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RTDSP Lab 4 Report

*Real-Time Implementation of FIR Filters*

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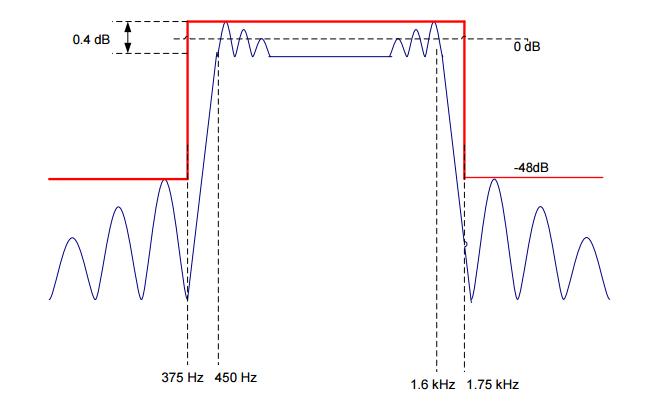
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# Matlab Filter Design

An FIR filter must be designed using MATLAB. The filter has the following specifications:

* Stop band attenuation of -48dB
* Maximum passband ripple of 0.4dB
* Passband frequencies between 450 and 1750Hz

*Figure 1* below highlights the full specifications for the pass band filter.

.

*Figure 1 - The full specifications for the pass band filter [1]*

To implement this filter, the Parks-MClellen algorithms is used. This algorithm uses iterative techniques (Remez algorithm) to design the filter to the specifications. Using a simple rectangular filter will produce large errors at the discontinuities due to Gibbs phenomenon. The Parks-MClellen algorithm has the advantage that the errors between the ideal and actual magnitude response can be controlled and minimised in each of the frequency bands.

The algorithm is implemented in MATLAB via the firpmord and firpm functions. firpmord takes parameters that describe the desired cutoff frequencies and amplitudes of the stopbands/passbands of the filter, the allowed deviation from the desired amplitudes, and the sampling frequency. It returns parameters that can be passed to firpm to produce an array of frequency coefficients.

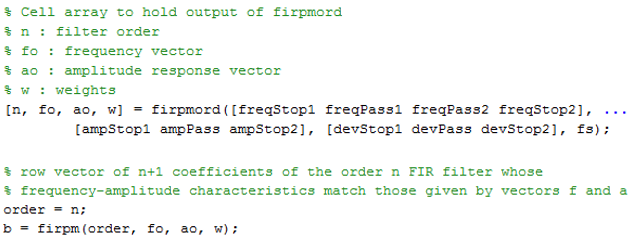
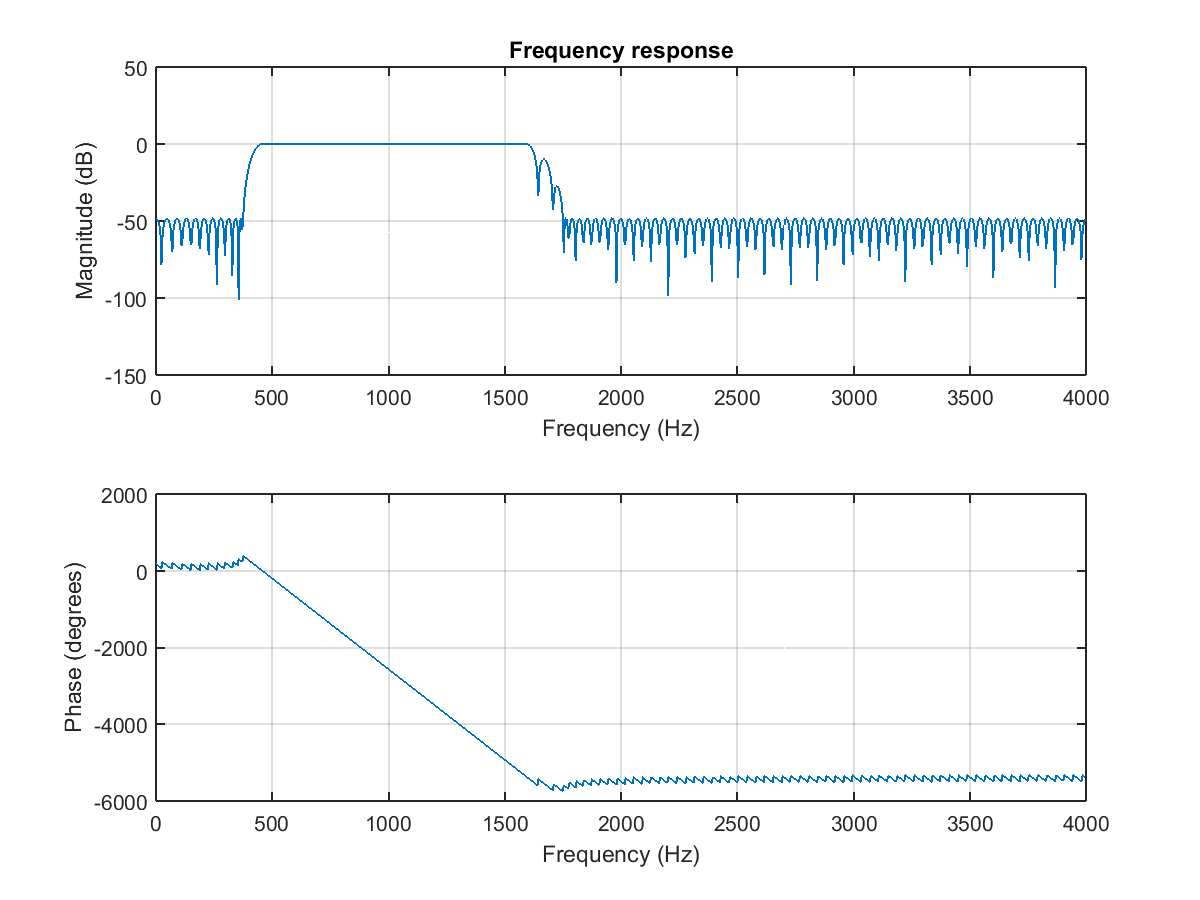


