Declaration: I/We confirm that this submission is my own work. In it, I give references and citations whenever I refer to or use the published, or unpublished, work of others. I am aware that this course is bound by penalties as set out in the College examination offenses policy. Signed: Hao Dong [illegible] [illegible]

REAL TIME DIGITAL SINGAL PROCESSING

LAB 4 REPORT

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CONTENT

MATLAB

1.1 Filter Design

The FIR filter required to build in MATLAB has specifications shown in Figure 1.

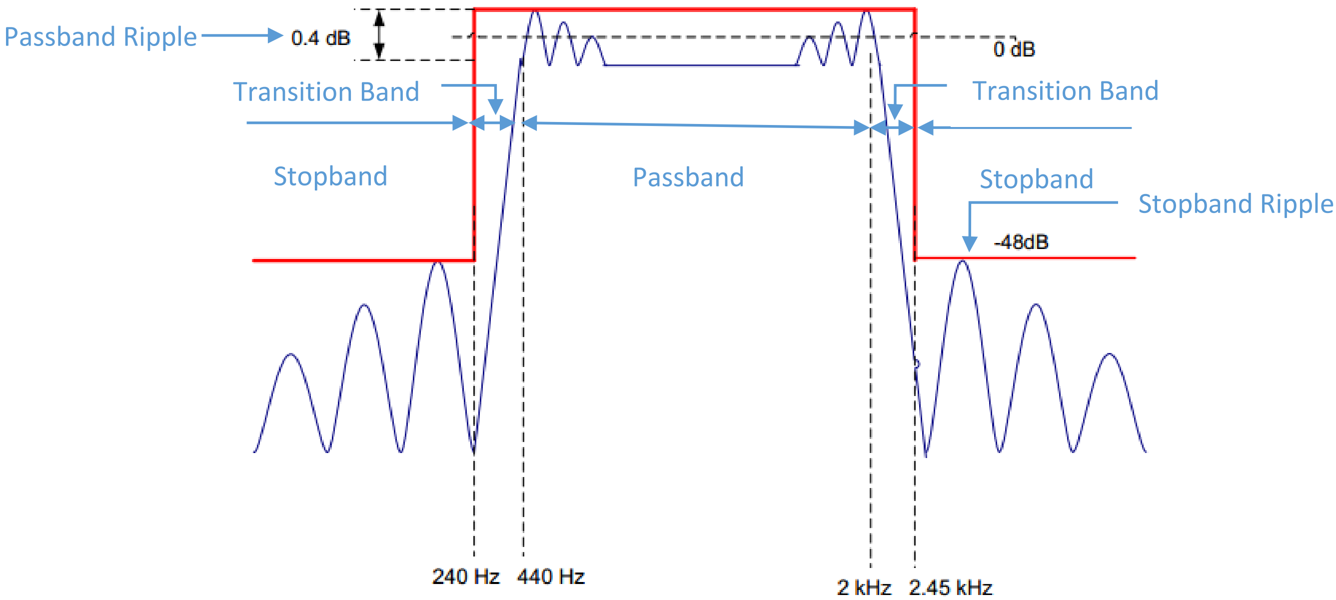


Figure 1. FIR specifications

In MATLAB, the Parks-McClelland algorithm is suggested to approximate required filters by using *firpmord* and *firpm* functions. Code shown as the following:

Figure 2. MATLAB filter design code

```
close all;
clear all;

f1=240;f2=440;f3=2000;f4=2300;% frequency boundary
f=[f1,f2,f3,f4];%make array of frequencies

rp=0.4;% ripple in dB
sa=48;% minimum stop band attenuation in dB
dev1=(10^(rp/20)-1)/(10^(rp/20)+1);%calculate pass band deviation
dev2=10^(-sa/20);%calculate stop band deviation
dev=[dev2,dev1,dev2];%make array of deviations

Fs=8000;%specify sampling rate
a=[0,1,0];%specify amplitude

[N,Fo,Ao,W] = firpmord(f,a,dev,Fs);%function to approximate filter
coefs = firpm(N,Fo,Ao,W);%function to calculate frequency coefficients.
```

```
freqz(coefs,1,1024,8000)%Plot frequency and phase response
H = tf(coefs,1);%derive transfer function
figure;
pzmap(H);%plot pole and zero map of filter
grid on;

%The following code store filter coefficients in format readable for c.
fileID = fopen('fir_coef.txt','w');
fprintf(fileID,'double coefs[]={');
for i = 1:length(coefs)
    fprintf(fileID,'%f,',coefs(1,i));
end
fprintf(fileID,'};');
fclose(fileID);
```

For *passband*, according to the equations:

$$Gain\ Max\ in\ dB = 20\log(1 + DEV)$$

$$Gain\ Min\ in\ dB = 20\log(1 - DEV)$$

We know that the difference between these is the ripple in dB, *rp.*

Thus, we can combine them:

$$DEV = \frac{10^{\frac{rp}{20}} - 1}{10^{\frac{rp}{20}} + 1}$$

Same procedure for stopband, the function is shown below:

$$\frac{V_{out}}{V_{in}} = 10^{-\frac{rp}{20}}$$

Since it is a FIR filter, when we use function *freqz* we need to set the second parameter to 1 (FIR only has one pole). Meanwhile, we use 1024 (resolution) and 8000 (recovering the frequency) to print the graph nicely.

1.2 Proof of the expected frequency response

Frequency and phase response as well as pole map are included in Figure 3, from which we can see that specifications described are satisfied well. However, at the end of pass band and middle of transition band exist two bumps, which may bring in potential risk, shown in Figure 4.

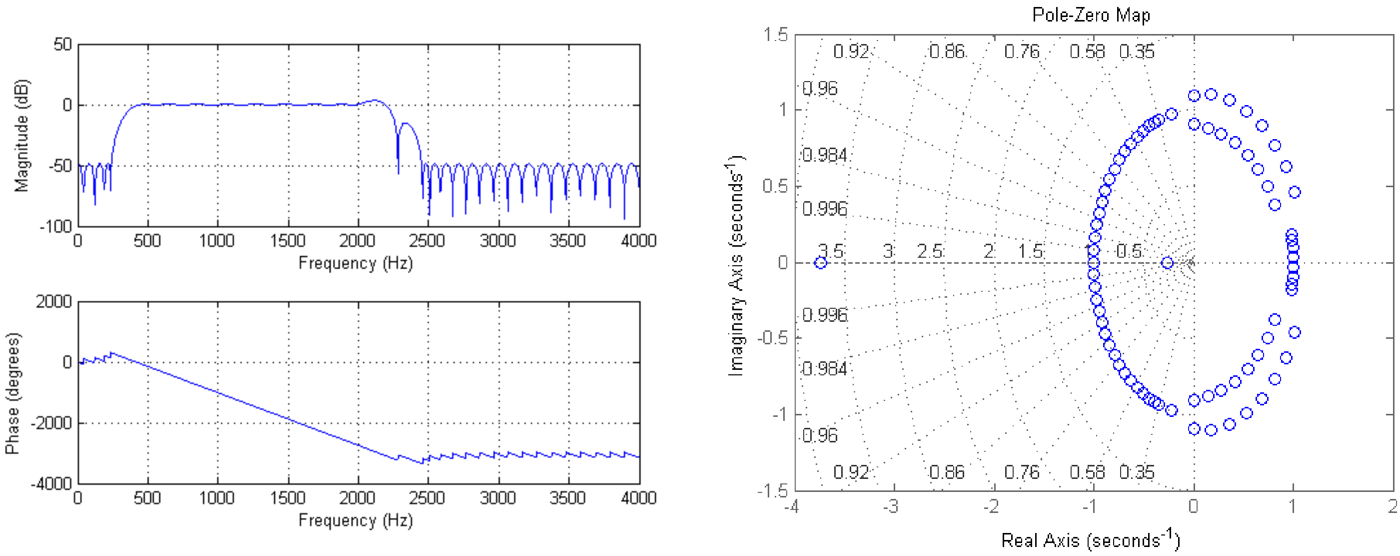


Figure 3. (Left) Frequency and phase response. (Right) Pole and zero map

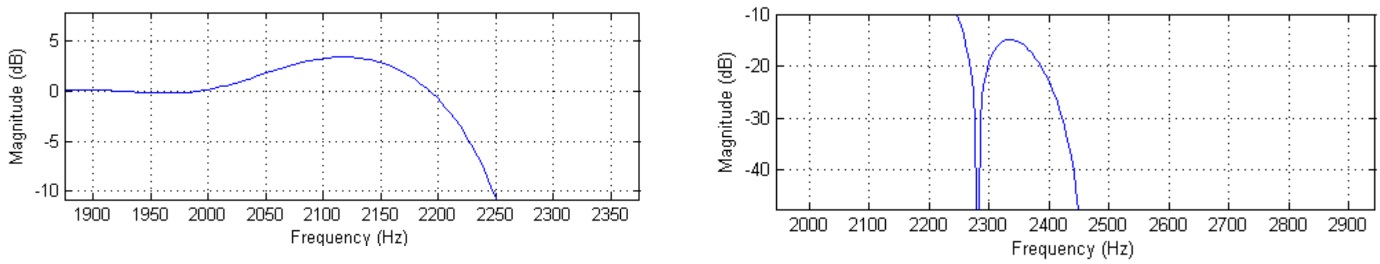


Figure 4. (Left) Bump at the end of pass band. (Right) Bump in the middle of transition band

