

EE2016

EE19B070

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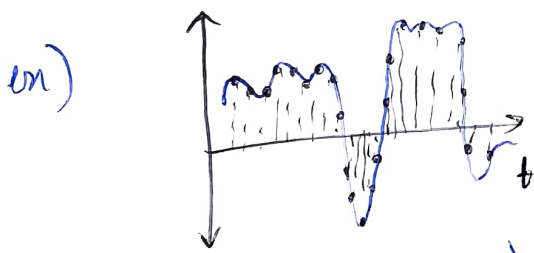
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PART-C (MIDSEM) REPORT :

1) Theory of FIR & FIR filters :

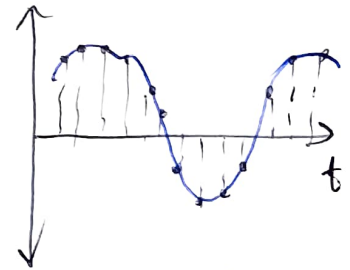
FIR stands for "Finite Impulse Response",  
and FIR filter is a "Finite Impulse response filter"

given a signal which is a superposition of various sinusoids with random frequencies, we can pass this signal (ONLY AFTER SAMPLING) through an FIR filter and the output <sup>sampled</sup> signal would only retain those sinusoids with frequencies we desire.



(frequency range is not known)

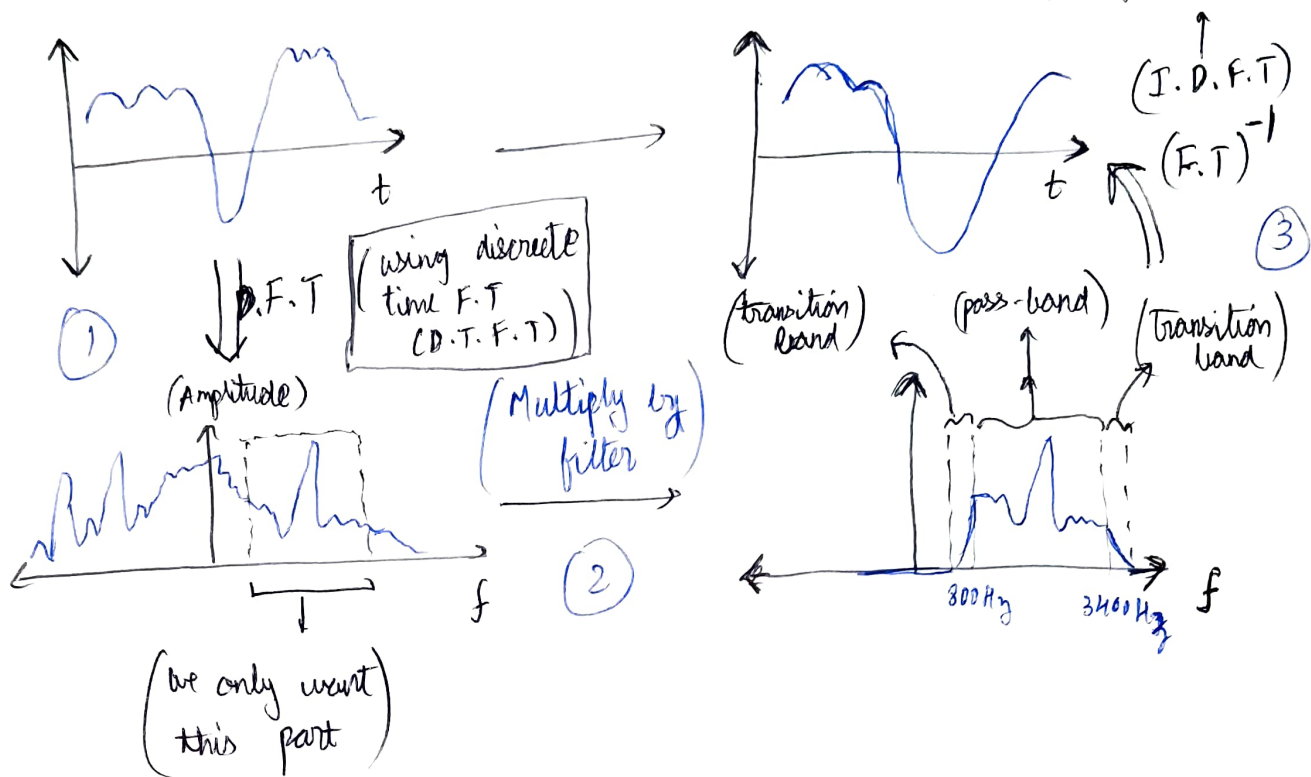
(passing through)  
FIR filter



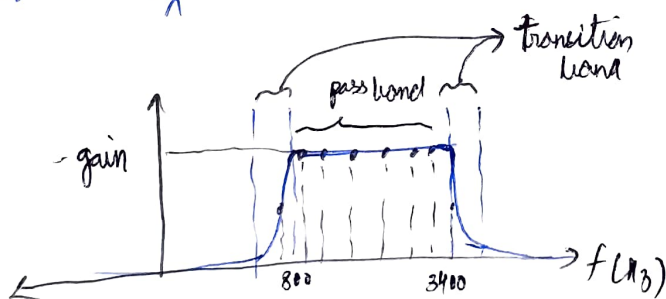
(frequencies only lie b/w  
300 Hz to 3400 Hz)

So the way we can do this is by,

(inverse discrete  
fourier transformation)



the filter in ~~freq.~~ <sup>freq.</sup> domain would be,



We <sup>also</sup> know that multiplication in freq. domain is the same as convolution in the time domain.

In the above schematic of realising a filter, the steps are:

- ① fourier transforming Input into freq. domain (SAMPLED INPUT) (using DTFT)
- ② multiplying it with the required filter
- ③ Inverse fourier transforming the result. (GIVES SAMPLED OUTPUT)

∴ the DT convolution,

(2)

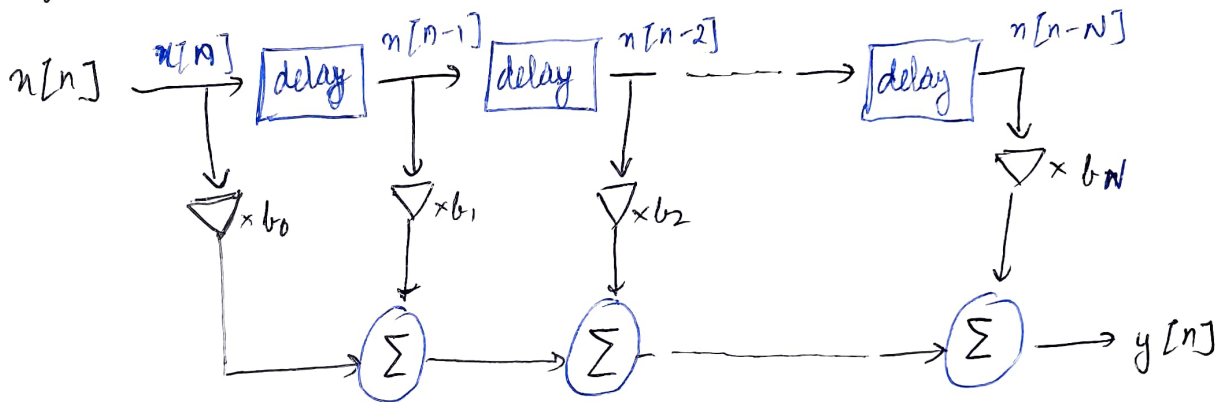
$$y[n] = \sum_{k=0}^N x[k] \times f[n-k] \equiv \sum_{k=0}^N x[n-k] \times f[k]$$

$$= x[0] \times f[n] + x[1] \times f[n-1] + x[2] \times f[n-2] \dots + x[N] \times f[n-N]$$

(commonly represented as

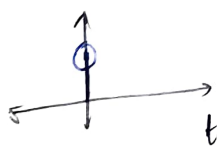
$$= b_0 x[n] + b_1 x[n-1] \dots + b_N x[n-N])$$

∴ <sup>N</sup> Tap FIR filter general design =



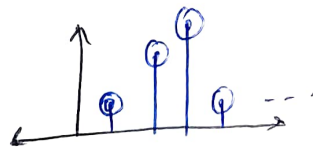
FIR = finite impulse response

= the response of the system when the simplest unit impulse function is given as the input



input = impulse response

system



output = impulse response

## Discrete time convolution :

is an operation on 2 discrete time signals defined by the integral

$$y[n] = (f * g)[n] = \sum_{k=-\infty}^{\infty} f[k]g[n-k]$$

## Discrete Fourier transformation (D.F.T) :

if we sample an input func. at  $\omega_k$ , S.T

$$\omega_k = \frac{2\pi}{N}k, \quad k=0,1,2,\dots,N-1$$

$$\therefore X[k] = X\left(e^{j \times \frac{2\pi}{N}k}\right) \quad (\text{DFT of } n[n])$$

$$= \sum_{n=0}^{N-1} n[n] e^{-j \times \frac{2\pi}{N} \times k \times n}$$

& IDFT (inverse discrete Fourier transformation) :

$$n[n] = \frac{1}{N} \sum_{k=0}^{N-1} X[k] e^{j \left(\frac{2\pi}{N}k\right)n} \quad (\text{IDFT of } X[k])$$

2) On the Fixed Point Arithmetic used!

(3)

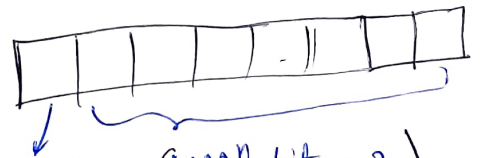
While searching for a suitable scaling factor, I had 2 goals in mind.

1) Although fractional F-P multiplication is also available, I <sup>only</sup> try to use integer multiplications, (i.e) after the inputs / coeff are multiplied with suitable scaling factor then they should be integers (then we can ignore the fractional part)

2) I need to try and use as many bits as possible so that we get the maximum precision of the filter.

$\therefore$  as we can see, the coeff are all lesser than "1" & when we give the input signal, we generally use an amplitude of "1"  $\Rightarrow$  the sampled input values are also lower than "1".

$\therefore$  in a byte, there are 8 bits



$\therefore$  since all inputs & coeff are (sign bit = 1) (mag^n bits = 7)

less than 1, we can

SCALE THEM using  $2^7$ , so that the mag^n of the scaled output is always lower than 7 bits ( $2^7$ ), and then the first bit can be used for the sign.



$$M \text{ bit of output} = (N_1 + N_2) + \log_2(N) + 1 \rightarrow \text{sign}$$

$$N_1 = \text{inputs} = \cancel{2} (7) \text{ bits} + 1 \text{ sign}$$

$$\therefore N_1 = 7$$

$$N_2 = N_1 = 7 \quad (N_2 = \text{coeff} = 7 \text{ bits} + 1 \text{ sign})$$

$$N = \text{no. of additions} = 5 \quad (5 \text{ coeff.})$$

$$\therefore M = 7 + 7 + \log_2 5 + 1$$

$$= 7 + 7 + 2.322 + 1$$

$$= 17.322$$

$$\therefore M = 18 \text{ bits} = \text{output}$$

$$= 3 \text{ bytes} \quad (\text{but we don't fully use } \overset{\text{one of}}{\text{the}} \text{ bytes})$$

#### 4) Actual implementation Details:

a) INPUTS into the AVR,  
we have 2 inputs.

1st Input set = COEFF, in the program I've submitted,  
there are 5 1byte inputs given.  
(signed)

this is given in a .db form,  $\therefore$  its stored in  
the program memory.

When we use a pointer to find the <sup>memory</sup> location its  
stored in, we see that its stored in

" 0x 0072, prog" in the "prog FLASH"

as there are 5 inputs in COEFF, they are stored in

0x 0072, prog upto 0x 0076, prog

2nd INPUT set = INPUTS, in the <sup>prog.</sup> I submitted, we  
have 10 1byte signed inputs.

they are stored in

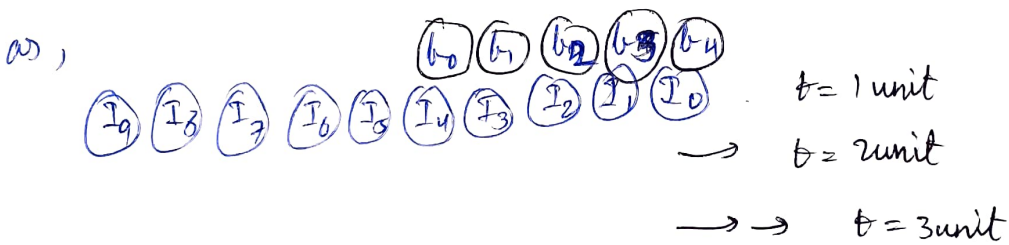
" 0x 0078, prog" in the "prog FLASH"

0x 0078, <sup>prog</sup> upto 0x 0082, prog

~~the~~ I then store the COEFF inputs into <sup>the</sup> SRAM, I store them from 0x0060 to 0x0064, data in the "prog FLASH"

Outputs : <sup>many</sup> 3 byte outputs which are stored successively, with the 1<sup>st</sup> byte of the 1<sup>st</sup> output stored at 0x008C.

$$\text{no. of 3 byte outputs} = (\text{No. of } \overset{\text{INPUTS}}{I_i}) - (\text{No. of COEFF.}) + 1$$



(pls. refer FIR diagram in page 2)



Output = 3 bytes,



(5)

b) no, I have not used circular buffers

c) Initially, the COEFF inputs are given in the .db form & they are initially stored in

"0x0072, prog." which is the program memory, (0x0072 to 0x0076)

Next, I copied these value into the SRAM in the "0x0060, data" location

(0x0060 to 0x0064)

d) Multiplication:

In the Small Loop, 5 multiplications Take place and they all need to be added together.

for this use MULS, ADD and ADC

In the first multiplication, we multiply the value pointed by z and that by x (points to 0x0064 first time),

We put them in registers R22 & R24, Then

we MULS R22, R24 which is the signed multiplication of R22 & R24, the resulting bytes (signed) are

stored in R1, R0.

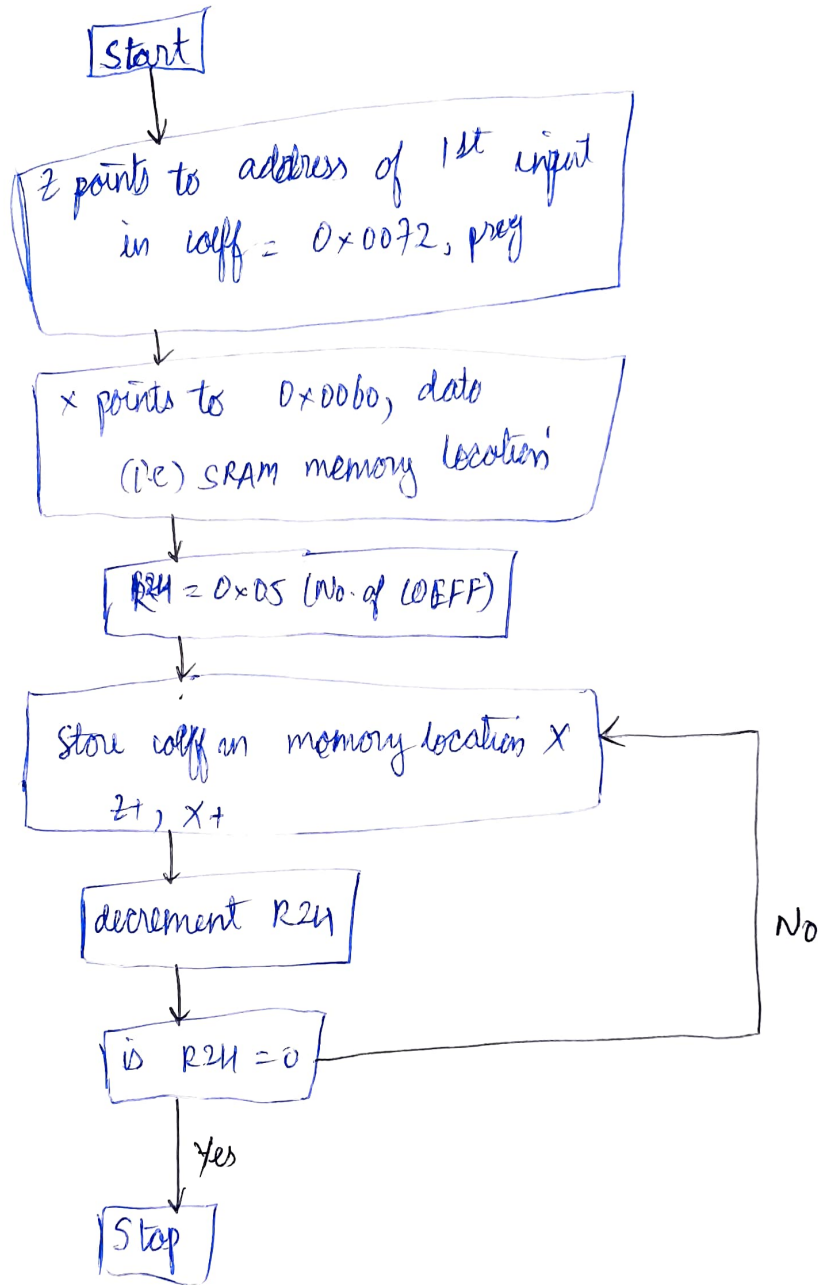
then we ADD R0, R16 then we ADC R1, R17. (ADD with prev. carry) if there's any overflow

∴ now, the sums are stored in R18, R17, R16

## Flow-Chart:

(6)

a) to store <sup>COEFF</sup> memory in SRAM,



## b) Program flow chart