Figure 2 - The firpmord function passes the passband specification as arguments to firpm, which returns an array of frequency coefficients, b.

Our first attempt used the given specifications for the ideal magnitude response of the filter as the parameters to firpmord and produced the magnitude and phase response shown in *Figure 3*. The filter order, however, needed to be increased to n+4 due to this filter giving insufficient attenuation on the stop band (-47.54dB). The results running the MATLAB script gives the specifications found on Table 1. For purposes that will be clear later, the filter with these specifications will be called FIR1.

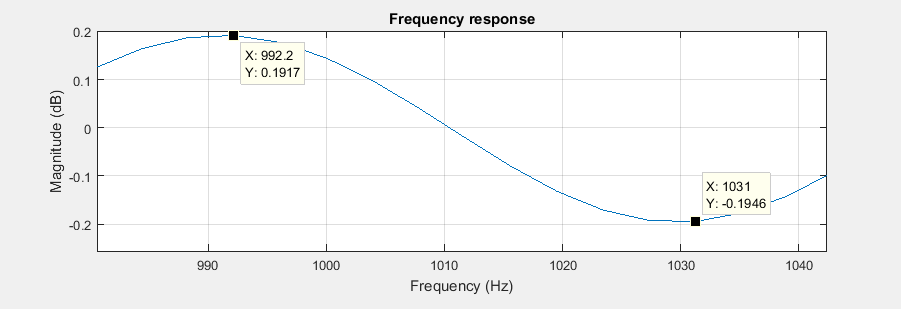
|  |  |
| --- | --- |
| FreqStop1 | 375Hz |
| FreqPass1 | 450Hz |
| freqPass2 | 1600Hz |
| freqStop2 | 1750Hz |
| devStop1 | 0.00251 |
| devPass | 0.02302 |
| devStop2 | 0.00251 |
| order | n+4 |
| No. of coefficients | 211 |

Table 1 First FIR filter specifications (FIR1)



*Figure 3 Magnitude and phase responses of the filter using the given specifications*

The new filter now gives good attenuation (-48.27dB) and pass band ripple (found to be 0.386dB) as shown in *Figure 4*.



*Figure 4 Pass band ripple*

# Non-Circular FIR Filter

An N-tap FIR filter algorithm must be designed. The output is given by:

The Z-transform can be found:

Giving



*Figure 5 Direct form of an FIR filter*

Essentially the Direct Form as shown in *Figure 5* must be implemented. To implement the delay line, an array buffer will be used. A new sample is always added to the start of this delay line and then shifted across to further down the array to implement this delay. To implement the filter, a multiply-accumulate function is used.

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
| New sample x(n) | 0 | 0 | 0 | 0 |

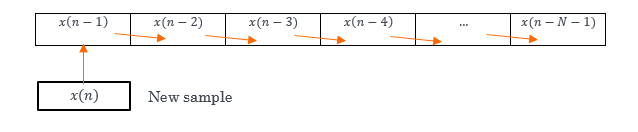
Figure 6 Initial empty buffer array

|  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- |
|  |  |  |  |  |  |

*Figure 7 Example array buffer with sampled components x(n)*

|  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- |
|  |  |  |  |  |  |

*Figure 8 Example array containing filter coefficients*

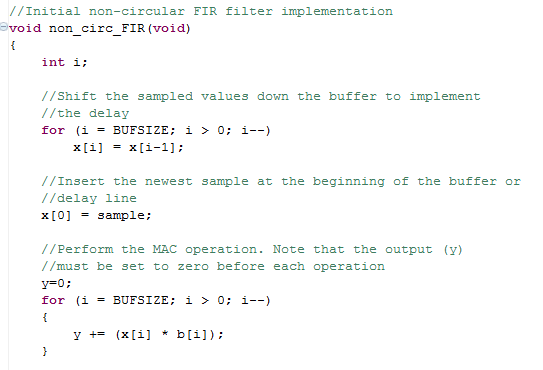
Each of the filter components are multiplied with each of the array components and then summed to give an output for the certain sample. An example of this operation is given in *Figure 9* and *Figure 10*

*Figure 9 An explanation of the operation when taking a new sample*

|  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- |
| *\** | \* | \* | \* | \* | \* |
|  |  |  |  |  |  |

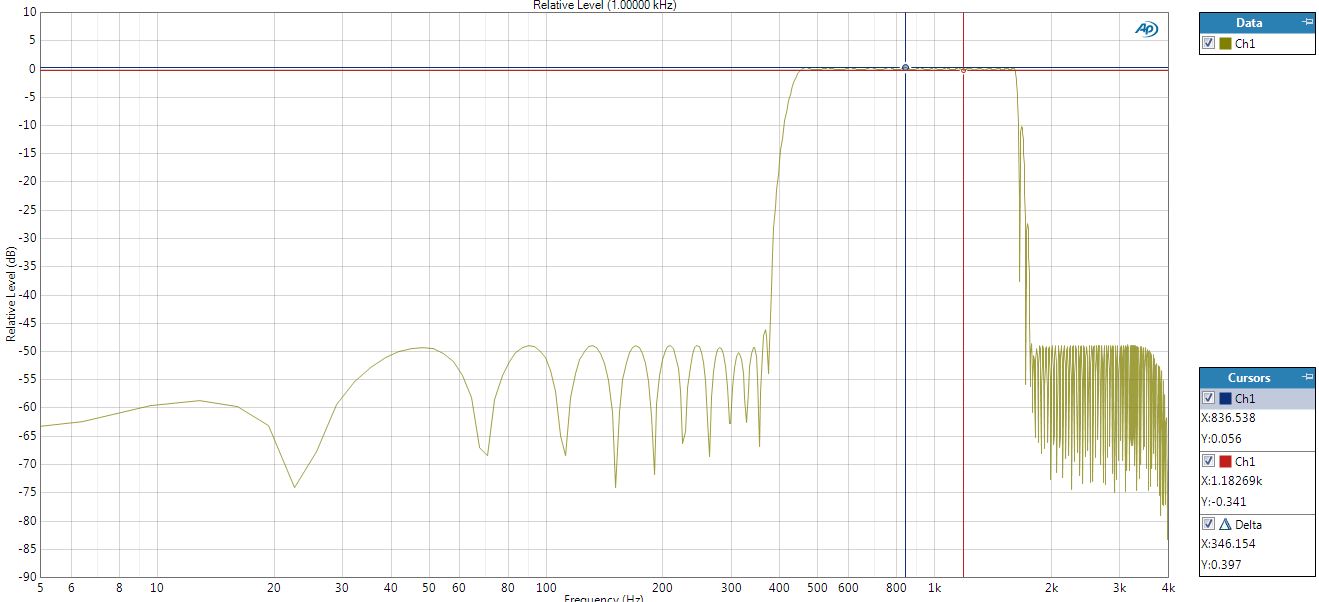
*Figure 10 The multiplications are then summed*

After the operation in *Figure 10* is performed the, values are then summed to give an output for the new sample. This operation that is being performed is:

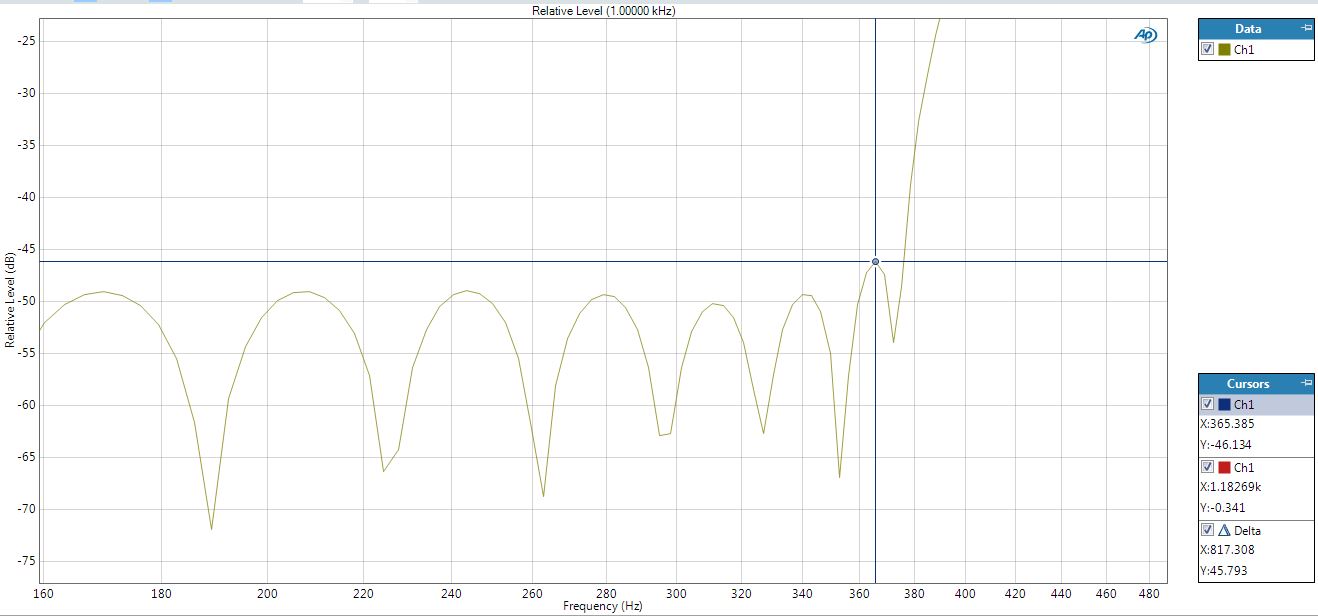


*Figure 11 The non-circular FIR filter implementation*

A frequency sweep can be performed using the APX520 Audio Analyser giving the actual frequency response of the filter. From the trace given in *Figure 12* it can be seen that the response closely matches the MATLAB trace for the magnitude response. However, it is found that the parameters differ slightly from the specification, as with the filter that was previously being produced, around the stop band edge of the first transition band, one of the lobes was not matching the specification according to the network analyser, even though it was found to be correct on MATLAB. This may be due to delay in the ADC/DAC of the DSP board. **Error! Reference source not found.** highlights this stop band attenuation (-45.79dB).

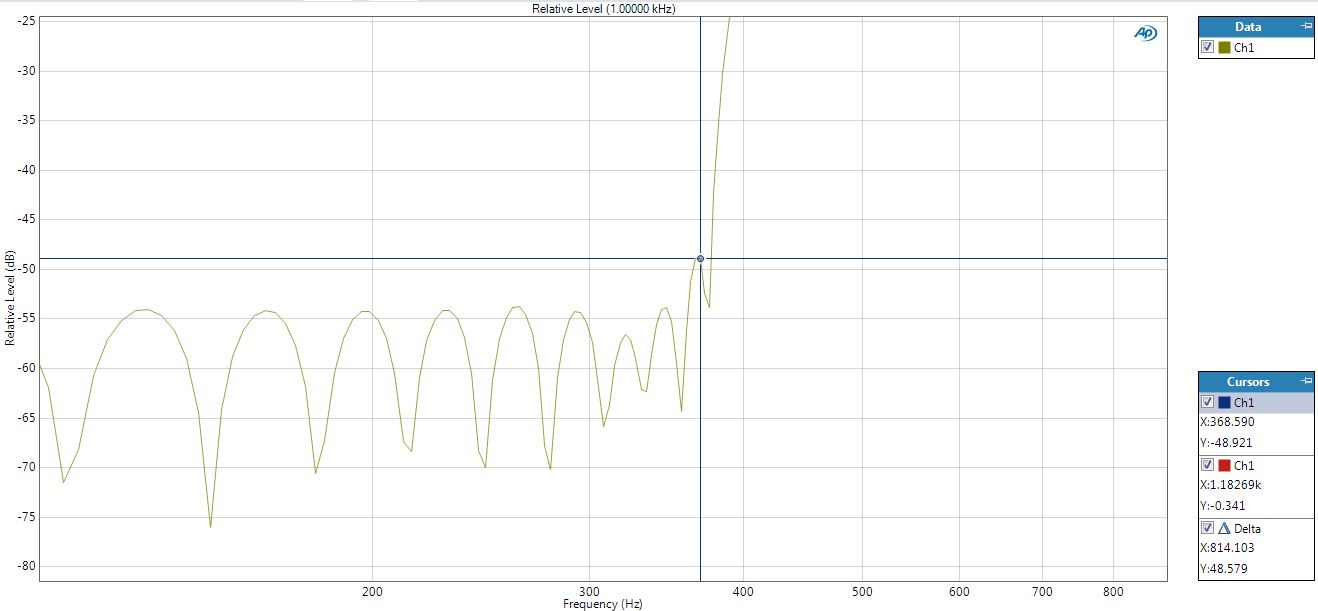


*Figure 12 Magnitude response of the initial filter*



*Figure 13 Stop band attenuation of the initial filter*

To correct for this, a new filter must be designed through MATLAB. As the process for designing the filter using the Parks-MClellen algorithm is complex, simple trial and error is used to achieve a more correct result. The new filter’s frequency response can be found on ---. The improved stop band attenuation of this new filter can be seen in *Figure 14*.



*Figure 14 Improved stop band attenuation*

The new filter now called FIR2 has the specifications summarized in Table 2. However, since the number of coefficients has now significantly increased, the performance of any FIR filter function will be hindered. Taking to consideration that this error is relatively small, to improve performance the filter FIR1 is used.

|  |  |
| --- | --- |
| freqStop1 | 375Hz |
| freqPass1 | 450Hz |
| freqPass2 | 1600Hz |
| freqStop2 | 1700Hz |
| devStop1 | 0.00251 |
| devPass | 0.02072 |
| devStop2 | 0.00251 |
| order | n+8 |
| No. of coefficients | 231 |

Table 2 Specifications for improved filter (FIR2)

The phase response of this filter given in Figure 15, shows a more significant phase shift than is expected (using the MATLAB results). The response seen in --- shows this phase shift. The reason is due to the output filter of the AIC23 Audio Chip (*Figure 16*) gives this additional phase shift. Running the program without using a filter and finding the phase response using the spectrum analyser gives the trace found in Figure 16.

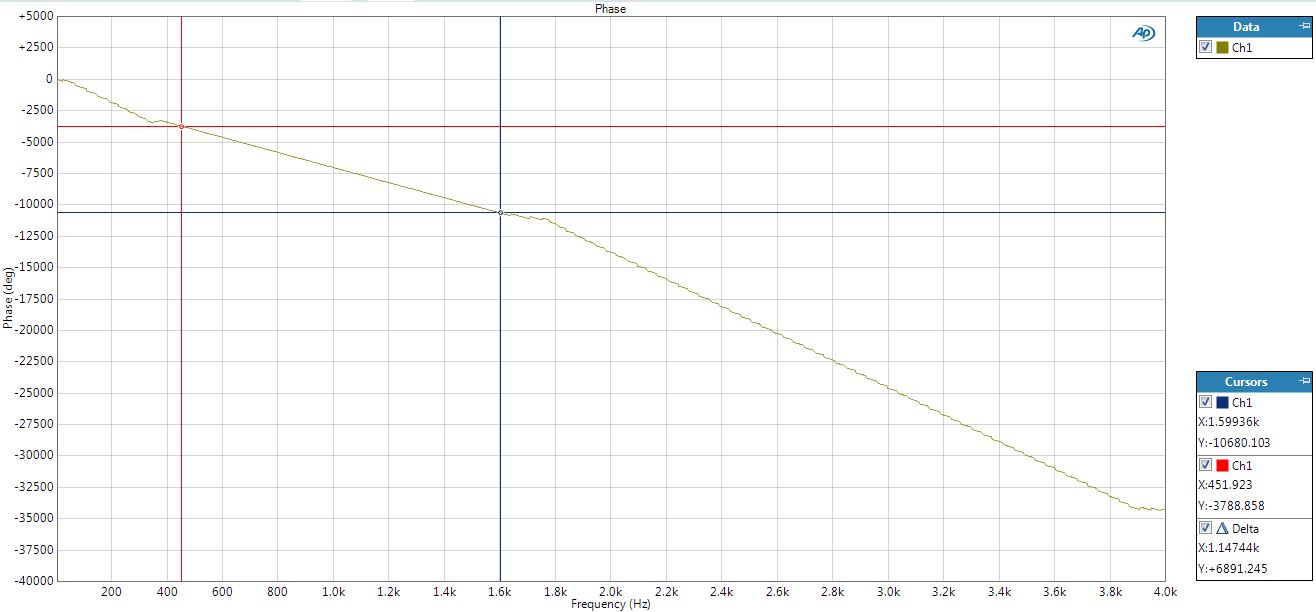


Figure 15 Phase response of the FIR filter

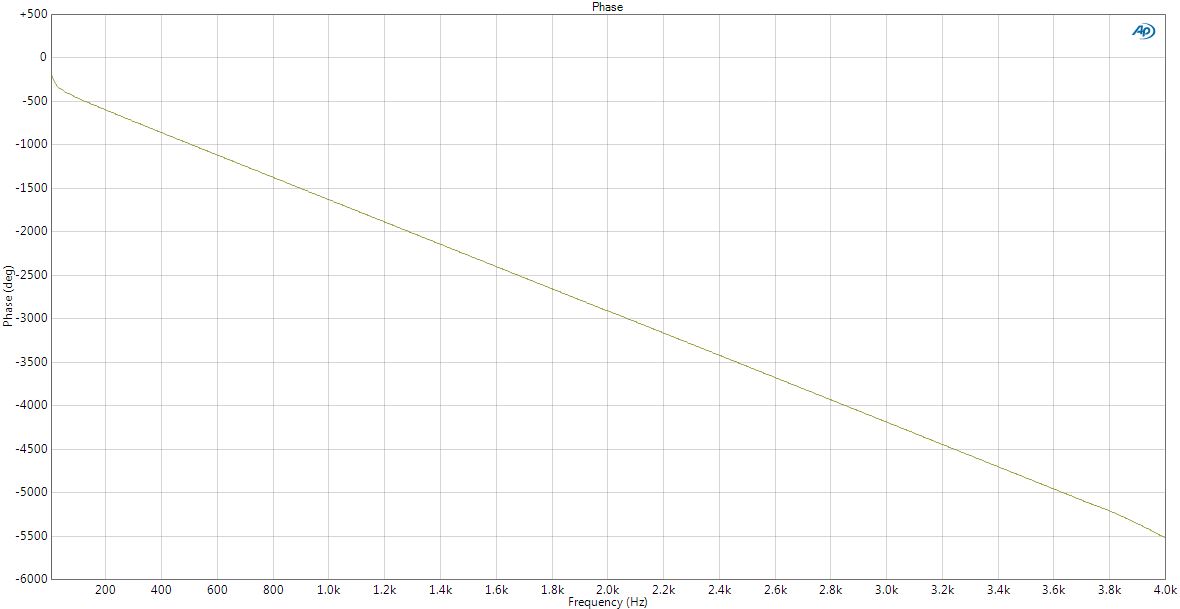


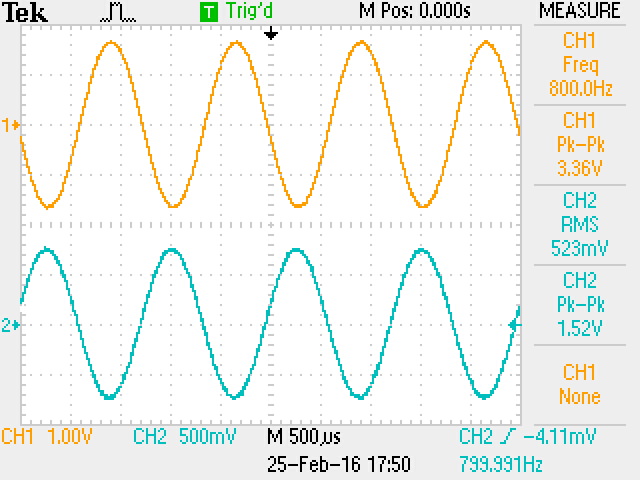
Figure 16 Phase response without using an FIR filter

Another effect that can be seen is the roughly 12dB attenuation shown through the actual magnitude response given by the spectrum analyser. This attenuation occurs due to two reasons:

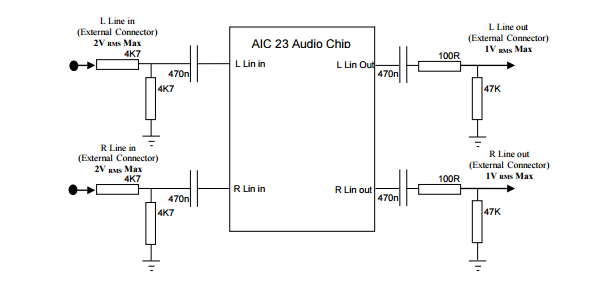
* The input potential divider of the AIC23 Audio Chip (*Figure 16*).
* The left and right line-in’s are averaged (only one line-in used in this case)

The output is therefore of the input, giving the 12dB attenuation. For simplicity, the spectrum analyser traces are shown with the gain at a relative level to help with the validation of the response (checking to see if the response meets the specification). A magnitude response trace showing this attenuation is given in ---.

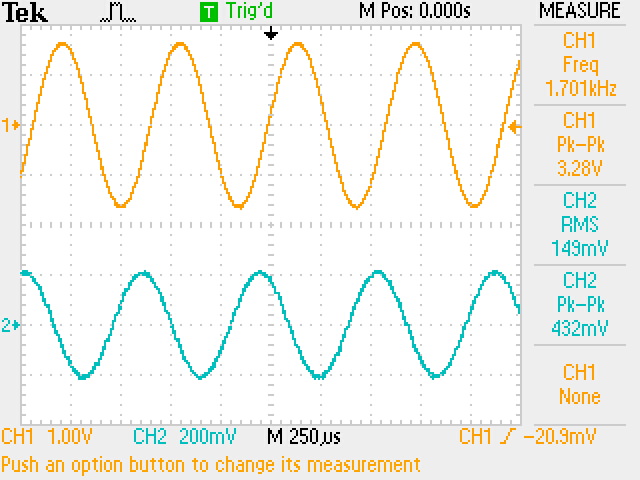
Using the filter and the non-circular implementation of the function, the output of the board can be seen in the scope trace. The input frequency used is 800Hz, which is well within the passband. As can be seen from *Figure 15* the output is given is as expected. There is a significant phase shift and a magnitude attenuation as expected. As can be seen, the output is roughly a quarter of the input.



*Figure 17 Scope trace for an input frequency of 800Hz (Input = Yellow)*



*Figure 18 AIC23 Audio chip [AIC23 Audio chip external components adapted from TMS320C6713 Technical ref (page A-14, 2003 revision A]*



*Figure 19 Scope trace in the transition band at 1700Hz giving a clear reduced amplitude (Output = Blue)*

---- and ---- show the outputs clearly in the transition and stop bands. With ---- the amplitude is clearly reduced as expected within the transition. With ----- the input frequency 2200Hz deep within the stop band, there is very large attenuation giving a very small amplitude noisy output.

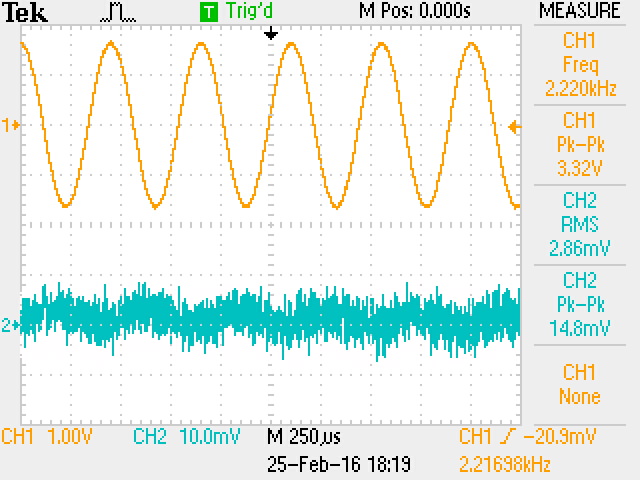


Figure 20 Scope trace with an input frequency deep within the stop band giving a noisy very low amplitude output (Output = blue)

The performance of this non-circular implementation of the FIR filter can be now found. The function is tested at different compiler optimization levels. An explanation of the different optimization levels can be summarised in Table 3.

|  |  |
| --- | --- |
| Compiler optimisation level | Effect |
| 0 | This is the lowest optimization level. Code that is unused is eliminated and allocates all variables to registers. Expressions and statements are also simplified. Using this level can sometimes even have a detrimental effect on the number of clock cycles used |
| 1 | All effects of optimisation level 0 and also eliminates local common expressions. Variables are checked to see if they can be turned into constants by performing local copy or constant propagation. |
| 2 | All optimisation level 1 effects are used. At this level the compiler also attempts to generate software pipeline loops. Loops are unrolled and array references in theses loops are converted to pointer form drastically decreasing the number of cycles used in loop and array intensive functions |

Table 3 Compiler optimization level effects [TMS320C6000 Optimizing Compiler v7.4 User's Guide]

It can be expected that the non-circular implementation takes a significant number of cycles for each of the optimization levels. This implementation is slow since there are two for loops with a running length of the size of the buffer array with the shifts at every time step (sample). The first for loop is used to implement the delay by shifting every element in the array once and inserting the newest sample at the zeroth element of the array. The second for loop performs the MAC function to give an output for each sample by multiplying every element in the buffer with the filter coefficients. To improve performance a circular-buffer filter (see next section) can be used. Table 4 highlights the number of cycles used for each performance level. It can be seen that the number of cycles significantly decreases between the zeroth and second optimization level.

|  |  |
| --- | --- |
| **Optimisation\algorithm** | **Non\_circ** |
| None | 13409 |
| O0 | 11286 |
| O2 | 973 |

# Circular-Buffer Filters

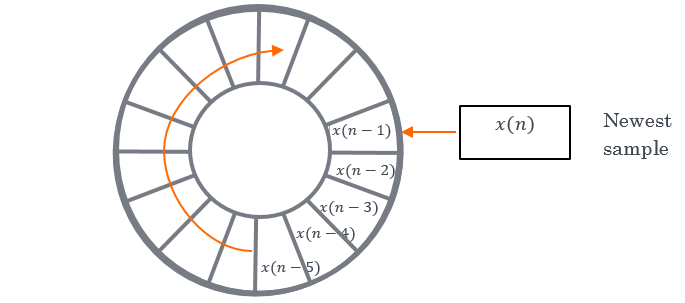
To improve performance a circular buffer can be used. Unlike the previous case the samples are now stored in a circular buffer.

Newest sample

|  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- |
|  |  |  |  |  |  |

Samples get older to the right

It is better to think of the



The operation that is now being performed is

The function for the simple circular buffer can be seen in ----. The explanation of the function is unlike the non-circular algorithm, a circular buffer works by changing which position in the buffer each new piece of data is written to, rather than writing to the first position and shuffling all the data up one space. This means we need a variable which keeps track of which position the newest piece of data was written to – in our program, this is the newest variable. newest is initialised to the last index of the buffer, and is decremented to the previous position each time data is read in, meaning data is stored in descending order of array index. When newest reaches the start of the array, it is explicitly set to the last position again, to prevent underflow.

The reason we need this variable is to keep the data buffer aligned with the filter coefficient buffer, to make sure we are multiplying the correct pairs of values. To achieve this, the convolution loop initialises the data buffer index, j, to the index of the newest piece of data, and the filter buffer index, i, to 0. The convolution continues similarly to the non-circular algorithm, with both indices incrementing, except that since j started mid-way through the array, at some point it will overflow, so an if statement is added to check for this overflow and correct it by returning j to the start of the array.

###########################circ-fir1#######################################

An improved version of circ\_FIR1 can be uses, since – instead of array indexes, pointers are used. To understand why this is more efficient, we must realise that an array index is composed of two parts: a pointer to the start of the array (the name of the array) and an offset from the start of the array (the number in square brackets). When the compiler evaluates an array index, it has to compute the position by adding the offset to the pointer, which costs cycles. What we do, instead, is to post-increment the pointer every time it is used, ensuring that it is already at the correct position when it is needed, which cuts the time that was being used for the index evaluation. Additionally, we make use of the register keyword: memory accesses take a lot of time, and the register keyword forces the variable to be kept in a register so that it is always easily accessible by the processor. This is especially useful when applied to the variable storing the accumulated result of the convolution, as it is accessed every iteration of the loop.

#######################circ-fir5############################################

This function can be improved since the condition in the if statement in the previous algorithm only ever evaluates to true once for the duration of the for loop, however it is run every loop, causing unnecessary extra cycles to be used. We can remove it by pre-calculating the point at which the overflow will occur, and splitting the function into two for loops instead. This way, the same operations are executed, without the overhead of the if statement.

##########################circ-fir2######################################

Another method of removing the if statement is using a data buffer of twice the length, and storing each sample at the index pointed to by newest, but also at the index newest + BUFSIZE. This way, we can start our convolution iteration over BUFSIZE elements without having to worry about overflowing the array, and also being confident that each of the data values is accounted for, as any stretch of BUFSIZE elements starting at any point will contain all the data samples. If we start at newest, then we can be sure the array is aligned properly also.

##########################circ-fir3########################################

Symmetrical FIR Filters

FIR filters usually have the property of linear phase. This property gives a filter response that is linear when taken as a function of frequency. For an FIR filter to be linear phase, it must be symmetrical. A symmetrical filter means that the first and last coefficients are similar i.e.

The proof of the linear phase property can be found through finding the Fourier transform:

Taking

Assuming N is even

Substituting

For

Since is real, the overall phase is , giving linear phase.

Symmetrical circular buffer explanation - i.e symmetrical FIR filter explanation

Since the

A different approach is to reduce the number of operations by taking advantage of the fact that our filter is of linear phase and so its coefficients are symmetrical. Since the first/last coefficients are equal, as well as the second/second last and so on, we only need to perform the half as many multiplications, as we can factorise x[0]\*b[0] + x[1]\*b[1] + ... x[BUFSIZE-2]\*b[BUFSIZE-2] + x[BUFSIZE-1]\*b[BUFZSIZE-1] into b[0]\*(x[0]+x[BUFSIZE-1]) + b[1]\*(x[1]+x[BUFSIZE-2]) + ...

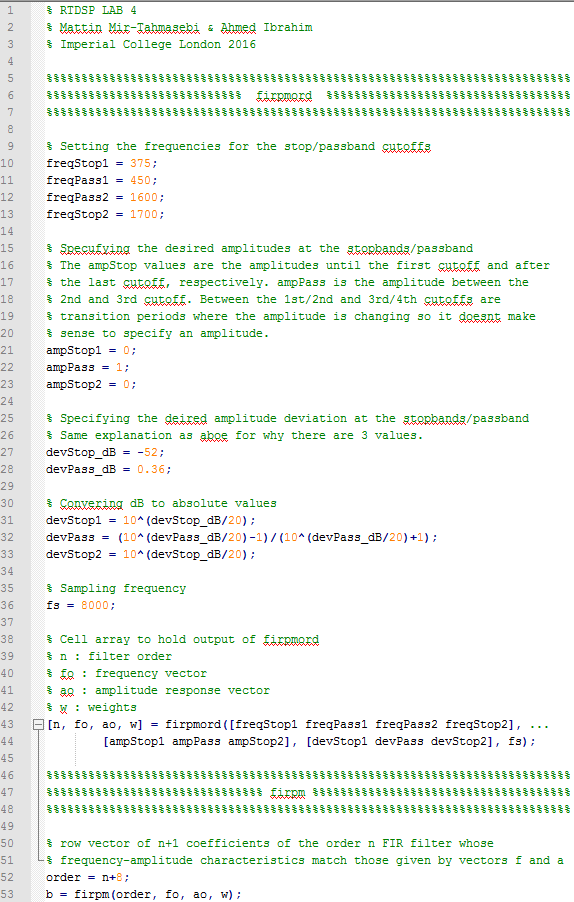
This is achieved by initialising an index, j, to start at the newest data value, and another index, k, to start at the oldest data value. Every iteration, j will increment and k will decrement, and the convolution is carried out as described by the factorisation above. Again there is the issue of under/overflow, which is mitigated by two if statements that work similarly to the one in circ\_FIR1

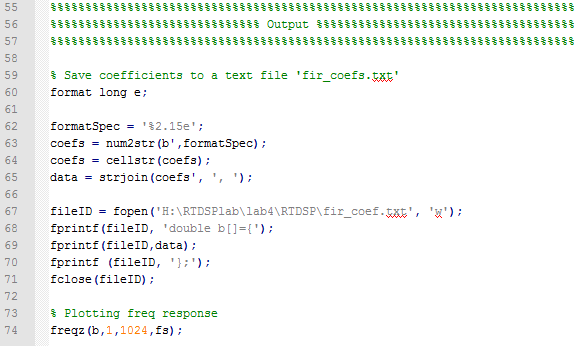
If the number of coefficients is odd, then j and k will not reach the middle element, since the condition of the for loop specifies to break before it is reached. Due to this, we manually add the last value to the result. It would be equally correct to allow the middle element to be added twice and then subtract it once afterwards, but this would require an extra unnecessary calculation, so the former method was chosen.

# Network Analyser Results

Gain is reduced by ~12dB due to : -6 from averaging of left and right channels, -6 from potential divider, -.8 from gain of dsk

### APPENDIX A: filter\_genterator.m





### APPENDIX B: initio.c

|  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- |
| 𝑥𝑛−1  New sample |  |  |  |  |  |

Newest sample

𝑥