Accordingly, we slightly narrow the transition band range by change the boundary from 2.45 kHz to 2.3 kHz. In z-domain, it is equivalent to a small shift of zeroes. Frequency response and pole map of modified filter is shown in Figure 5, with pole shift marked. At position **A**, poles are closer to each other, which avoids a single zero to pull down amplitude too early so that the bump in middle of transition band is removed. At position **B**, a pole pair is moving to opposite direction, which reduces gain at that frequency and removes the bump at the end of pass band.

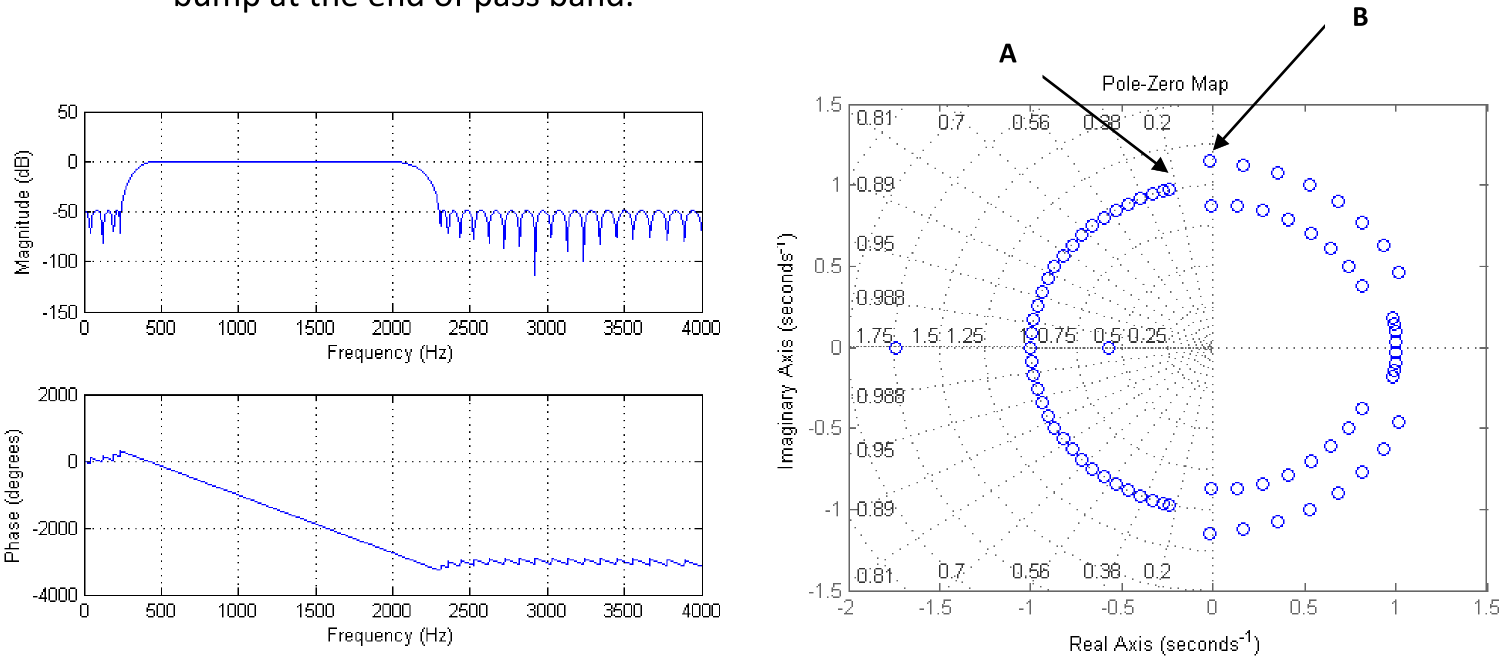


Figure 5. (Left) Frequency and phase response. (Right) Pole and zero map

Actually, we can increase the order by 1 or 2 to make sure the filter fits the specs (*firpmord* only generate the approximate order required for filter). But once we use this way, it will make our program complicated (more calculations needed for convolution function). We prefer to minimise the N as small as possible. The order of N (77) is calculated.

Coefficients of filter with modified boundary frequency is then included and used in Code composer. Method of using coefficients is explained in next section.

Code explanation

2.1 Operation principles

The basic idea of this section is similar to the previous lab. Interrupt ISA_AIC function is called every time signal is received. Input sample is read from codec, using mono_read_16Bit (). Current and some past samples are stored and convoluted with FIR filter. The corresponding time equation of filtering is

$$y_n = b_o x(n) + b_1 x(n-1) + \cdots b_M x(n-M)$$

For any instant time n, an output is generated based on current and past input samples. Then it is expected that actual output varies as frequency, as described in the frequency response graph in MATLAB.

FIR filter coefficients are saved in a text file in an array format:

double coefs[]={0.002154,….};

The text file is then saved in the same folder as c file, and included using command:

#include "fir_coef.txt"

2.2 Non-circular FIR filter design

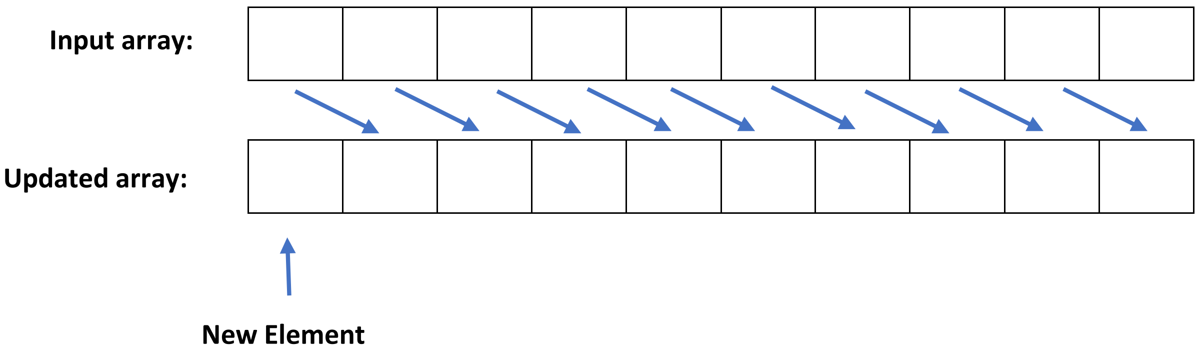
Code:

```
#include "fir_coef.txt"

#define N 78
short x[N]={0}; //initialise array to zero

void ISR_AIC()
{
	int i;
	for(i=N-1;i>0;i--){//loop through every element from last to first
		x[i]=x[i-1];//shift elements to right and delete the last one
	}
	x[0]=mono_read_16Bit(); //store input
	non_circ_FIR();
}

void non_circ_FIR()
{
	int i;
	double y=0;
	for(i=0;i<N;i++){ //loop through the array and perform convolution
		y+=x[i]*coefs[i];
```

```
        }
        mono_write_16Bit((short)y);//write to output port
}
```

A naïve implementation of non-circular FIR filter is shown above. Necessary files and filter coefficient number are defined and included in the beginning. Input array is initialised to zero.

Inside the interrupt function is a for-loop which shifts all elements one position to right and delete the last element. After that, input is read and saved in position 0. Figure is shown below.



Since the order is 77, we need 78 parameters in our convolution function.

The function non_circ_FIR () performs convolution by looping through the whole array and doing multiply accumulation on input elements and corresponding filter coefficients.

2.3 Code efficiency with compiler optimisation

Table 1. Optimisation level and clock cycle

--opt_level	Clock cycle
no	5145
0	4208
2	1024

Table 1 shows that clock cycle number is reduced significantly with optimisation level. Notice that bread points are placed at the beginning and end of ISR_AIC function, which measures the total time cycle of interrupt function, including reading and writing samples. Each cycle count is the best case found after running code a few times. In this report, all clock cycle are measured using this method.

The opt_level is an option for user to optimise their code.

Table 2. Optimisation priority

--opt_level	performance
no	Disable optimisation
0	Optimisation priority is compilation time and debugging ease
2	Compiler optimise primarily for performance

Circular FIR filter design

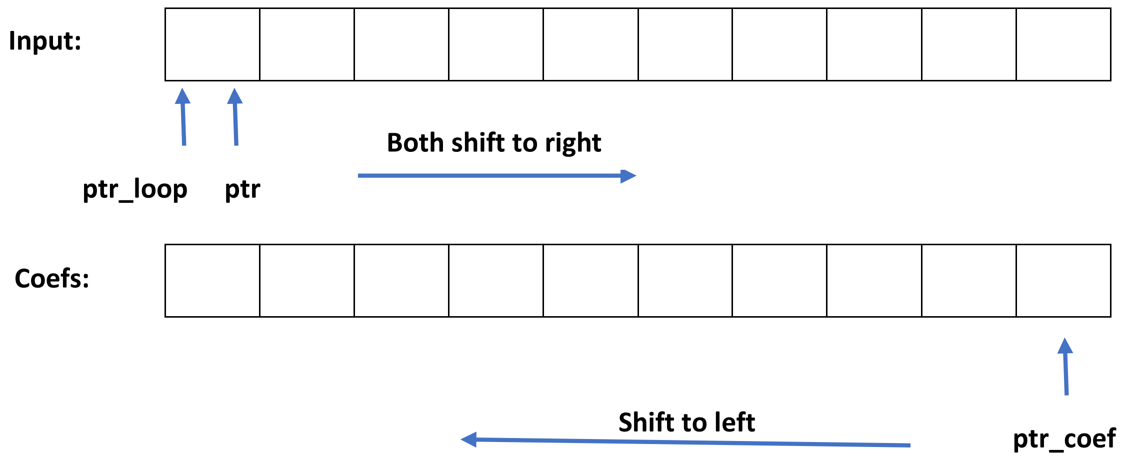
In this section, three different implementation of circular FIR filters are constructed and compared, including naïve implementation, using linear phase property and using double array size. The fundamental ideal of circular buffer is to store input data along the array repeatedly, which avoid the trouble to shift input data every time a new input is logged.

3.1 Naïve Circular FIR Filter Design

This is a basic realisation of the idea of a circular buffer.

```
#include "fir_coef.txt" //include filter coefficients
#define N 78       //filter coefficient number
short x[N]={0}; //initialise array to zero
short * ptr=&x[0]; //initilise the pointer to first element

void circ();

void ISR_AIC()
{
      circ();
}

void circ(){
      int i;
      double y=0;
      short* ptr_loop;         //pointer to loop through input array
      double* ptr_coef=&coefs[0];//pointer to filter coefficient array

      *ptr=mono_read_16Bit(); //read input
      ptr_loop=ptr;            //initilise loop pointer to latest reading

      for(i=0;i<N;i++){
            y+=*ptr_loop*(*ptr_coef);//perform convolution
            if(ptr_loop==&x[0]){
                  ptr_loop=&x[N-1]; //loop back to beginning if loop
pointer reaches end
                  ptr_loop++;       //adjustment to make sure pointer at
0 position at
            }                       //the end of for loop
            ptr_coef++;             //increment filter pointer
            ptr_loop--;             //decrement loop pointer
      }

      if(ptr==&x[N-1]){ //pointer to store input loop back to beginning
            ptr=&x[0];  //after reaching the end of the array
            ptr--;
      }
      ptr++;            //increment pointer

      mono_write_16Bit((short)y);
```

In the naïve implementation, we built an array with size N and used two pointers *ptr_coef* and *ptr_loop*, which controls filter coefficient array and input array respectively.



Ptr is a global input pointer which is initialised to the first position of input array and incremented each time new data is logged. So *ptr* always points to the latest element and the oldest element is always at *ptr+1*. Therefore, *ptr* should multiply with *coefs*[0], *ptr+1* should multiply with *coefs*[1],etc. In our program, we perform convolution from the oldest sample to latest, so loop pointer shift to right while coefficient pointer shift from right to left.

Notice that there are some special cases when pointers are required to be manually set to the correct position to show circularity.

The first case is inside the for-loop. Every time loop pointer reaches the right most element after multiply accumulation, it is reset to position 0. Loop pointer is also incremented to cancel out the effect of decrement at the end of for-loop.

```
if(ptr_loop==&x[0]){
ptr_loop=&x[N-1]; //loop back to beginning if loop pointer reaches end
ptr_loop++;       //adjustment to make sure pointer at     0 position at
}
```

Another case is about input pointer, which is incremented every time interrupt function is called. Similarly, it should loop back to position 0 once reaches end of array.

```
if(ptr==&x[N-1]){ //pointer to store input loop back to beginning
```

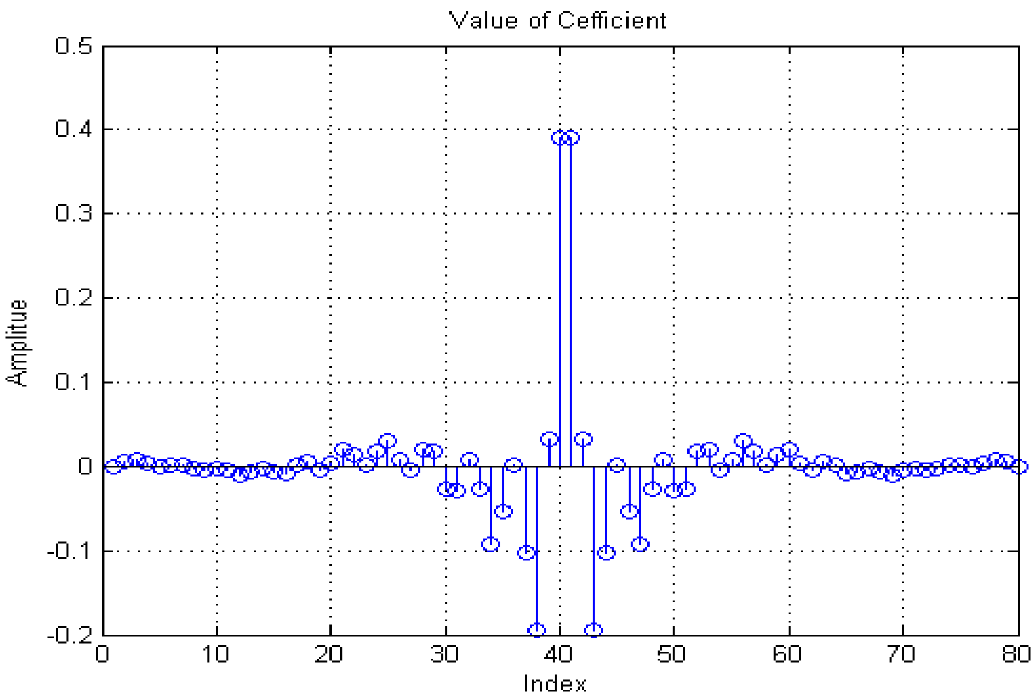
```
ptr=&x[0];  //after reaching the end of the array
ptr--;
}
```

3.2 Improvement on Basic Design

Linear phase

Since the function we used in MATLAB is *firpmord* which follows the Parks-MaClellan algorithm, in which case we will get a linear phased FIR filter. The trick of linear phase filter (also referred to as constant group delay) is the coefficients within it are real and symmetric. The details are shown below:

$$index_n = index_{(N-n-1)}$$

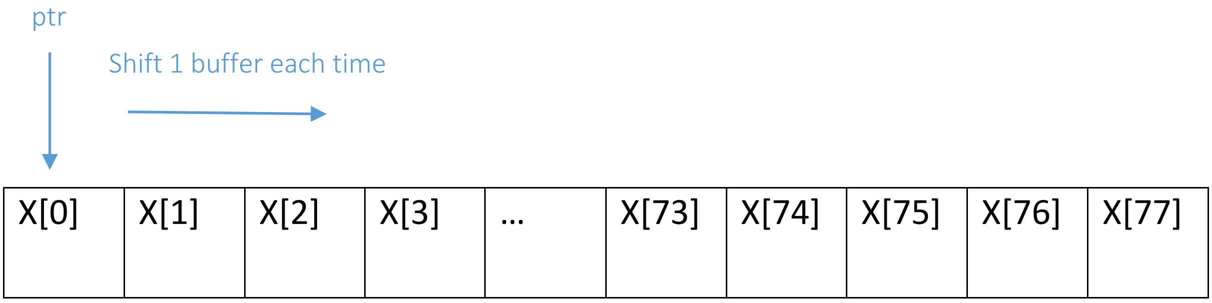


Because the newest and the oldest have the same weights, it will reduce nearly half cycles if we combine them together. We use two pointers to represent the newest and the oldest value stored in buffer (named p_f and p_l respectively), and manipulating them simultaneously by the function:

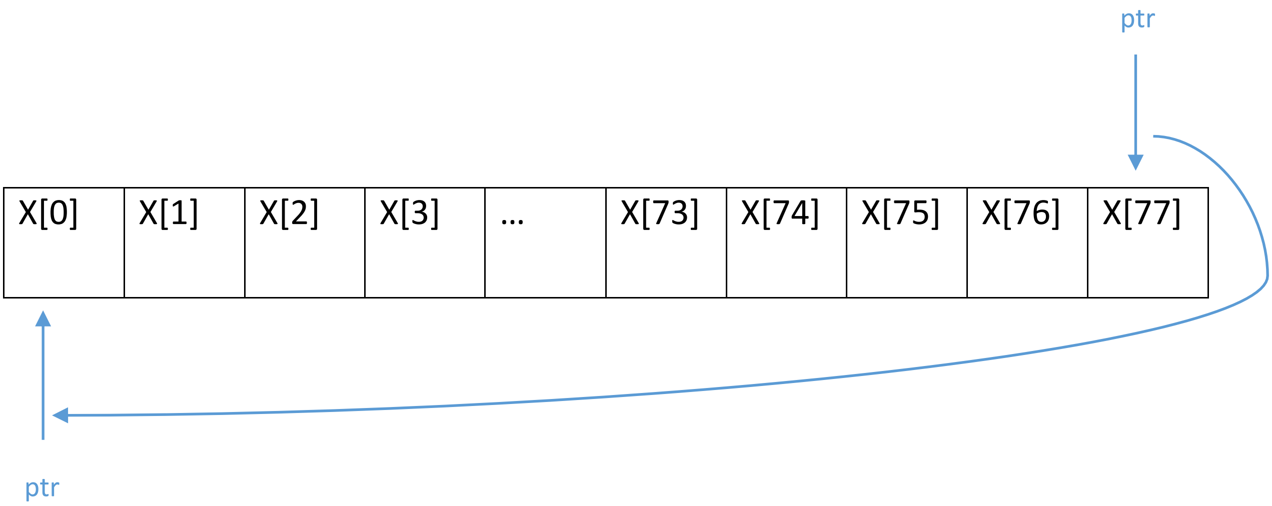
$$Output = \sum_{i=0}^{(\frac{N}{2}-1)} coefs[i] \times (x[i] + x[N-i-1])$$

```
	void linear phase(){
	//declare parameters initially, including type and name.
	int i;
	// we set y to double which avoid overflow.
	double y=0;
	//p_f pointed to newest number, p_l pointed to oldest number.
	short* p_f=ptr;
	short* p_l=ptr;
	double* p_fir=&coefs[0];
	//get readings from input port.
	*ptr=mono_read_16Bit();
	//increasing p_l by 1 to make sure it points to the oldest number.
	p_l++;
	//Since the buffer size is fixed, we need to reset the pointer
index
	//when it reaches boundary.

	if(ptr==&x[N-1]){
		p_l=&x[0];
		ptr=&x[0];
		ptr--;
	}
	//we use a for loop to perform convolution function.
	for(i=0;i<N/2;i++){
		y+=*p_fir*(*p_f+*p_l);
		//avoding overflow, we need to reset p_f once it reaches
		//boudary.
		if(p_f==&x[0]){
			p_f=&x[N-1];
			p_f++;
		}
		//avoding overflow, we need to reset p_l once it reaches
		//boudary.
		if(p_l==&x[N-1]){
			p_l=&x[0];
			p_l--;
		}
		//changing coefficient to meet our function
		p_fir++;
		p_f--;
		p_l++;
	}
      //increase pointer by 1. get ready for next interrupt
	ptr++;
	//write down the result to output port.
	//since it is 16bits, we need to change the type from double to
	//short.
	mono_write_16Bit((short)y);
}
```

For each interrupt, we will add ptr by 1 to deflect losing our old data (shift the position of our pointer by 1).



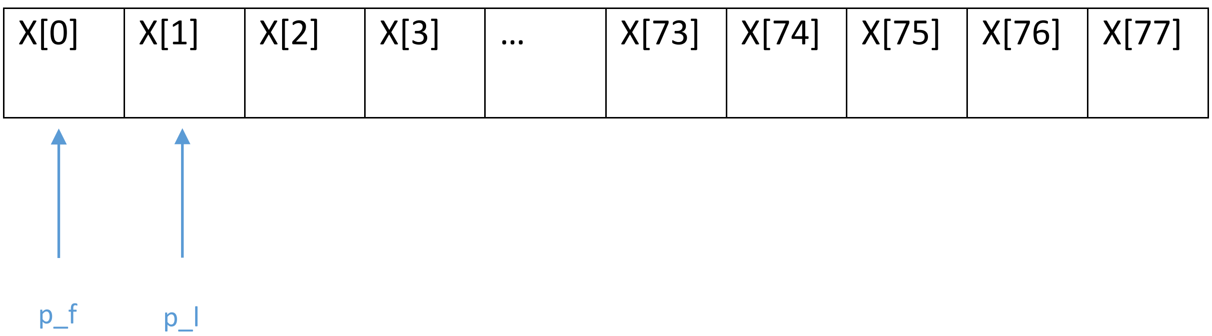
There is a special case, once our pointer reaches boundary, we have to pull it back to the initial buffer (avoiding overflow).



We add an offset to ptr to maintain the whole procedure.

```
if(ptr==&x[N-1]){
            p_l=&x[0];
            ptr=&x[0];
            ptr--;
      }
```

Since the ptr is moving from left to right, the oldest value will next to the newest value.



As before, we need to change the position of them when either of them reaches boundary.

```
//avoding overflow, we need to reset p_f once it reaches
            //boudary.
            if(p_f==&x[0]){
                  p_f=&x[N-1];
                  p_f++;
            }
            //avoding overflow, we need to reset p_l once it reaches
            //boudary.
            if(p_l==&x[N-1]){
                  p_l=&x[0];
                  p_l--;
            }
```

Offset is used here as well.

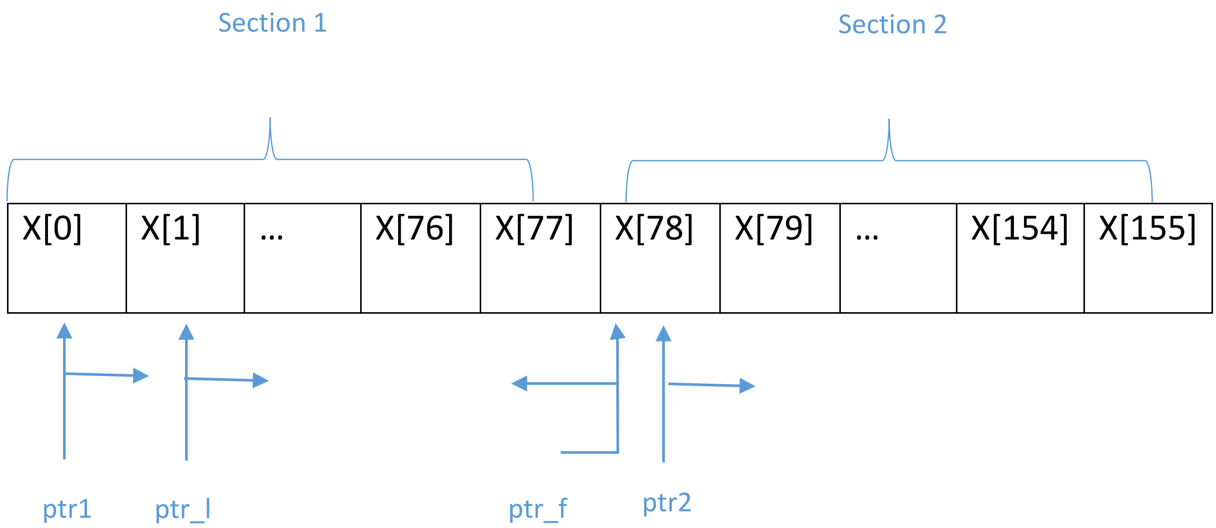
Double Memory

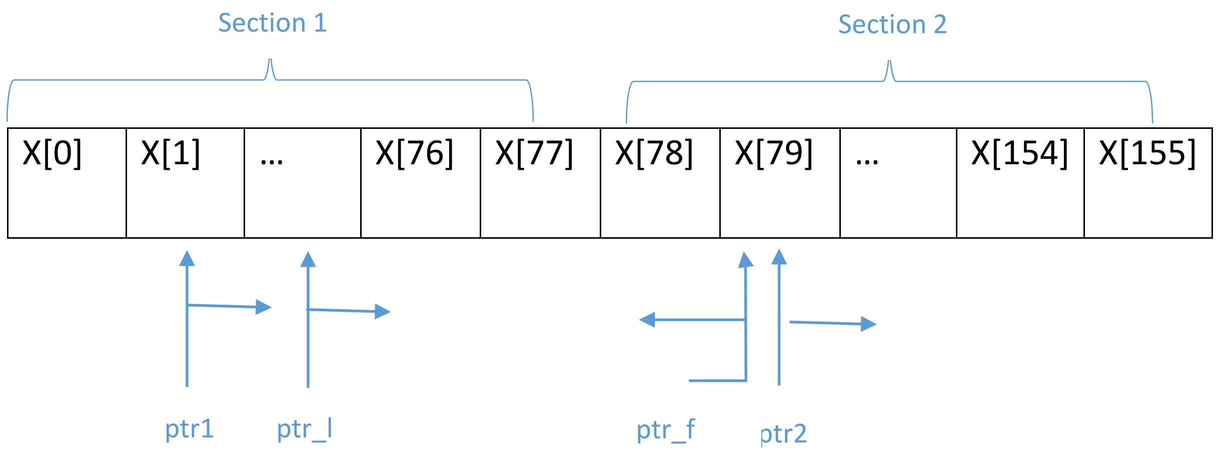
Using the property of linear phase, we do need to execute several branches depending on conditions are satisfied or not. Since the judgements are included in for loop which complex our code and wasting time. We can simplify this by double memory, in which case judgement is eliminated.

In order to double our original data set, we add one more pointer (ptr2) whose value is synchronise with ptr1, but with a different place. The starting point of *ptr_f* is correspond to ptr2 (input will always be the newest value), the gap between *p_f* and *p_l* is fixed which is 76, so by processing the for-loop, *p_f* and *p_f* will never overlap.

```
void Double memory(){
      //declare parameters initially, including type and name.
      int i;
      // we set y to double which avoid overflow.
      double y=0;
      //p_f pointed to newest number, p_l pointed to oldest number.
      short* p_f=ptr2;
      short* p_l=ptr1;
      double* p_fir=&coefs[0];
      //get readings from input port.
      *ptr1=mono_read_16Bit();
      //copying data to fulfil the buffer.
      *ptr2=*ptr1;
      //increasing p_l by 1 to make sure it points to the oldest number.
      p_l++;
      //Although the buffer size is doubled, we still need to reset the
pointer index
      //when it reaches boundary.
      if(ptr1==&x[N-1]){
           ptr1=&x[0];
           ptr1--;
      }
      if(ptr2==&x[2*N-1]){
             ptr2=&x[N];
```

```
            ptr2++;
        }
        //using the property of linear phase, we match the inputs with
        //the same coefficient together.
        for(i=0;i<N/2;i++){
              y+=*p_fir*(*p_f+*p_l);
              p_fir++;
              p_f--;
              p_l++;
        }
        //increment pointer avoiding over-written.
        ptr1++;
        ptr2++;
        mono_write_16Bit((short)y);
}
```





When ptr 2 and ptr 1 reach their boundary, since x[78] was the oldest value, it won't influence the result if ptr_l jump to section '2'.

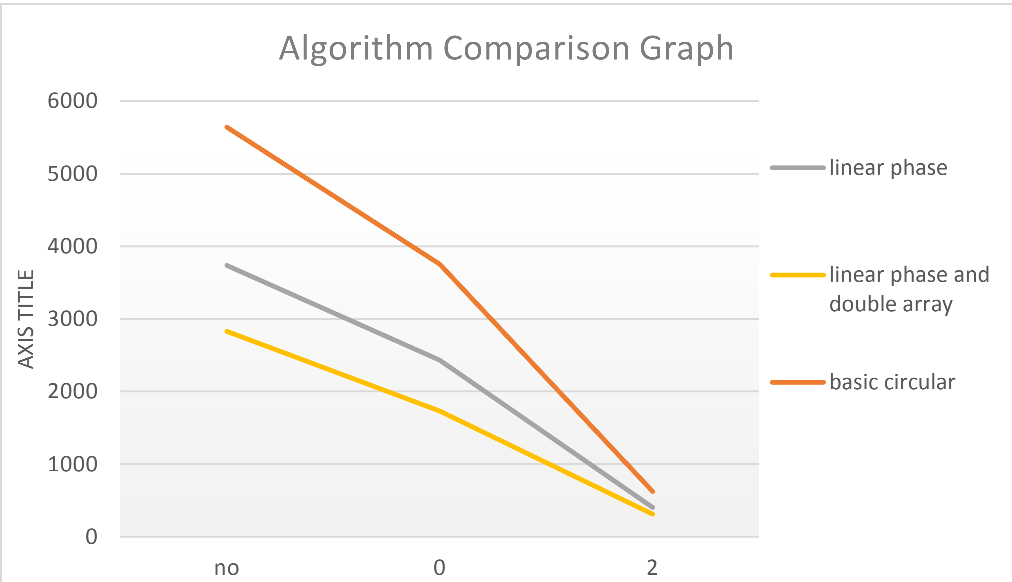
3.3 Code Efficiency and comparison

Code performance with and without compiler optimisation is measured as the following:

Table 2. Optimisation level and clock cycle

--opt_level	Basic Circular	Circular using linear phase	Circular with linear phase and double input array
no	5640	3737	2826
0	3756	2432	1729
2	623	403	313

Table 2 shows that the code efficiency is improved significantly as optimisation level increases.



It can be seen from the figure that compiler optimisation significantly improves code performance, especially at optimisation level 2, where pipelining, loop unrolling and optimisation happen. It ensures that processor units are maximally used, pipeline stalls are maximally avoided. While optimisation level

0 only remove used code. It also focuses on improving debugging and compiling ease, which may result in worse code speed.

Algorithms with linear phase have the advantage that less multiply accumulation is required. In the for-loop, it only requires to loop through half of the input array, which significantly reduce the amount of calculation for processor.

Double array algorithm is efficient in the sense that it reduces removes some if-statement inside for-loop. Hence, cycle count in each loop is reduced, resulting in less total time necessary for execution.

Notice that TI processor is highly optimised for MAC process. However, if the MAC part of user code is not written in a format recognised by compiler. Code turned out to be very slow since optimisation no longer performs. We have experience that a 'smart' algorithm turned out to be much slower than a normal algorithm.

3.4 final Improvement

```
#include "fir_coef.txt" //include filter coefficients
#define N sizeof(coefs)/sizeof(double)     //filter coefficient number
#define k N%2//define integer k

short x[2*N]={0}; //initialise array to zero
short* ptr1=&x[0]; //initilise the pointer to first element
short* ptr2=&x[N];

void ISR_AIC()
{
      int i;
      double y=0;
      short* p_f=ptr2;
      short* p_l=ptr1;
    double* p_fir=&coefs[0];
      *ptr1=mono_read_16Bit();
      *ptr2=*ptr1;
      p_f=ptr2;
      p_l++;
      if(ptr1==&x[N-1]){
           ptr1=&x[0];
           ptr1--;
      }
      if(ptr2==&x[2*N-1]){
             ptr2=&x[N];
             ptr2--;
      }
      for(i=0;i<(N-k)/2;i++){
            y+=*p_fir*(*p_f+*p_l);
            p_fir++;
```

```
            p_f--;
            p_l++;
            if(i==N/2){ //detect middle term
            y-=k*x[N/2]*coefs[N/2]; //subtract repeated term
            }
      }
    ptr1++;
      ptr2++;
      mono_write_16Bit((short)y);
}
```

Since our filter coefficient number is even, our initial design is for even number coefficient filters. Here is the improved version, which also works for odd number.

An integer *k* is defined, either 0 or 1 depending on whether N is even or odd.

```
#define k N%2
```

If N is odd, every time when looping through the array, the middle term is added up twice since two pointers *p_f* and *p_l* meet in the middle. So the product of middle term is then subtracted from y to product a correct value. Other operations are similar to the version for even number.

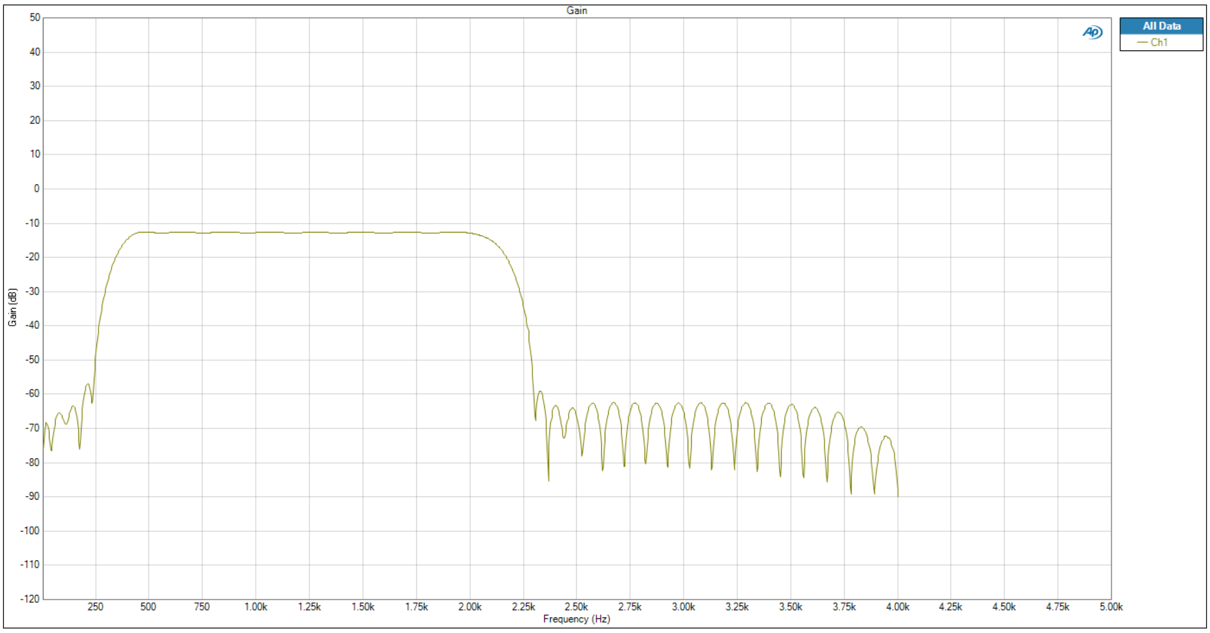
Another improvement of code is about defining number of filter coefficients. The following code allows software to determine N instead of manually enter the number N.

```
#define N sizeof(coefs)/sizeof(double)     //filter coefficient number
```

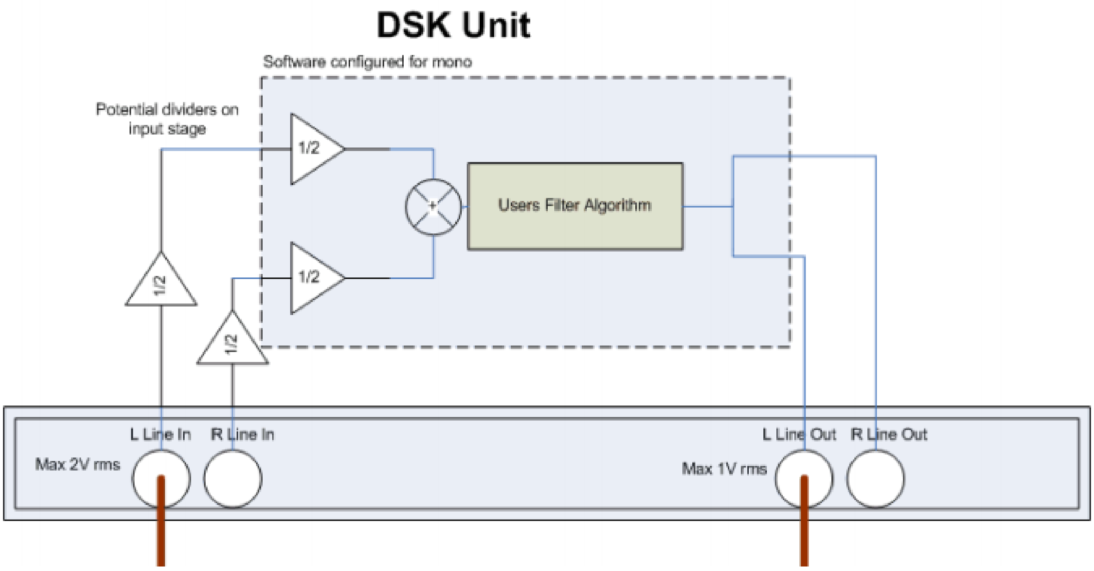
The performance of this code turned out to be same as the previous version. Extra code to detect even or odd number does not add to clock count.

Filter analysis

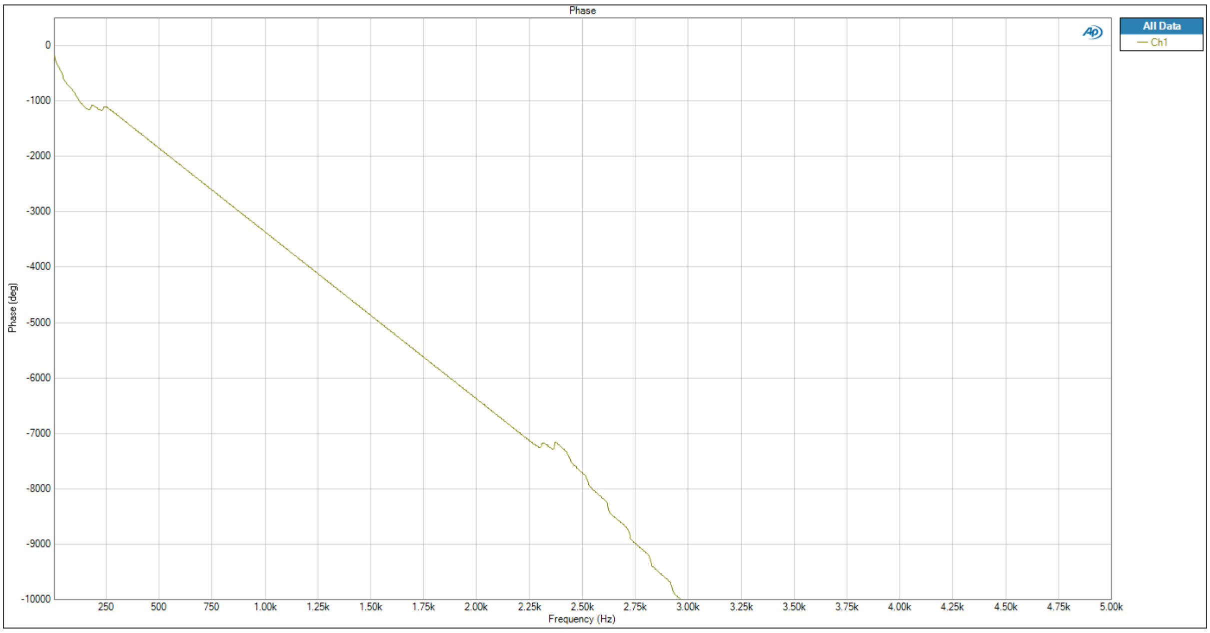
Our fastest implementation of FIR filter is used in this section to investigate frequency and phase response. Frequency response is shown below:



From the frequency response, most specifications of required filter are satisfied, including boundary frequency, stop-band attenuation, and pass-band ripple. However, the pass-band gain is expected to be 0 dB. This is due to the structure of audio analyser, shown below.

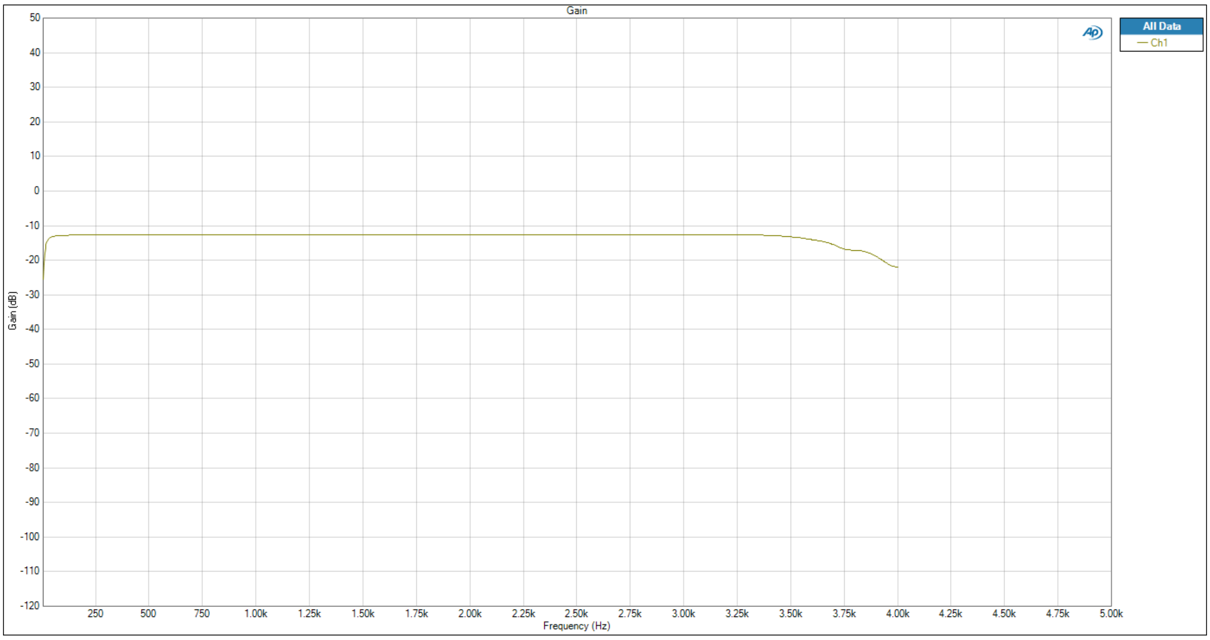


There are two amplifier with gain one half. In decibel, 20log$^{-0.25}$= -12, which agrees with our observation. So that the actual pass-band gain of our filter is around 1.



Phase response is plotted. In the frequency range between 240 Hz and 2.25 Hz, a straight downward sloping line is observed. This is the evidence of linear phase in pass-band.

```
void ISR_AIC()
{
	mono_write_16Bit(mono_read_16Bit());
}
```



At the highest and lowest frequency, observed filter response is different with our expectation. Gain is lower than the theoretical value produced in MATLAB. Therefore, we generate a frequency response graph when sample is directly read to output without filtering. It turned out that the frequency response is not a horizontal line at high and low frequencies. It explains our observation, and our filter behaviour is close to theory.

Appendix

MATLAB

```
close all;
clear all;

f1=240;f2=440;f3=2000;f4=2300;% frequency boundary
f=[f1,f2,f3,f4];%make array of frequencies

rp=0.4;% ripple in dB
sa=48;% minimum stop band attenuation in dB
dev1=(10^(rp/20)-1)/(10^(rp/20)+1);%calculate pass band deviation
dev2=10^(-sa/20);%calculate stop band deviation
dev=[dev2,dev1,dev2];%make array of deviations

Fs=8000;%specify sampling rate
a=[0,1,0];%specify amplitude

[N,Fo,Ao,W] = firpmord(f,a,dev,Fs);%function to approximate filter
coefs = firpm(N+2,Fo,Ao,W);%function to calculate frequency coefficients.

freqz(coefs,1,1024,8000)%Plot frequency and phase response
H = tf(coefs,1);%derive transfer function
figure;
pzmap(H);%plot pole and zero map of filter
grid on;

%The following code store filter coefficients in format readable for c.
fileID = fopen('fir_coef.txt','w');
fprintf(fileID,'double coefs[]={');
for i = 1:length(coefs)
    fprintf(fileID,'%f,',coefs(1,i));
end
fprintf(fileID,'};');
fclose(fileID);
figure;
stem (coefs);
title ('Value of Cefficient');
ylabel ('Amplitue');
xlabel ('Index');
grid on;
```

Non-circular

```
/***************************************************************************
*************
                         DEPARTMENT OF ELECTRICAL AND ELECTRONIC
ENGINEERING
                                               IMPERIAL COLLEGE LONDON

                              EE 3.19: Real Time Digital Signal
Processing
```

```
                                       Dr Paul Mitcheson and Daniel Harvey

                                              LAB 3: Interrupt I/O

                                    ********* I N T I O. C **********

  Demonstrates inputing and outputing data from the DSK's audio port
using interrupts.


*************************************************************************
************
                        Updated for use on 6713 DSK by Danny Harvey: May-
Aug 2006
                        Updated for CCS V4 Sept 10

*************************************************************************
***********/
/*
 *    You should modify the code so that interrupts are used to service
the
 *  audio port.
 */
/**************************** Pre-processor statements
******************************/

#include <stdlib.h>
#include <stdio.h>
//  Included so program can make use of DSP/BIOS configuration tool.
#include "dsp_bios_cfg.h"

/* The file dsk6713.h must be included in every program that uses the
BSL.  This
   example also includes dsk6713_aic23.h because it uses the
   AIC23 codec module (audio interface). */
#include "dsk6713.h"
#include "dsk6713_aic23.h"

// math library (trig functions)
#include <math.h>

// Some functions to help with writing/reading the audio ports when using
interrupts.
#include <helper_functions_ISR.h>

//include fir coefficients

/******************************** Global declarations
********************************/

/* Audio port configuration settings: these values set registers in the
AIC23 audio
   interface to configure it. See TI doc SLWS106D 3-3 to 3-10 for more
info. */
DSK6713_AIC23_Config Config = { \

/**********************************************************************/
                   /*   REGISTER               FUNCTION
SETTINGS         */

/**********************************************************************/\
```

```
    0x0017,  /* 0 LEFTINVOL  Left line input channel volume  0dB
*/\
    0x0017,  /* 1 RIGHTINVOL Right line input channel volume 0dB
*/\
    0x01f9,  /* 2 LEFTHPVOL  Left channel headphone volume   0dB
*/\
    0x01f9,  /* 3 RIGHTHPVOL Right channel headphone volume  0dB
*/\
    0x0011,  /* 4 ANAPATH    Analog audio path control       DAC on, Mic
boost 20dB*/\
    0x0000,  /* 5 DIGPATH    Digital audio path control      All Filters
off        */\
    0x0000,  /* 6 DPOWERDOWN Power down control              All Hardware
on        */\
    0x0043,  /* 7 DIGIF      Digital audio interface format  16 bit
*/\
    0x008d,  /* 8 SAMPLERATE Sample rate control             8 KHZ
*/\
    0x0001   /* 9 DIGACT     Digital interface activation    On
*/\

/************************************************************************/
};


// Codec handle:- a variable used to identify audio interface
DSK6713_AIC23_CodecHandle H_Codec;

#define N 78
#include "fir_coef.txt"

short x[N]={0};

 /******************************** Function prototypes
********************************/
void init_hardware(void);
void init_HWI(void);
void ISR_AIC(void);
void non_circ_FIR();
/********************************** Main routine
************************************/
void main(){

      // initialize board and the audio port
  init_hardware();

  /* initialize hardware interrupts */
  init_HWI();

  /* loop indefinitely, waiting for interrupts */

  while(1)
  {};

}

/********************************** init_hardware()
**********************************/
void init_hardware()
{
    // Initialize the board support library, must be called first
```

```
    DSK6713_init();

    // Start the AIC23 codec using the settings defined above in config
    H_Codec = DSK6713_AIC23_openCodec(0, &Config);

      /* Function below sets the number of bits in word used by MSBSP
(serial port) for
      receives from AIC23 (audio port). We are using a 32 bit packet
containing two
      16 bit numbers hence 32BIT is set for  receive */
      MCBSP_FSETS(RCR1, RWDLEN1, 32BIT);

      /* Configures interrupt to activate on each consecutive available
32 bits
      from Audio port hence an interrupt is generated for each L & R
sample pair */
      MCBSP_FSETS(SPCR1, RINTM, FRM);

      /* These commands do the same thing as above but applied to data
transfers to
      the audio port */
      MCBSP_FSETS(XCR1, XWDLEN1, 32BIT);
      MCBSP_FSETS(SPCR1, XINTM, FRM);


}

/********************************** init_HWI()
***************************************/
void init_HWI(void)
{
      IRQ_globalDisable();                  // Globally disables interrupts
      IRQ_nmiEnable();                      // Enables the NMI interrupt
(used by the debugger)
      IRQ_map(IRQ_EVT_RINT1,4);             // Maps an event to a physical
interrupt
      IRQ_enable(IRQ_EVT_RINT1);            // Enables the event
      IRQ_globalEnable();                         // Globally enables
interrupts

}

/******************** WRITE YOUR INTERRUPT SERVICE ROUTINE
HERE***********************/

void ISR_AIC()
{
      int i;
      for(i=N-1;i>0;i--){             //loop through every element from
last to first
            x[i]=x[i-1];              //shift all element to right and
abandon the last one
      }
      x[0]=mono_read_16Bit(); //store input
      non_circ_FIR();
}

void non_circ_FIR()
{
      int i;
      double y=0;
```

```
	for(i=0;i<N;i++){	//loop through the array and perform convolution
		y+=x[i]*coefs[i];
	}
	mono_write_16Bit((short)y);//write to output port
}
```

Circular

```
void ISR_AIC()
{
	circ();
}

void circ(){
	int i;
	double y=0;
	short* ptr_loop;	//pointer to loop through input array
	double* ptr_coef=&coefs[0];//pointer to filter coefficient array

	*ptr=mono_read_16Bit(); //read input
	ptr_loop=ptr;		//initilise loop pointer to latest reading

	for(i=0;i<N;i++){
		y+=*ptr_loop*(*ptr_coef);//perform convolution
		if(ptr_loop==&x[0]){
			ptr_loop=&x[N-1]; //loop back to beginning if loop pointer reaches end
			ptr_loop++;	//adjustment to make sure pointer at 0 position at
		}			//the end of for loop
		ptr_coef++;		//increment filter pointer
		ptr_loop--;		//decrement loop pointer
	}

	if(ptr==&x[N-1]){ //pointer to store input loop back to beginning
		ptr=&x[0];  //after reaching the end of the array
		ptr--;
	}
	ptr++;		//increment pointer

	mono_write_16Bit((short)y);
}
void linear phase(){
	//declare parameters initially, including type and name.
	int i;
	// we set y to double which avoid overflow.
	double y=0;
	//p_f pointed to newest number, p_l pointed to oldest number.
	short* p_f=ptr;
	short* p_l=ptr;
	double* p_fir=&coefs[0];
	//get readings from input port.
	*ptr=mono_read_16Bit();
```

```
        //increasing p_l by 1 to make sure it points to the oldest number.
        p_l++;
        //Since the buffer size is fixed, we need to reset the pointer
index
        //when it reaches boundary.
         if(ptr==&x[N-1]){
              p_l=&x[0];
              ptr=&x[0];
              ptr--;
        }
        //we use a for loop to perform convolution function.
        for(i=0;i<N/2;i++){
              y+=*p_fir*(*p_f+*p_l);
              //avoding overflow, we need to reset p_f once it reaches
              //boudary.
              if(p_f==&x[0]){
                    p_f=&x[N-1];
                    p_f++;
              }
              //avoding overflow, we need to reset p_l once it reaches
              //boudary.
              if(p_l==&x[N-1]){
                    p_l=&x[0];
                    p_l--;
              }
              //changing coefficient to meet our function
              p_fir++;
              p_f--;
              p_l++;
        }
      //increase pointer by 1. get ready for next interrupt
        ptr++;
        //write down the result to output port.
        //since it is 16bits, we need to change the type from double to
        //short.
        mono_write_16Bit((short)y);
}

void Double memory(){
        //declare parameters initially, including type and name.
        int i;
        // we set y to double which avoid overflow.
        double y=0;
        //p_f pointed to newest number, p_l pointed to oldest number.
        short* p_f=ptr2;
        short* p_l=ptr1;
        double* p_fir=&coefs[0];
        //get readings from input port.
        *ptr1=mono_read_16Bit();
        //copying data to fulfil the buffer.
        *ptr2=*ptr1;
        //increasing p_l by 1 to make sure it points to the oldest number.
        p_l++;
        //Although the buffer size is doubled, we still need to reset the
pointer index
        //when it reaches boundary.
        if(ptr1==&x[N-1]){
             ptr1=&x[0];
             ptr1--;
        }
        if(ptr2==&x[2*N-1]){
```

```
            ptr2=&x[N];
            ptr2++;
      }
      //using the property of linear phase, we match the inputs with
      //the same coefficient together.
      for(i=0;i<N/2;i++){
            y+=*p_fir*(*p_f+*p_l);
            p_fir++;
            p_f--;
            p_l++;
      }
      //increment pointer avoiding over-written.
      ptr1++;
      ptr2++;
      mono_write_16Bit((short)y);
}
```

Final Version

```
/*************************************************************************
*************
                        DEPARTMENT OF ELECTRICAL AND ELECTRONIC
ENGINEERING
                                              IMPERIAL COLLEGE LONDON

                             EE 3.19: Real Time Digital Signal
Processing
                                    Dr Paul Mitcheson and Daniel Harvey

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/**********************************************************************/
};


// Codec handle:- a variable used to identify audio interface
DSK6713_AIC23_CodecHandle H_Codec;

#include "fir_coef.txt" //include filter coefficients
#define N sizeof(coefs)/sizeof(double)    //filter coefficient number
```

```
#define k N%2

short x[2*N]={0}; //initialise array to zero
short* ptr1=&x[0]; //initilise the pointer to first element
short* ptr2=&x[N];

int ptr1_index=0;


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void init_HWI(void);
void ISR_AIC(void);

/********************************** Main routine
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{
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      double y=0;
      short* p_f=ptr2;
      short* p_l=ptr1;
    double* p_fir=&coefs[0];
      *ptr1=mono_read_16Bit();
      *ptr2=*ptr1;
      p_f=ptr2;
      p_l++;
      if(ptr1==&x[N-1]){
           ptr1=&x[0];
           ptr1--;
      }
      if(ptr2==&x[2*N-1]){
             ptr2=&x[N];
             ptr2--;
      }
      for(i=0;i<(N-k)/2;i++){
            y+=*p_fir*(*p_f+*p_l);
            p_fir++;
            p_f--;
            p_l++;
            if(i==N/2){
            y-=k*x[N/2]*coefs[N/2];
            }
      }
    ptr1++;
      ptr2++;
      mono_write_16Bit((short)y);
}
```