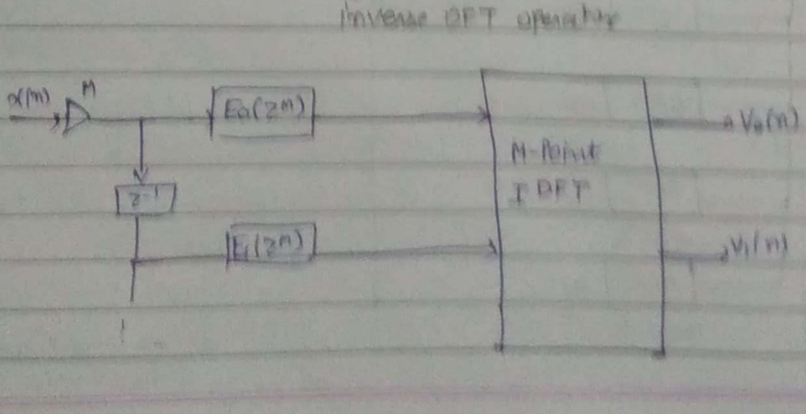
$\because \; W_M^{kM} = 1$

$\Downarrow$

$$H_k(z) = \begin{bmatrix} 1 & W_M^{-k} & W_M^{-2k} & \cdots \end{bmatrix} \begin{bmatrix} E_0(z^M) \\ z^{-1} E_1(z^M) \\ \vdots \end{bmatrix}$$

$\Downarrow$ matrix form

$$\begin{bmatrix} H_0(z) \\ H_1(z) \\ \vdots \end{bmatrix} = \underbrace{\begin{bmatrix} 1 & 1 & 1 & \cdots \\ 1 & W_M^{-1} & W_M^{-2} & \cdots \\ 1 & \vdots & \vdots & \vdots \\ \vdots & \vdots & \vdots & \vdots \\ 1 & & & W_M^{-(M-1)^2} \end{bmatrix}}_{\text{inverse DFT operator}} \begin{bmatrix} E_0(z^M) \\ z^{-1} E_1(z^M) \\ \vdots \\ z^{-(M-1)} E_{M-1}(z^M) \end{bmatrix}$$



Quadrature mirror filter bank

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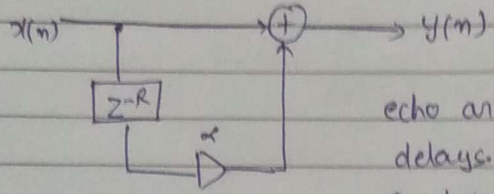
Date: 16-Nov

Applications of DSP:

- Echo
- flenging (time duration
- reverberation (reflected signal is listen within 1 msec)
  (100 of reflected signal, within 0.1 sec

Single echo filter:

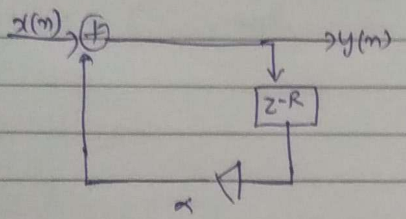
FIR:



- Single echo generated, i/p is taken echo are genrated by provide the delays.

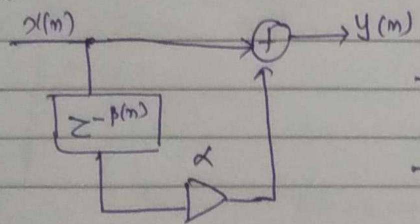
R → delays total time delay $= R \times \frac{1}{f_s}$

IIR:



Its provided many echos because O/P is feedback.

Flenging:



- delay changes by func of R
- time varying delay

Chorous Generator: