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MELBOURNE, VICTORIA

Aircraft Structures Technical Memorandum 555

**DESIGN AND IMPLEMENTATION OF DIGITAL FILTERS
FOR ANALYSIS OF F/A-18 FLIGHT TEST DATA**

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by

W. WALDMAN

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SUMMARY

This technical memorandum describes the methods and computer programs used to specify, design, and implement time-domain and frequency-domain FIR digital filters for use in the analysis of F/A-18 flight test data. Two bandpass filters covering the 10-20 Hz and 32-52 Hz frequency bands have been developed, and they can be used for analysing the two dominant modes of structural vibration response occurring on the F/A-18 empennage. A highpass filter with a cutoff frequency of 8 Hz was also designed for filtering strain gauge data for use in producing fatigue load sequences for coupon testing. The effects of bandpass filtering on the transient response of short-term vibrations lasting about one second have also been investigated, leading to the conclusion that FIR digital filters have a negligible effect on the transient response characteristics of short bursts of vibration.

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1.0 INTRODUCTION

A joint effort between Canada and Australia to perform a full-scale structural fatigue test on the F/A-18 Hornet aircraft is currently underway. This project is called the International Follow On Structural Test Program (IFOSTP), and in 1989 a series of flight trials were performed at the Aerospace Engineering Test Establishment (AETE) in Canada to determine the Hornet vibration response under typical service conditions [1].

The airflow around the F/A-18 aircraft at moderate to high angles of attack is characterized by separated vortices produced by the highly-swept leading edge extensions (LEXes). When these vortices impinge on the horizontal and vertical tails, the buffet pressures produce high levels of structural vibration. In an attempt to reduce the levels of vibration response and the attendant fatigue damage, aerodynamic fences have been fitted to the leading edge extensions of all F/A-18 aircraft. This modification was designed by the manufacturer, McDonnell Douglas Corporation.

The AETE flight trials measured typical sequences of dynamic loads on the vertical and horizontal tails, engine mounts, and other aft fuselage locations. The measurands consisted of accelerometer and strain gauge channels, and flight testing was carried out for ACM (Air Combat Manoeuvre) training, as well as for test points covering a range of AOA (angle of attack) and Q (dynamic pressure) conditions. The flights were conducted both with and without the LEX fence.

The measurements that were collected are to be used to characterize the empennage vibration response environment experienced by the F/A-18. They will be processed to produce reference response spectra at a number of locations, and a suitable vibration loading is to be applied to the fatigue test article to match the in-flight vibration environment as closely as possible. Due to the broad-band nature of the vortex buffet pressure spectrum, there is significant structural response at the resonance frequencies of the vertical and horizontal tails. The 10–20 Hz and 32–52 Hz bands show a significant modal response, and the response in these two bands is commonly referred to as Mode I and Mode II, respectively. There is also a broad-band response in the 80–100 Hz region under some AOA-Q conditions. The root-mean-square (RMS) response in a given band is a function of both AOA and Q, and the vibration levels of one mode relative to another also vary significantly.

In order to be able to characterize the vibration response of the F/A-18, it is necessary to be able to determine the modal response in each of the important frequency bands. This calls for bandpass filtering of the data and, because it is stored in sampled serial digital form, the use of digital bandpass filters to perform the required signal processing is an efficient and convenient proposition. McDonnell Douglas performed flight tests during the development of the F/A-18 aircraft, and their data analysis relied heavily on the use of bandpass filtering to determine the RMS response in the two main frequency regions of modal vibration. By filtering the AETE data into similar frequency bands, it will be possible to compare our results with existing data. Highpass filtering of strain gauge data is also required in order to leave only the dynamic response, which will later be combined with typical manoeuvre loads to produce fatigue load sequences for coupon testing.

2.0 DESIGN OF BANDPASS AND HIGHPASS DIGITAL FILTERS

2.1 Magnitude and phase response considerations

Several parameters are used to describe a filter's performance, and one of the most important of these is the magnitude response as a function of frequency. A filter's passband is the frequency region where the signal undergoes little or no modification, while the stopband is where the signal is attenuated to be less than some specified level. The transition band, as its name implies,

is the region between the passband and the stopband, and the signal in this region is attenuated to intermediate degree. When selecting a suitable filter, it is necessary to consider the flatness of response in the filter's passband, and limits on this are defined by how much variation or ripple we are prepared to tolerate. Another important parameter is the rate of attenuation in the transition band, which will determine how quickly a given amount of attenuation will be achieved. For many types of filters it is also necessary to consider the level of the ripple in the stopband.

Evaluating the transfer function of a filter results in both a magnitude and a phase characteristic. Most filters have a non-linear phase shift characteristic, which causes each significant frequency component in the signal to undergo a different delay (non-linear group delay). This behaviour introduces distortion of the output waveform in addition to the simple removal of unwanted frequency components. As a result, the peaks in the output waveform may be different in the presence of delay distortion than without it, and this can affect the results of peak-valley processing as commonly carried out in fatigue damage studies.

2.2 FIR digital filters

In terms of magnitude and phase response, it is possible to design digital filters that possess the same characteristics as classical analogue filter designs. For data that have already been measured and are stored on computer disk, it is possible to completely eliminate the non-linear phase behaviour by processing the data in both the forward and reverse directions. This procedure exactly cancels the phase shifts, and also results in a squaring of the magnitude response function.

There exists a class of digital filters that can be designed to exhibit a linear phase behaviour, together with very high rates of attenuation in the transition band. The latter property is particularly desirable to enable the separation of F/A-18 vibration response data into its individual modal responses with as little interaction between bands as possible. These filters are called finite impulse-response (FIR) linear phase filters, and a Fortran program which enables optimum equi-ripple FIR lowpass, highpass, bandpass and bandreject filters to be designed is available and is based on the McClellan-Parks-Rabiner algorithm [2]. An updated version of this program appears in [3], and is the version that was eventually adopted for the FIR filter design work.

In order to enable the computed filter coefficients to be easily verified, the program described in [3] was modified to calculate and display the magnitude and phase of the Fourier transform of the filter impulse response. The group delay is also computed and output, making it easy to check the filter's linear phase performance in the passband. The program implemented here is called EQFIR.

Once the filter design has been completed and a set of suitable filter coefficients determined, the filtering action can be obtained by time-domain convolution of the input signal with the filter's symmetrical impulse response. An efficient algorithm for performing time domain filtering has been presented by Rabiner [4], together with Fortran source code for implementation as a callable function. This function was modified to make it possible to apply it to simultaneously filtering multiplexed channels of data, as found in AETE data files. The new version is presented in Appendix A and the function is called FILT1. An example program illustrating the use of this function is presented later.

2.3 Design specification for FIR digital filters

In order to design an FIR filter using program EQFIR, it is necessary to prepare a specification that defines the filter performance characteristics. Figure 1 shows a general bandpass filter that can be used as a basis for developing a suitable design specification. M_{pb} is the desired magnitude response in the passband, and is usually set to unity. The passband is specified by the

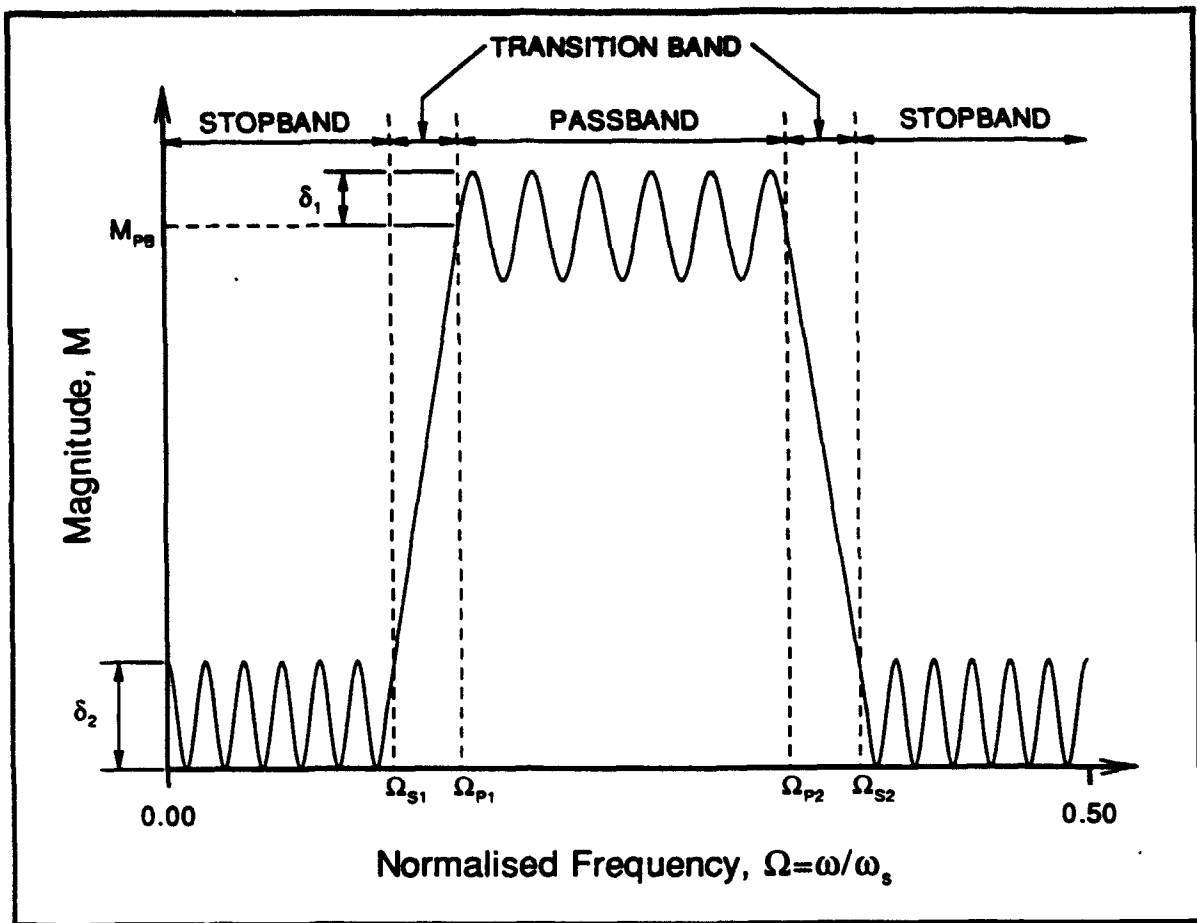


Figure 1: Parameters for FIR filter specification

two edge frequencies of the interval $[\Omega_{P1}, \Omega_{P2}]$, and the two stopbands are specified by the edge frequencies for the intervals $[0, \Omega_{S1}]$ and $[\Omega_{S2}, 0.50]$. Note that the frequency $\Omega = \omega/\omega_s$ has been normalized by the sampling frequency, $\omega_s = 2\pi f_s$, so $\Omega = 0.50$ corresponds to the Nyquist frequency. The desired ripple in the passband is denoted by δ_1 , and is measured as a peak value relative to M_{PB} . The ripple in the stopband is denoted by δ_2 , and is measured as a peak value relative to the zero level.

If multiple transition bands are present in the filter, it is important to note that they must all be of the same width. If they are not, then the filter design program may not be able to achieve a smooth rolloff through the transition bands, resulting in unsatisfactory filter characteristics.

2.4 Design procedure for FIR digital filters

The FIR filter design program EQFIR expects four lines of data to be provided, and the required format is presented below:

```
NFILT  JTYPE  NBANDS  LGRID  SFREQ
EDGE(1) ... EDGE(2*NBANDS)
DESR(1) ... DESR(NBANDS)
WGHT(1) ... WGHT(NBANDS)
```

Each of the parameters required by the specification of the filter design problem are defined as follows:

NFILT The filter length in samples. NFILT must satisfy $3 \leq \text{NFILT} \leq \text{NFMAX}$. In the present implementation NFMAX has been set to 1024.

JTYPE The type of filter.

JTYPE = 1 for multiple passband/stopband.

JTYPE = 2 for differentiator.

JTYPE = 3 for Hilbert transformer.

The last two filter types are outside the scope of this memorandum and will not be discussed any further.

NBANDS The number of frequency bands, up to a maximum of 10 bands.

LGRID The grid density in each band used for computing the extremal frequencies, assumed to be 16 if a value of zero is specified.

SFREQ The sampling frequency for the filter design. Used to scale the frequency axis for listings of filter frequency response and group delay characteristics.

EDGE An array of size 20, containing the frequency bands, specified by upper and lower cutoff frequencies, up to a maximum of 10 bands.

DESR An array of size 10, containing the desired frequency response in each band. In the passband the desired value is usually set to unity, and in the stopband it is usually set to zero.

WGHT An array of size 10, containing the positive weight function for each band. Each value is the ratio of the passband ripple and ripple in the given band. For stopbands it is usually calculated as δ_1/δ_2 , while for passbands it is usually set to unity.

Note that the frequencies to be input in the EDGE array are normalized by the sampling frequency, and each frequency band pair specifies a closed subinterval of the frequency axis $[0, 1/2]$. The arrays DESR and WGHT then specify the ideal desired response and the weight function in each band.

When designing a digital filter using EQFIR, it is necessary to specify the length of the filter in samples. A program for computing an estimate for the value of N is presented in Appendix B. It is based on a formula developed for estimating the value of N for lowpass digital FIR filters [5], and can also be applied to estimating N for bandpass filters. Given the filter parameters Ω_s , Ω_p , δ_1 and δ_2 , the minimum filter impulse response duration required to meet the specifications can be estimated from the relation

$$N = D_{\infty}(\delta_1, \delta_2)/\Delta\Omega - (0.51244 \log_{10} K + 11.01217)\Delta\Omega + 1$$

where

$$K = \delta_1/\delta_2$$

$$D_{\infty}(\delta_1, \delta_2) = [0.005309 (\log_{10} \delta_1)^2 + 0.07114 (\log_{10} \delta_1) - 0.4761] \log_{10} \delta_2 - [0.00266 (\log_{10} \delta_2)^2 + 0.5941 (\log_{10} \delta_2) + 0.4278]$$

$$\Delta\Omega = | \Omega_p - \Omega_s |$$

The above equation generally provides a good first estimate of N, particularly for low-order FIR filters, as the relative error is 1.3% for $\delta_1 \leq 0.1$ and $\delta_2 \leq 0.1$ [5]. If the iterated filter design obtained using EQFIR does not meet the required filter specification, then the value of N can be

increased and the filter coefficients re-computed. Using this technique of successive approximation, a satisfactory solution is usually obtained after only a few additional trials.

The theoretical group delay G of an FIR filter can be quickly determined by using the following formula

$$G = \frac{(N - 1)}{2f_s} \quad (\text{seconds})$$

When designing analogue filters, it is common to specify the rolloff rate in the transition band in terms of dB/octave. For n -pole Butterworth filters, the rolloff rate is simply $6n$ dB/octave ($20n$ dB/decade), which yields a linear plot on a logarithmic frequency scale. When designing digital filters using EQFIR, it is the width of the transition band that has to be specified.

If the required rate of attenuation in dB/octave is known, and the magnitude response in the passband is assumed to be unity, then the width of the lower and upper transition bands, $\Delta\Omega_{\text{LTB}}$ and $\Delta\Omega_{\text{UTB}}$, can be estimated from the following two formulae

$$\Delta\Omega_{\text{LTB}} = \Omega_{p1}(1 + R_{\text{LTB}}/(20 \log_{10} \delta_2 - R_{\text{LTB}}))$$

$$\Delta\Omega_{\text{UTB}} = -\Omega_{p2}(20 \log_{10} \delta_2)/R_{\text{UTB}}$$

where

R_{LTB} = desired average rolloff rate in the lower transition band in dB/octave

R_{UTB} = desired average rolloff rate in the upper transition band in dB/octave

For a given rolloff rate, the width of the lower transition band will always be larger than the width of the upper transition band. If a minimum average rolloff rate is required, then this will enforce a constraint on the maximum width of the lower transition band. The edge frequencies of the upper and lower transition bands can be computed using the relations

$$\Omega_{s1} = \Omega_{p1} - \Delta\Omega_{\text{LTB}}$$

$$\Omega_{s2} = \Omega_{p2} + \Delta\Omega_{\text{LTB}}$$

It is also common practice in filter design to specify the passband and stopband ripple in dB relative to the passband amplitude. Hence it is necessary to be able to convert such specifications into a format compatible with the input parameters of EQFIR. If δ_{dB1} and δ_{dB2} are the passband and stopband ripple specified in dB relative to the passband amplitude, then the corresponding values of δ_1 and δ_2 can be computed from the following relations

$$\delta_1 = 10^{\delta_{\text{dB1}}/20} - 1$$

$$\delta_2 = 10^{\delta_{\text{dB2}}/20}$$

2.5 Design of a 10–20 Hz bandpass FIR filter

In order to isolate the Mode I response in the 10–20 Hz band, a digital FIR bandpass filter was designed using the techniques described above. The sampling rate was $f_s = 606.06$ Hz, and the bandpass edge frequencies were taken to be 10 Hz and 20 Hz. The passband ripple was specified to be no greater than 0.05 dB, while the stopband ripple was to be –60 dB or better. In order to provide good rejection of any low-frequency response, the desired average rolloff rate in the lower transition band was specified to be 140 dB/octave. Hence we have that

$f_s = 606.06 \text{ Hz}$		
$\delta_{dB1} = 0.05 \text{ dB}$	$\Leftrightarrow$	$\delta_1 = 0.00577$
$\delta_{dB2} = -60 \text{ dB}$	$\Leftrightarrow$	$\delta_2 = 0.001$
$R_{LTB} = 140 \text{ dB/octave}$		
$f_{p1} = 10 \text{ Hz}$	$\Leftrightarrow$	$\Omega_{p1} = 0.01650$
$f_{p2} = 20 \text{ Hz}$	$\Leftrightarrow$	$\Omega_{p2} = 0.03300$
$\Delta f_{LTB} = 3 \text{ Hz}$	$\Leftrightarrow$	$\Delta \Omega_{LTB} = 0.00495$
$f_{s1} = 7 \text{ Hz}$	$\Leftrightarrow$	$\Omega_{s1} = 0.01155$
$f_{s2} = 23 \text{ Hz}$	$\Leftrightarrow$	$\Omega_{s2} = 0.03795$

Now, in the formula for estimating N, $\Delta\Omega = \Delta\Omega_{LTB} = 0.00495$, and using the above values of δ_1 and δ_2 we obtain

$$N = 549$$

$$K = 5.77$$

The data file for EQFIR for the 10–20 Hz bandpass filter then becomes

549	1	3	16	606.06	
0.0	0.01155	0.01650	0.03300	0.03795	0.5
0.0	1.0	0.0			
5.77	1.0	5.77			

The results output by program EQFIR for N = 549 are presented below.

	BAND 1	BAND 2	BAND 3
LOWER BAND EDGE HZ	0.000	10.000	23.000
UPPER BAND EDGE HZ	7.000	20.000	303.030
LOWER BAND EDGE	0.0000000	0.0165000	0.0379500
UPPER BAND EDGE	0.0115500	0.0330000	0.5000000
DESIRED VALUE	0.0000000	1.0000000	0.0000000
WEIGHTING	5.7700000	1.0000000	5.7700000
DEVIATION	0.0014360	0.0082859	0.0014360
DEVIATION IN DB	-56.8567657	0.0716737	-56.8567657

Looking at the bottom two rows giving the deviation, it can be seen that this filter does not quite meet the design specifications, as both the passband and stopband ripple are slightly greater than desired. After a number of additional trials a final value of N = 571 was obtained, and the results of the design process are given below.

	BAND 1	BAND 2	BAND 3
LOWER BAND EDGE HZ	0.000	10.000	23.000
UPPER BAND EDGE HZ	7.000	20.000	303.030
LOWER BAND EDGE	0.0000000	0.0165000	0.0379500
UPPER BAND EDGE	0.0115500	0.0330000	0.5000000
DESIRED VALUE	0.0000000	1.0000000	0.0000000
WEIGHTING	5.7700000	1.0000000	5.7700000
DEVIATION	0.0008924	0.0051490	0.0008924
DEVIATION IN DB	-60.9890862	0.0446090	-60.9890862

The upper plot in Figure 2 shows the symmetric impulse response of the filter. The middle plot shows the magnitude response of the filter over a frequency range of 0–100 Hz, and it is seen that the passband ripple is within 0.05 dB, and the stopband ripple is less than –60 dB. The lower plot shows the group delay characteristics of the bandpass filter computed from the filter's phase response, and it is seen that a uniform group delay of 0.4703 seconds is obtained in the passband.

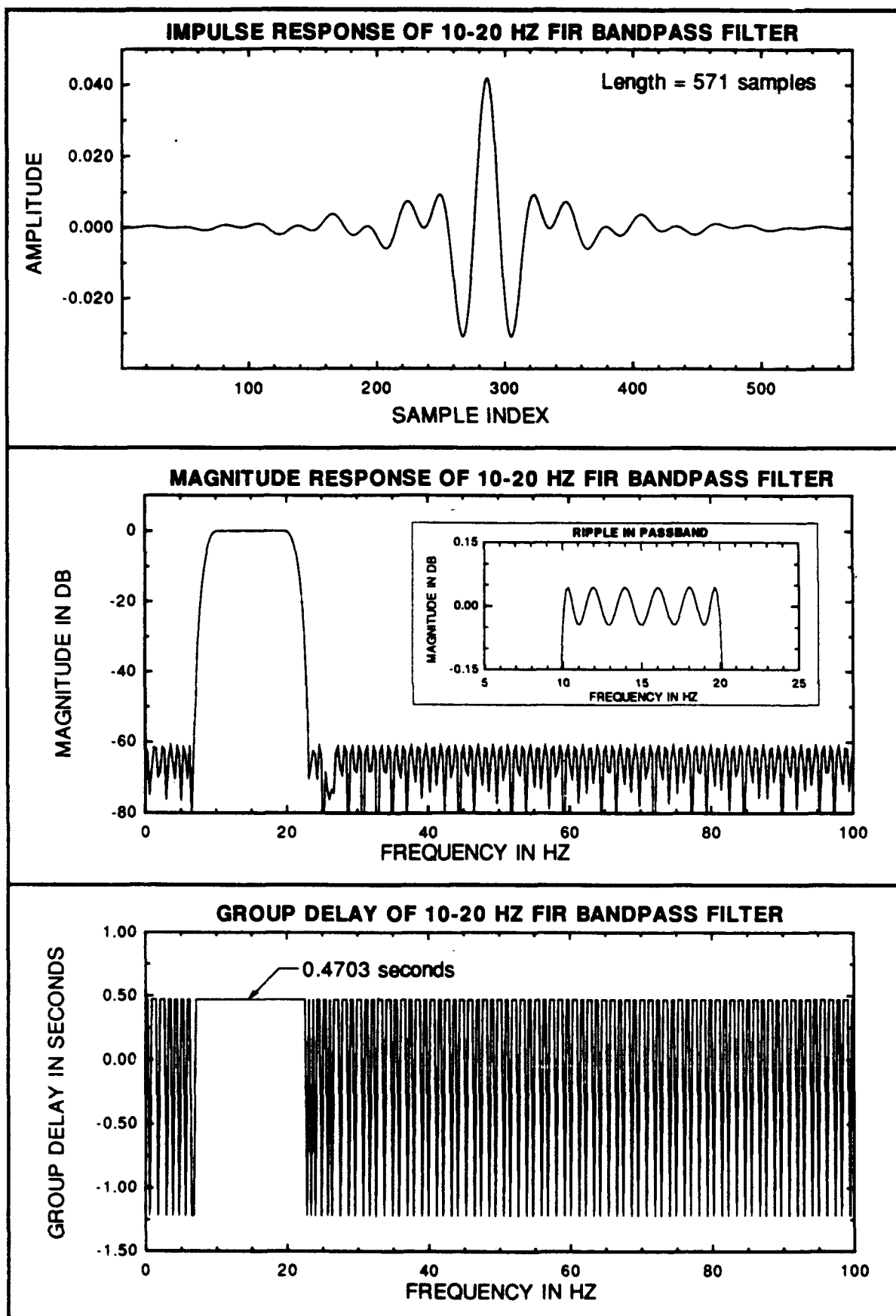


Figure 2: Impulse response, magnitude response and group delay of 10-20 Hz FIR bandpass filter.

The time delay indicated by the lower plot corresponds closely to the group delay $G = 0.4711$ seconds computed from the theoretical formula presented earlier. The group delay plot confirms that the filter exhibits linear-phase characteristics in the passband. This group delay is also evident in the stopband, except that a 180° phase change is present at each of the zeros in the transfer function.

2.6 Design of a 32–52 Hz bandpass FIR filter

In order to isolate the Mode II response in the 32–52 Hz band, a digital FIR bandpass filter was designed to work with the data sampled at $f_s = 606.06$ Hz, and the bandpass edge frequencies were taken to be 32 Hz and 52 Hz. The passband ripple was specified to be no greater than 0.05 dB, while the stopband ripple was to be no greater than -60 dB. In order to provide good rejection of any low-frequency response, particularly any Mode I response that might be present, the desired average rolloff rate in the lower transition band was specified to be 144 dB/octave. Hence we have that

$$\begin{array}{lll}
 f_s = 606.06 \text{ Hz} & & \\
 \delta_{dB1} = 0.05 \text{ dB} & \Leftrightarrow & \delta_1 = 0.00577 \\
 \delta_{dB2} = -60 \text{ dB} & \Leftrightarrow & \delta_2 = 0.001 \\
 R_{LTB} = 144 \text{ dB/octave} & & \\
 f_{p1} = 32 \text{ Hz} & \Leftrightarrow & \Omega_{p1} = 0.05280 \\
 f_{p2} = 52 \text{ Hz} & \Leftrightarrow & \Omega_{p2} = 0.08580 \\
 \Delta f_{LTB} = 9.41 \text{ Hz} & \Leftrightarrow & \Delta \Omega_{LTB} = 0.01553 \\
 f_{s1} = 22.59 \text{ Hz} & \Leftrightarrow & \Omega_{s1} = 0.03727 \\
 f_{s2} = 61.41 \text{ Hz} & \Leftrightarrow & \Omega_{s2} = 0.10133
 \end{array}$$

Now, $\Delta \Omega = \Delta \Omega_{LTB} = 0.01553$, and using δ_1 and δ_2 in the formula for estimating N we obtain

$$N = 176$$

$$K = 5.77$$

A data file suitable for use with program EQFIR was created, and the resulting filter for $N = 176$ did not quite meet the design specifications for stopband and passband ripple. After additional trials with increased values of N , a suitable design was obtained with $N = 195$. The final version of the data file is given below.

195	1	3	16	606.06	
0.0	0.03727	0.05280	0.08580	0.10133	0.5
0.0	1.0	0.0			
5.77	1.0	5.77			

The results obtained by program EQFIR for $N = 195$ are presented below.

	BAND 1	BAND 2	BAND 3
LOWER BAND EDGE HZ	0.000	32.000	61.412
UPPER BAND EDGE HZ	22.588	52.000	303.030
LOWER BAND EDGE	0.0000000	0.0528000	0.1013300
UPPER BAND EDGE	0.0372700	0.0858000	0.5000000
DESIRED VALUE	0.0000000	1.0000000	0.0000000
WEIGHTING	5.7700000	1.0000000	5.7700000
DEVIATION	0.0009579	0.0055273	0.0009579
DEVIATION IN DB	-60.3732834	0.0478770	-60.3732834

The upper plot in Figure 3 shows the symmetric impulse response of the filter. The middle plot shows the magnitude response of the filter over a frequency range of 0–100 Hz, and it is seen that the passband ripple is within 0.05 dB, and the stopband ripple is less than -60 dB.

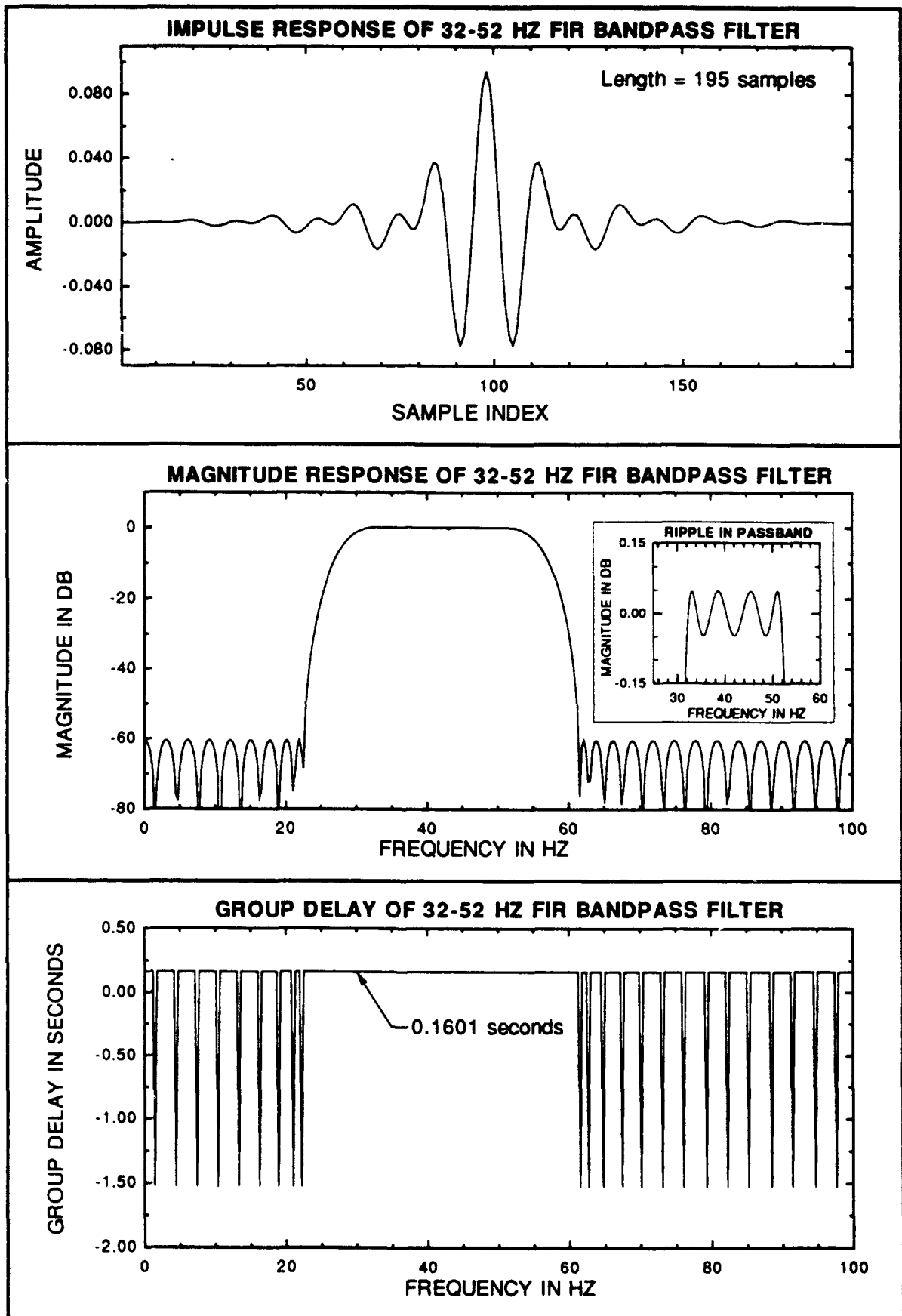


Figure 3: Impulse response, magnitude response and group delay of 32-52 Hz FIR bandpass filter.

The lower plot shows the group delay characteristics computed from the bandpass filter's phase response, and it is seen that a uniform group delay of 0.1601 seconds is obtained in the passband which, to all intents and purposes, is identical to the theoretical group delay.

2.7 Design of an 8 Hz cutoff highpass FIR filter

In order to filter the measured vibration response to remove any very low-frequency or static response, a digital FIR highpass filter with 8 Hz cutoff was designed to work with the data sampled at $f_s = 606.06$ Hz. The cutoff frequency was chosen so that there would be no significant filtering of any Mode I response present in the measured data. The passband ripple was specified to be no greater than 0.05 dB, while the stopband ripple was to be no greater than -60 dB. In order to provide good rejection of any low-frequency response, the upper edge of the stopband was set to 5 Hz.

$$\begin{array}{lll}
 f_s = 606.06 \text{ Hz} & & \\
 \delta_{dB1} = 0.05 \text{ dB} & \Leftrightarrow & \delta_1 = 0.00577 \\
 \delta_{dB2} = -60 \text{ dB} & \Leftrightarrow & \delta_2 = 0.001 \\
 f_{S1} = 0 \text{ Hz} & \Leftrightarrow & \Omega_{S1} = 0.00000 \\
 f_{S2} = 5 \text{ Hz} & \Leftrightarrow & \Omega_{S2} = 0.00825 \\
 f_{P1} = 8 \text{ Hz} & \Leftrightarrow & \Omega_{P1} = 0.01320 \\
 f_{P2} = 303.03 \text{ Hz} & \Leftrightarrow & \Omega_{P2} = 0.50000 \\
 \Delta f_{TB} = 3 \text{ Hz} & \Leftrightarrow & \Delta \Omega_{TB} = 0.00495
 \end{array}$$

Now, $\Delta \Omega = \Delta \Omega_{TB} = 0.00495$, and using δ_1 and δ_2 in the formula for estimating N we obtain

$$N = 549$$

$$K = 5.77$$

A data file suitable for use with program EQFIR was created, and the resulting filter for $N = 549$ did not quite meet the design specifications for stopband and passband ripple. After additional trials with increased values of N, a suitable design was obtained with $N = 559$, and the final version of the data file is given below.

559	1	2	16	606.06
0.0	0.00825	0.01320	0.5	
0.0	1.0	0.0		
5.77	1.0	5.77		

The results output by program EQFIR for $N = 559$ are presented below.

	BAND 1	BAND 2
LOWER BAND EDGE HZ	0.000	8.000
UPPER BAND EDGE HZ	5.000	303.030
LOWER BAND EDGE	0.0000000	0.0132000
UPPER BAND EDGE	0.0082500	0.5000000
DESIRED VALUE	0.0000000	1.0000000
WEIGHTING	5.7700000	1.0000000
DEVIATION	0.0009408	0.0054287
DEVIATION IN DB	-60.5296059	0.0470253

The upper plot in Figure 4 shows the symmetric impulse response of the filter. The middle plot shows the magnitude response of the filter over a frequency range of 0-100 Hz, and it is seen that the passband ripple is within 0.05 dB, and the stopband ripple is less than -60 dB. The lower plot shows the group delay characteristics computed from the highpass filter's phase response, and it is seen that a uniform group delay of 0.4604 seconds is obtained in the passband. This is in close agreement with the theoretical group delay $G = 0.4579$ seconds.

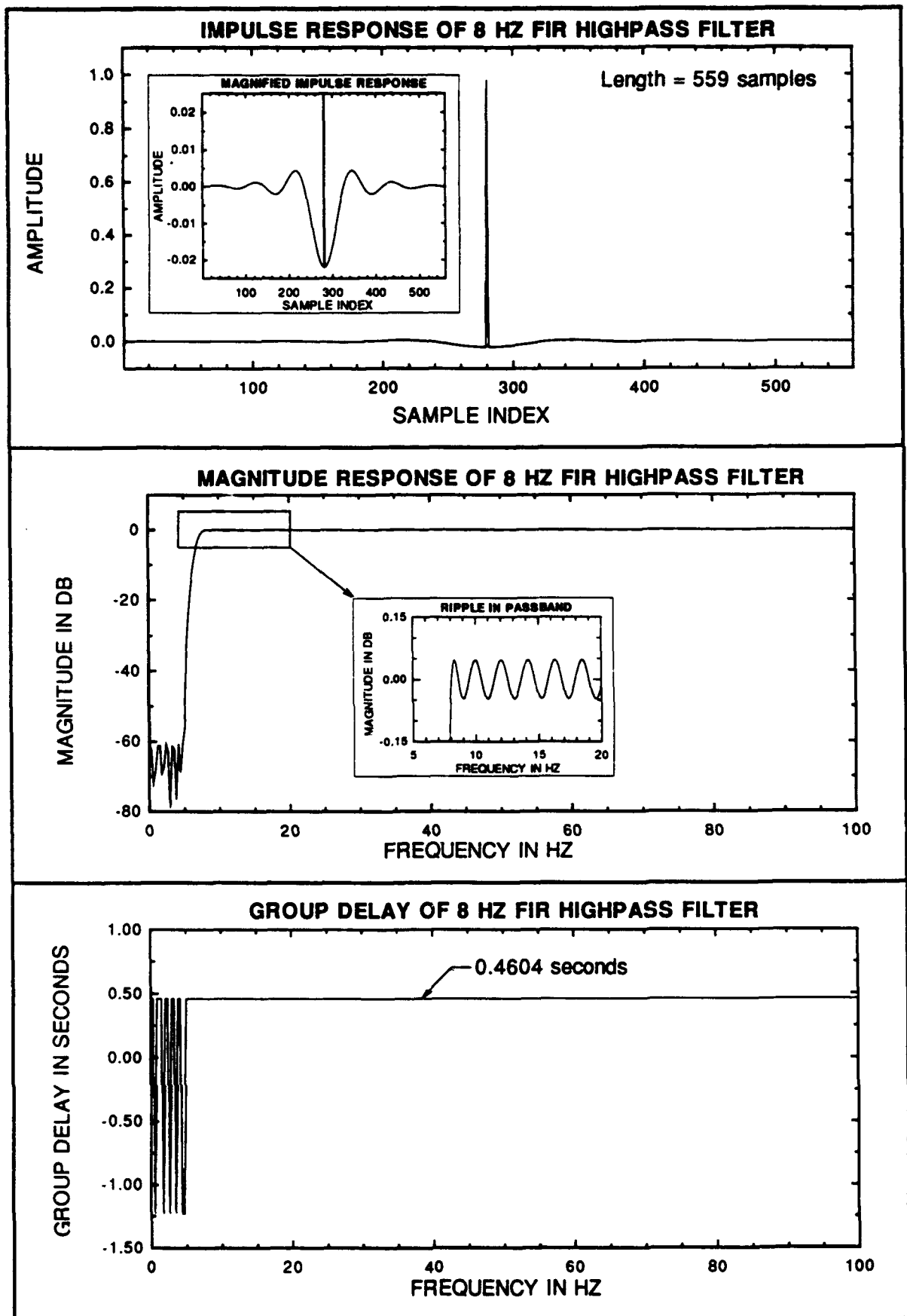


Figure 4: Impulse response, magnitude response and group delay of 8 Hz cutoff FIR highpass filter.

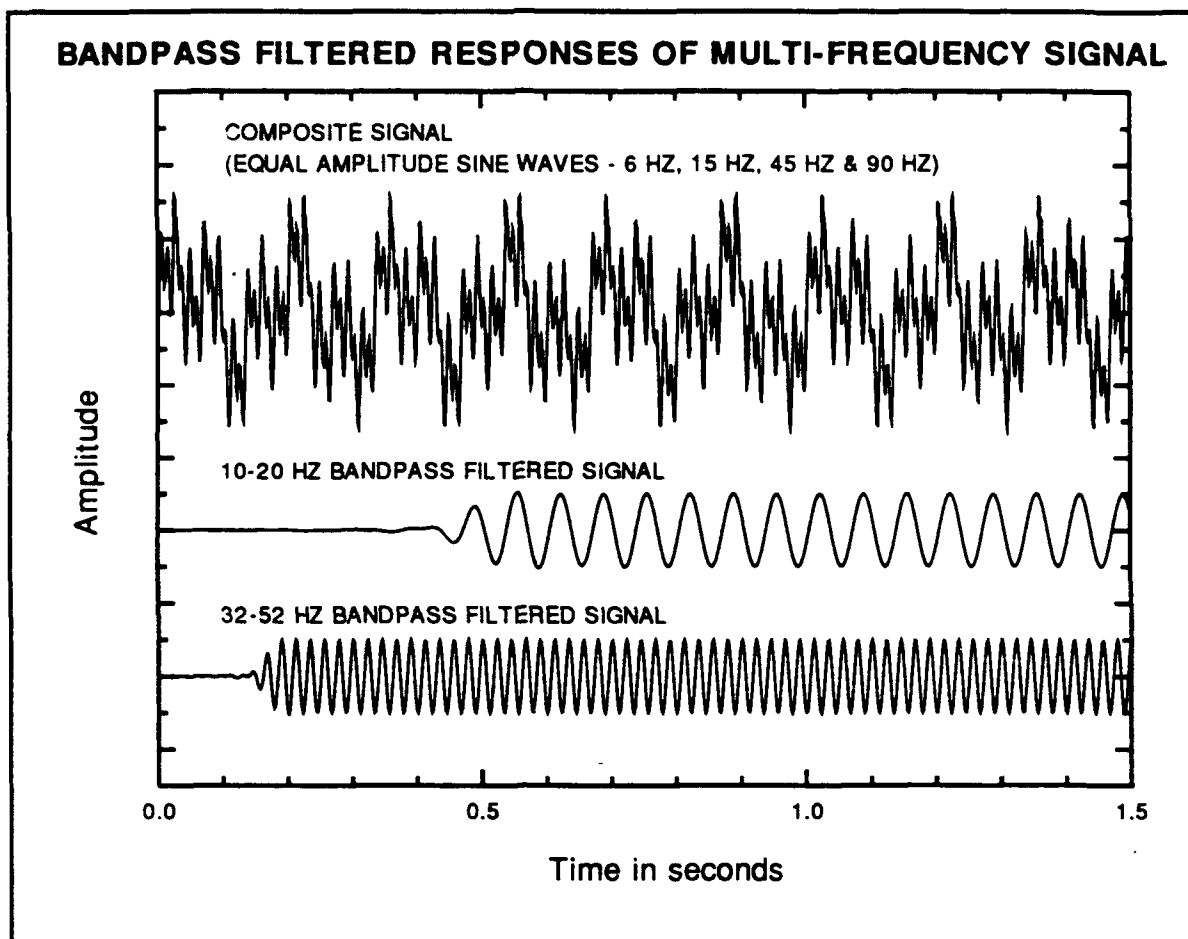


Figure 5: Bandpass filtered responses of multi-frequency signal.

3.0 FILTERING PERFORMANCE OF FIR FILTERS

Convolution of a filter's impulse response with the input signal in order to perform the filtering operation is a process that is very computationally intensive. This belies the relative simplicity of the time-domain FIR filtering algorithm. In order to reduce the computation time, a frequency domain technique using FFTs has been developed [6]. For larger filter lengths it offers significant speed gains, which is beneficial when filtering large amounts of flight test data.

A Fortran subroutine implementing the "overlap-add" technique of fast convolution in the frequency domain can be found in [3]. The name of the subroutine is RFILT, and a listing is provided in Appendix C.

A Fortran program TESTFILT was written in order to verify the operation of the filtering algorithms when using the filter coefficients produced by program EQFIR, and a listing is provided in Appendix D. TESTFILT generates a multi-frequency signal consisting of four equal amplitude sine waves with frequencies at 6 Hz, 15 Hz, 45 Hz and 90 Hz. These frequencies were chosen to be representative of those that occur in F/A-18 empennage vibrations. Both time-domain and frequency-domain filtering techniques were used to permit a comparison of filtering quality, as well as providing an independent check of the results of each method.

The upper plot in Figure 5 shows the waveform produced using the four equal amplitude sine waves. The middle and lower plots show the results after applying the 10–20 Hz and 32–52 Hz bandpass filters. Note that the results of filtering in the time and frequency domains were effectively identical to about five significant figures, so there was no discernible difference

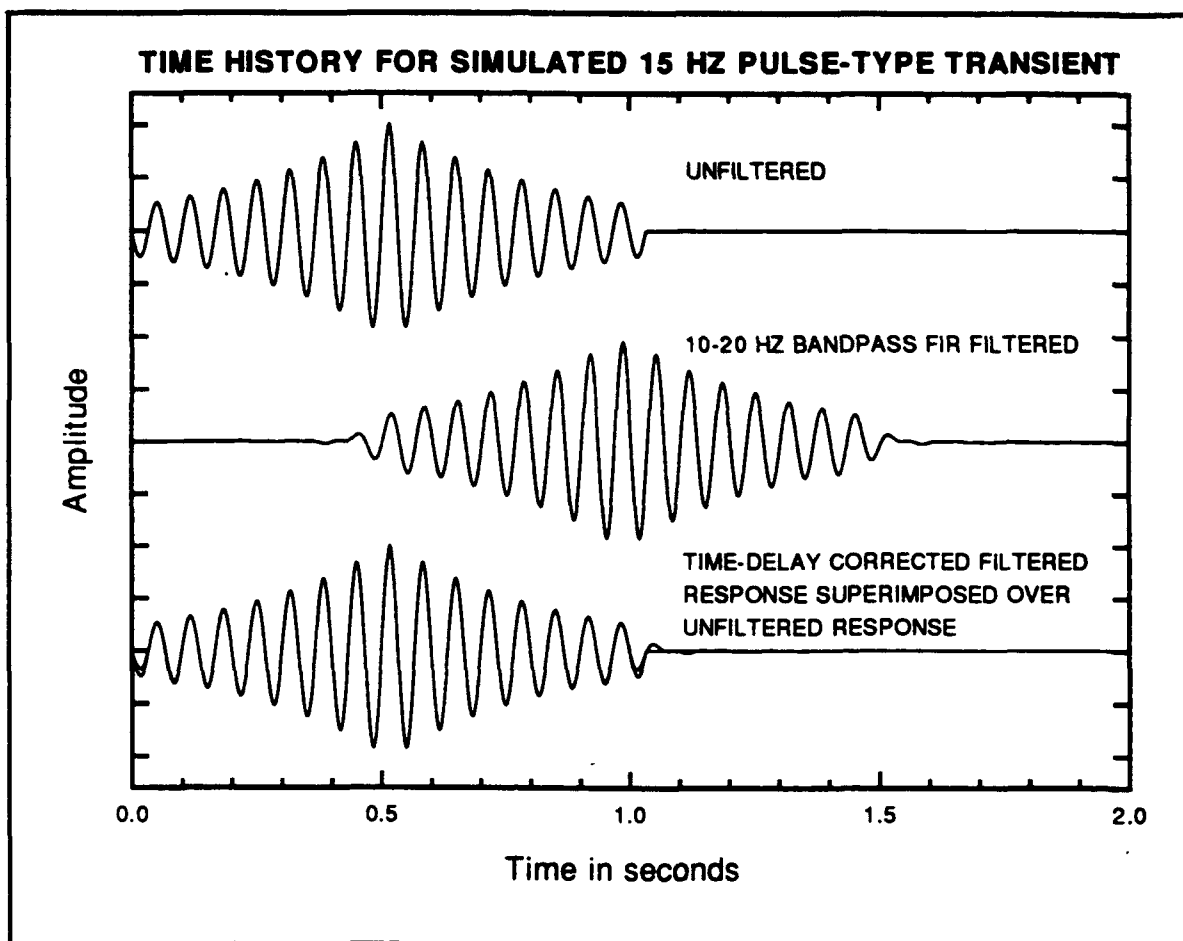


Figure 6: Comparison of simulated 15 Hz response transient with the 10–20 Hz bandpass filtered response.

between the results when the plots were overlaid. The bandpass filtered responses also show clearly the time delay introduced by the filtering process.

For the 10–20 Hz bandpass filter, it was found that the frequency-domain filtering technique was 7.7 times faster than its time-domain counterpart. The frequency-domain 32–52 Hz bandpass filter was found to be 2.4 times faster than the time-domain implementation. For a constant size of FFT block length, the speed of the frequency-domain filter was independent of the number of filter coefficients, while the speed of the time domain filter was approximately inversely proportional to the number of coefficients.

The frequency-domain filtering technique relies on the swiftness of the forward and inverse FFTs to obtain its speed improvement. A number of other implementations of the FFT algorithm were incorporated into subroutine RFILT, but the original code using subroutines FAST and FSST (given in [3]) proved to be the most efficient. As a result of their computational efficiency and well-documented performance characteristics, these two subroutines were also adopted for all other work involving FFT computations.

4.0 TRANSIENT PERFORMANCE OF FIR FILTERS

Filtering operations are normally accompanied by changes in the waveform of the signal, particularly if the signal has many frequency components and some of these are removed or significantly attenuated. Before applying the FIR filters designed above to F/A-18 vibration responses, the effects of the filtering action on a simulated pulse-type transient waveform that represents a burst of vibration at a particular modal frequency was studied. The Fortran program CHECKQ in Appendix E can generate a 15 Hz or 45 Hz sinusoidal waveform that is

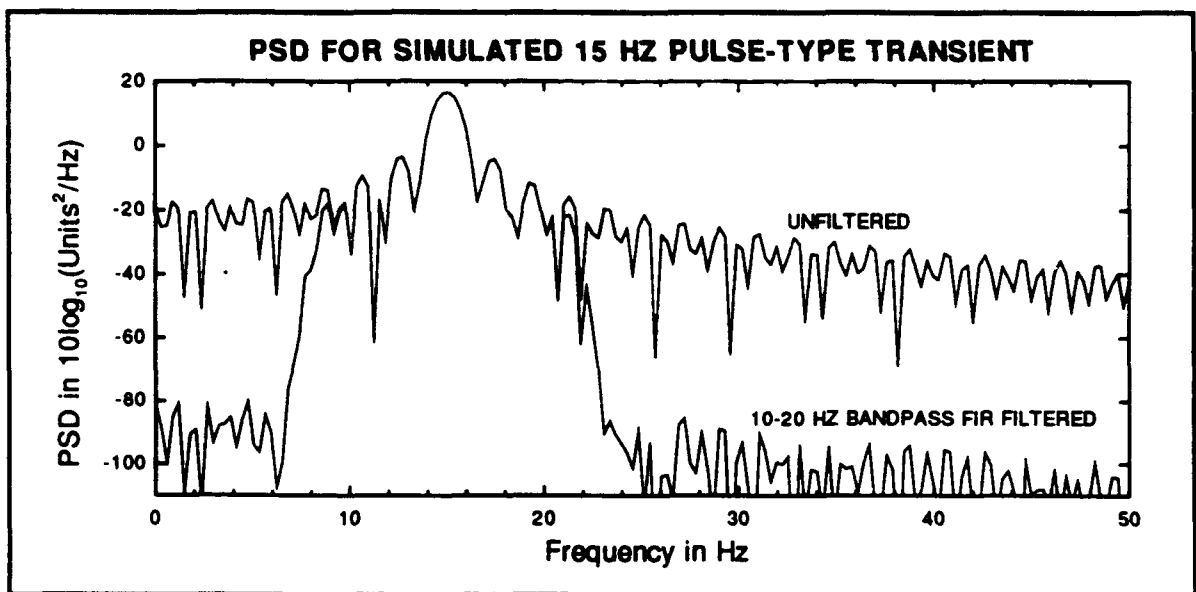


Figure 7: Comparison of power spectra for simulated 15 Hz response transient before and after 10–20 Hz bandpass filtering.

approximately one second in length. To make this pulse better simulate the build-up and decay of a typical structural vibration response transient, the waveform can be windowed using an exponential or a sine-squared weighting function. The damping ratio of the decay in the exponential window was chosen to be 3% of critical, which is typical of aircraft structural vibration modes. The program then filters the transient, and the output consists of listings of the time histories and power spectra of the unfiltered and filtered responses. The time history listing includes the filtered sequence with the theoretical time delay removed, making it easy to conduct a point by point comparison of the filtered waveform with the unfiltered data.

The upper plot in Figure 6 shows the unfiltered response of the simulated 15 Hz exponentially-weighted response transient, and the middle plot shows the resulting 10–20 Hz FIR bandpass filtered response. The lower plot shows the time-delay corrected version of the filtered response superimposed over the unfiltered transient. It is evident that the bandpass filtered response is very similar to the raw data, except at the leading and trailing edges of the pulse where there is a little extra oscillation that is not present in the unfiltered data. This is caused by the removal of some of the frequency components associated with the discontinuity at the leading and trailing edges of the simulated pulse.

Figure 7 shows the power spectra obtained from the time histories of the unfiltered and bandpass filtered 15 Hz pulse. In the 10–20 Hz passband of the bandpass filter, there is very good agreement between the two spectra, with all the side lobes being present with the correct amplitude. As expected, the energy in the stopbands has been significantly reduced by the action of the bandpass filter.

The results shown in Figures 6 and 7 show that the processing performed by the FIR filters will provide only a minor modification to any transient waveforms falling within the passband of the filter. Even better results were obtained for the sine-squared weighting of the sinusoidal pulse, but these are not presented here as it was found that the sine-squared weighting had created a less demanding transient because it goes smoothly to zero at either end.

5.0 CONCLUSION

A complete design procedure for computing FIR digital filter coefficients has been presented. A set of filters suitable for processing F/A-18 flight test data has been determined, and includes 10-20 Hz and 32-52 Hz bandpass filters, as well as an 8 Hz highpass filter. A program for performing time-domain and frequency-domain filtering was written and has been used to verify the filter coefficients and the operation of the filtering algorithms. A program for studying the effects of bandpass filtering on a simulated pulse-like sinusoidal transient waveform whose fundamental frequency lies in the filter's passband was also written. The results indicate that the FIR digital filters used here have a minimal effect on the waveforms of transients whose frequency content lies within the passband of the filter.

6.0 REFERENCES

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A Fortran Subroutine For Digital Filtering In The Time Domain

A subroutine that can read in filter coefficients from an output file written by the filter design program EQFIR is also provided, and can be found at the end of the listing.

```

REAL FUNCTION FILT1(X,XSAV,H,N,IPT,INIT)
C      X      A SINGLE, REAL DATA SAMPLE
C
C      XSAV    A WORK ARRAY, DIMENSION OF 2*N
C
C      H      FILTER COEFFICIENT ARRAY, DIMENSION OF N
C
C      N      NUMBER OF FILTER COEFFICIENTS
C
C      IPT     A LOCAL VARIABLE THAT MUST BE SAVED BETWEEN
C              SUCCESSIVE CALLS TO THIS FILTER ROUTINE.
C              A SEPARATE ARGUMENT WAS ADDED TO ALLOW THIS
C              ROUTINE TO BE USED WHEN FILTERING MORE THAN
C              ONE SEQUENCE AT A TIME.
C
C      INIT=0  TO INITIALIZE FILTER BEFORE RUNNING
C              -1  WHEN RUNNING
C
C      FILT1   ON RETURN, FILT1 IS THE FILTERED OUTPUT SAMPLE
C
      REAL      X,H(1),XSAV(1),Y
      INTEGER N,INIT,N2,IPT,I
C
      IF (INIT.EQ.0) THEN
        DO I = 1,2*N
          XSAV(I) = 0.0
        ENDDO
        IPT = N + 1
        FILT1 = 0.0
        RETURN
      ENDIF
C
      XSAV(IPT)   = X
      XSAV(IPT-N) = X
      Y=0.0
      DO I = 1,N
        Y = Y + H(I)*XSAV(IPT-I+1)
      ENDDO
      IPT = IPT + 1
      IF (IPT.GT.2*N) IPT = N + 1
      FILT1 = Y
C
      RETURN
      END

```

```

SUBROUTINE READH(NFILT,H,FILENAME,UNITNO)
C THIS PROCEDURE READS IN THE COEFFICIENTS OF THE FILTER IMPULSE
C RESPONSE INTO THE ARRAY H.

```


APPENDIX B

A Fortran Program For Estimating N For FIR Digital Filters

The following Fortran program can be used for estimating the value of N for FIR digital filters. The required input data consists of the sampling rate f_s , the width of the transition band Δf_{TB} , and the passband and stopband ripple δ_1 and δ_2 . Note that the frequency information can be supplied in any consistent units, as it is automatically non-dimensionalised with respect to f_s within the program.

[illegible]

PROGRAM ESTIMN

C This program is used to estimate the order of the FIR filter
C required to meet the filter specifications.

IMPLICIT NONE

INTEGER N

REAL FSAMPLE, DELTAF, DELTA1, DELTA2

```
REAL      A1,A2,A3,A4,A5,A6,B1,B2,DF,F,D,LOGD1,LOGD2
```

REAL ALOG10

```
WRITE(*, '(/,1X,A,$)') 'INPUT Fs, Delta_F, Delta_1 and Delta_2: '
```

```
READ (*,*) FSAMPLE, DELTAF, DELTA1, DELTA2
```

A1 = 5.309E-3

A2 = 7.114E-2

A3 = -4.761E-1

A4 = -2.660E-3

A5 = -5.941E-1

A6 = -4.278E-1

B1 = 11.01217

B2 = 0.51244

$$DF = \Delta F / F_{SAMPLE}$$

LOGD1 = ALOG10 (DELTA1)

$$\text{LOGD2} = \text{ALOG10}(\text{DELTA2})$$
$$F = B1 + B2 * (LOGD1 - LOGD2)$$
$$D = (A1*LOGD1*LOGD1 + A2*LOGD1 + A3)*LOGD2 + A4*LOGD1*LOGD1 + A5*LOGD1 + A6$$
$$N = D/DF - F*DF + 1$$

```
WRITE(*, '(/,1X,A,I5,/)') 'Estimated N = ',N
```

STOP

END

APPENDIX C

A Fortran Subroutine For Digital Filtering In The Frequency Domain

The Fortran subroutine RFILT [3] can be used to perform digital filtering of signals in the frequency domain. A listing of this subroutine is presented below.

```
C SUBROUTINE: RFILT
C FILTER ONE FRAME (I.E., NP POINTS) OF DATA
C
C PROGRAM ASSUMES:
C   1. R(NP+1) TO R(2*NP) HAS NOT BEEN USED SINCE LAST CALL TO
C      'RFILT' AND WAS ZERO BEFORE FIRST CALL.
C   2. INPUT AND OUTPUT DATA (TIME SERIES) ARE IN R(1) TO R(NP)
C   3. F(1) TO F(2*NP+2) IS THE FILTER FREQUENCY RESPONSE.
C   4. DATA ARE STORED IN F AS: DC,0.,FIREAL,FIIMAG,
C      F2REAL,F2IIMAG,...,FSAMPLING/2.,0.
C   5. S(1) TO S(NP) IS A SCRATCH ARRAY; IT MAY BE USED IN 'MAIN'.
C   6. IMPULSE RESP. OF FILTER F IS ZERO FOR NP+1 THRU 2*NP
C      (I.E. SECOND HALF OF H(T)=0, WHERE F(W)=FFT(H) )
C-----
C
C   SUBROUTINE RFILT(R, F, S, NP)
C     DIMENSION S(1), R(1), F(1)
C     NPT2 = NP*2
C
C   STORE PREVIOUS TAIL IN SCRATCH ARRAY S(.);
C   ZERO SECOND HALF OF R--FIRST HALF OF R IS NEW DATA TO BE FILTERED
C
C     DO 10 K=1,NP
C       NPPK = NP + K
C       S(K) = R(NPPK)
C       R(NPPK) = 0.
C 10  CONTINUE
C
C   TAKE FFT OF DATA
C
C     CALL FAST(R, NPT2)
C
C   HANDLE VALUES AS COMPLEX VALUES
C
C     KMAX = NPT2 + 1
C     DO 20 K=1,KMAX,2
C       X = F(K)*R(K) - F(K+1)*R(K+1)
C       Y = F(K)*R(K+1) + F(K+1)*R(K)
C       R(K) = X
C       R(K+1) = Y
C 20  CONTINUE
C
C   INVERSE TRANSFORM PRODUCT
C
C     CALL FSST(R, NPT2)
C
C   ADD IN TAIL FROM PREVIOUS FRAME WHICH WAS STORED IN S(.)
C
C     DO 30 K=1,NP
C       R(K) = R(K) + S(K)
C 30  CONTINUE
C     RETURN
C     END
```

APPENDIX D

A Fortran Program For Testing FIR Filter Designs

The Fortran program TESTFILT can be used to test the efficacy of the FIR filtering algorithms using filter coefficients that have been generated by the filter design program EQFIR. The listing shown below is an example of how to implement digital filtering in the time-domain and frequency-domain using the Fortran function and subroutine presented in Appendices A and C.

CC

PROGRAM TESTFILT

C This program filters a time series point by point in the time
C domain and in the frequency domain.
C
C The execution times for both types of filtering are displayed
C for the purpose of easy comparison.

IMPLICIT NONE

INTEGER NH,NHT2,NFAST,NDATA
REAL SRATE,PI,TWOPI,DT

PARAMETER (NH = 1024)
PARAMETER (NHT2 = 2*NH)
PARAMETER (NFAST = 2*NHT2+2)
PARAMETER (NDATA = 3*NHT2)
PARAMETER (SRATE = 606.06)
PARAMETER (DT = 1.0/SRATE)
PARAMETER (PI = 3.141592654)
PARAMETER (TWOPI = 2.0*PI)

REAL*8 FSB1(10),FPB(10),FSB2(10),T
REAL XSB1,XSB2,XPB,WT,CPUTD,CPUFD
REAL H(NH),XSAV(NHT2),F(NFAST),R(NFAST),WORK1(NHT2)
REAL DTOT(NDATA),DPB(NDATA),DFIL(NDATA),DFILT(NDATA)
INTEGER NFILT,IFILTER,I,J,NSB1,NSB2,NPB
INTEGER IB,NBLOCKS,IPART,IPOS,NPT
CHARACTER FILENAME*20

! EXTERNAL FUNCTIONS

REAL SIN,SECNDS,FILT1
REAL*8 DMOD
INTEGER JMOD

IFILTER = 0

DO WHILE (IFILTER.EQ.0)

WRITE(*,*)
WRITE(*,*) 'PROGRAM FOR TESTING FIR FILTER DESIGNS'
WRITE(*,*) '-----'
WRITE(*,*) ' 1 = 10 HZ TO 20 HZ BANDPASS FILTER'
WRITE(*,*) ' 2 = 32 HZ TO 52 HZ BANDPASS FILTER'
WRITE(*,*) ' 3 = 5 HZ LOWPASS FILTER'
WRITE(*,*) ' 4 = 8 HZ HIGHPASS FILTER'
WRITE(*,*)
WRITE(*, '(1X,A,\$)') 'INPUT YOUR SELECTION: '
READ(*,*) IFILTER
IF (IFILTER.LT.1 .OR. IFILTER.GT.4) IFILTER = 0
ENDDO

IF (IFILTER.EQ.1) THEN
FILENAME = 'BP15.OUT'
NSB1 = 1
NPB = 1
NSB2 = 2

```

      FSB1(1) = 6.0
      FPB(1) = 15.0
      FSB2(1) = 45.0
      FSB2(2) = 90.0
    ELSE IF (IFILTER.EQ.2) THEN
      FILENAME = 'BP45.OUT'
      NSB1 = 2
      NPB = 1
      NSB2 = 1
      FSB1(1) = 6.0
      FSB1(2) = 15.0
      FPB(1) = 45.0
      FSB2(1) = 90.0
    ELSE IF (IFILTER.EQ.3) THEN
      FILENAME = 'LP05.OUT'
      NPB = 1
      NSB1 = 1
      NSB2 = 0
      FPB(1) = 2.0
      FSB1(1) = 200.0
    ELSE IF (IFILTER.EQ.4) THEN
      FILENAME = 'HP08.OUT'
      NPB = 1
      NSB1 = 1
      NSB2 = 0
      FPB(1) = 15.0
      FSB1(1) = 5.0
    ENDIF

    ! CREATE DATA SEQUENCE

    DO I = 1, NDATA
      T = (I-1)*DT
      XSB1 = 0.0
      XPB = 0.0
      XSB2 = 0.0
      DO J = 1, NSB1
        WT = TWOPI*DMOD(FSB1(J)*T, 1.0D0)
        XSB1 = XSB1 + SIN(WT)
      ENDDO
      DO J = 1, NPB
        WT = TWOPI*DMOD(FPB(J)*T, 1.0D0)
        XPB = XPB + SIN(WT)
      ENDDO
      DO J = 1, NSB2
        WT = TWOPI*DMOD(FSB2(J)*T, 1.0D0)
        XSB2 = XSB2 + SIN(WT)
      ENDDO
      DTOT(I) = XSB1 + XPB + XSB2
      DPB(I) = XPB
    ENDDO

    ! READ FILTER COEFFICIENTS

    CALL READH(NFILT, H, FILENAME, 1)

    ! INITIALIZE TIME DOMAIN FILTER AND PERFORM FILTERING

    DFILT(1) = FILT1(DTOT(1), XSAV, H, NFILT, NPT, 0)

    CPUTD = SECNDS(0.0)

    DO I = 1, NDATA
      DFILT(I) = FILT1(DTOT(I), XSAV, H, NFILT, NPT, 1)
    ENDDO

    CPUTD = SECNDS(CPUTD)

    ! TAKE FFT OF FILTER IMPULSE RESPONSE TO GET FILTER SPECTRUM
    ! AND CARRY OUT ADDITIONAL PREPARATIONS REQUIRED FOR PERFORMING
    ! FREQUENCY DOMAIN FILTERING.

    DO I = 1, NFILT
      F(I) = H(I)
    ENDDO
    ! Copy the filter coefficients

```

```

DO I = NFILT+1,NH      ! Pad out filter coefficients with zero
  F(I) = 0.0
ENDDO

DO I = NH+1,NHT2      ! Force second half of filter to zero
  F(I) = 0.0
ENDDO

CALL FAST(F,NHT2)      ! Get filter spectrum

DO I = NH+1,NHT2      ! Zero second half of R() prior to use
  R(I) = 0.0
ENDDO

! PERFORM FREQUENCY DOMAIN FILTERING BY LOOPING OVER
! SUCCESSIVE BLOCKS OF DATA

IF (JMOD(NDATA,NH).EQ.0) THEN
  NBLOCKS = NDATA/NH
  IPART = 0
ELSE
  NBLOCKS = NDATA/NH + 1
  IPART = 1
ENDIF

CPUFD = SECNDS(0.0)

DO IB = 1,NBLOCKS
  IPOS = (IB-1)*NH
  ! Transfer data for each block
  DO I = 1,NH
    R(I) = DTOT(IPOS+I)
  ENDDO
  ! Filter block of data
  CALL RFILT(R,F,WORK1,NH)
  ! Save filtered data
  DO I = 1,NH
    DFIL(IPOS+I) = R(I)
  ENDDO
ENDDO

CPUFD = SECNDS(CPUFD)

! WRITE OUT RESULTS TO A FILE

WRITE(*,'(/,1X,A,/)')
& 'WRITING FILTERED SEQUENCES TO FILE TESTFILT.OUT'

OPEN(UNIT=1,FILE='TESTFILT.OUT',STATUS='UNKNOWN',
& CARRIAGECONTROL='LIST')

WRITE(1,'(/,1X,A,1X,A)') 'FILTER COEFFICIENT FILE      ',FILENAME
WRITE(1,'(/,1X,A,I5,/)') 'NO. OF FILTER COEFFICIENTS :',NFILT
WRITE(1,'(1X,A,6F7.1)') 'FREQUENCIES IN 1st STOPBAND:',
& (FSB1(I),I=1,NSB1)
WRITE(1,'(1X,A,6F7.1)') 'FREQUENCIES IN PASSBAND      :',
& (FPB(I),I=1,NPB)
WRITE(1,'(1X,A,6F7.1)') 'FREQUENCIES IN 2nd STOPBAND:',
& (FSB2(I),I=1,NSB2)
WRITE(1,'(2(/,1X,A,F10.3,A) )')
& 'CPU TIME FOR FREQ DOMAIN FILTERING = ',CPUFD,' secs',
& 'CPU TIME FOR TIME DOMAIN FILTERING = ',CPUTD,' secs'
WRITE(1,'(/,1X,A,/,1X,A)')
& 'INDEX      TIME      X_ORIG      X_PB      X_FIL TIME X_FIL FREQ',
& '-----'

DO I = 1,NDATA
  WRITE(1,'(1X,I5,3F11.4,2F11.5)')
& I,I*DT,DTOT(I),DPB(I),DFIL(I),DFILT(I)
ENDDO

CLOSE(UNIT=1)

STOP
END

```

APPENDIX E

A Fortran Program For Studying The Effects of FIR Bandpass Filtering on Pulse-Type Transients

The following Fortran program can be used to study the effects of FIR bandpass filtering on a simulated pulse-like transient that is typical of vibration responses on the F/A-18.

CC

```

PROGRAM CHECKQ

IMPLICIT NONE

INTEGER NFFT,NFFT2,NPTS,NFAST
REAL SRATE,PI,TWOPI,DT

PARAMETER ( NFFT = 2048 )
PARAMETER ( NFFT2 = 2*NFFT )
PARAMETER ( NFAST = NFFT*2+2 )
PARAMETER ( NPTS = 3000 )
PARAMETER ( SRATE = 606.06 )
PARAMETER ( DT = 1.0/SRATE )
PARAMETER ( PI = 3.141592654 )
PARAMETER ( TWOPI = 2.0*PI )

REAL XSAV(NFFT2),H(NFFT)
REAL XDATA(NPTS),XFILT(NPTS),WINDOW(NPTS)
REAL FFTFILT(NFAST),FFTDATA(NFAST)
REAL FSB1(10),FPB(10),FSB2(10)
REAL XSB1,XSB2,XPB,OMEGA,GAMMA
REAL MagFac,PSDfac,NoiseBW,MFFTDATA,MFFTFILT,FREQ,T
INTEGER NFILT,INIT,IFILTER,I,J,N,NSB1,NSB2,NPB,NWIN
INTEGER WindowCode,IPOS,IOUT

CHARACTER ASCII1*4,ASCII2*15,ASCII3*4
CHARACTER FILENAME*20,CHOICE*20,WindowDesc*80

REAL FILT1,COS,SIN,FLOAT,CABS,EXP
INTEGER LENSTR
COMPLEX CMLPX

IFILTER = 0

DO WHILE (IFILTER.EQ.0)
  WRITE(*,*)
  WRITE(*,*) 'PROGRAM TO TEST TRANSIENT RESPONSE OF BANDPASS '//
6   'FILTER DESIGNS'
  WRITE(*,*) '-----'//
6   '-----'
  WRITE(*,*) ' 1 = 10 HZ TO 20 HZ BANDPASS FILTER'
  WRITE(*,*) ' 2 = 32 HZ TO 52 HZ BANDPASS FILTER'
  WRITE(*,*)
  WRITE(*,'(1X,A,$)') 'INPUT YOUR SELECTION: '
  READ(*,*) IFILTER
  IF (IFILTER.NE.1 .AND. IFILTER.NE.2) IFILTER = 0
ENDDO

WindowCode = 0

DO WHILE (WindowCode.EQ.0)
  WRITE(*,*)
  WRITE(*,*) 'SELECT WINDOW TO BE APPLIED TO TRANSIENT'
  WRITE(*,*) '-----'
  WRITE(*,*) ' 1 = SINE SQUARED WINDOW'
  WRITE(*,*) ' 2 = EXPONENTIAL WINDOW'
  WRITE(*,*) ' 3 = UNIFORM WINDOW'
  WRITE(*,*)
  WRITE(*,'(1X,A,$)') 'INPUT YOUR SELECTION: '
  READ(*,*) WindowCode

```

```

      IF (WindowCode.LT.1 .OR. WindowCode.GT.3) WindowCode = 0
ENDDO

IF (IFILTER.EQ.1) THEN
  FILENAME = 'BP15.OUT'
  NSB1     = 0
  NPB      = 1
  NSB2     = 0
  FPB(1)   = 15.0
  NWIN     = 627
ELSE IF (IFILTER.EQ.2) THEN
  FILENAME = 'BP45.OUT'
  NSB1     = 0
  NPB      = 1
  NSB2     = 0
  FPB(1)   = 45.0
  NWIN     = 613
ENDIF

! READ FILTER COEFFICIENTS

CALL READH(NFILT,H,FILENAME,1)

! OPEN FILE FOR WRITING FILTER OUTPUT

WRITE(*, '(/,1X,A,/)' )
& 'WRITING RESULTS TO FILE CHECKQ.OUT'

IOUT = 1

OPEN(UNIT=IOUT,FILE='CHECKQ.OUT',STATUS='UNKNOWN',
& CARRIAGECONTROL='LIST')

WRITE(IOUT,'(/,1X,2A,/)' ) 'FILTER FILE IS ',FILENAME
WRITE(IOUT,'(1X,A,6F7.1)' ) 'FREQUENCIES IN 1st STOPBAND:',
& (FSB1(I),I=1,NSB1)
WRITE(IOUT,'(1X,A,6F7.1)' ) 'FREQUENCIES IN PASSBAND      :',
& (FPB(I),I=1,NPB)
WRITE(IOUT,'(1X,A,6F7.1)' ) 'FREQUENCIES IN 2nd STOPBAND:',
& (FSB2(I),I=1,NSB2)
WRITE(IOUT,*)

! Set up data points for a cosine wave, ensuring that the unity
! value occurs at the NWIN/2+1 point, which is the centre of the
! transient that we are trying to simulate.
!
! NWIN must be an odd number.

DO I = 1,NWIN
  T = FLOAT(I-(NWIN/2+1))*DT
  XSB1 = 0.0
  XPB  = 0.0
  XSB2 = 0.0
  DO J = 1,NSB1
    XSB1 = XSB1 + COS(TWOPI*FSB1(J)*T)
  ENDDO
  DO J = 1,NPB
    XPB = XPB + COS(TWOPI*FPB(J)*T)
  ENDDO
  DO J = 1,NSB2
    XSB2 = XSB2 + COS(TWOPI*FSB2(J)*T)
  ENDDO
  XDATA(I) = XSB1 + XPB + XSB2
ENDDO

DO I = NWIN+1,NPTS
  XDATA(I) = 0.0
ENDDO

IF (WindowCode.EQ.1) THEN
  ! Set up a sine squared window which is similar to a
  ! Hanning window, except that both the end points are
  ! zero. It is created by squaring the values for the
  ! first half cycle of a sine wave.
  DO I = 1,NWIN
    WINDOW(I) = ( SIN(FLOAT(I-1)*Pi/FLOAT(NWIN-1)) )**2

```

```

      ENDDO
      WindowDesc = 'COSINE SEQUENCE WITH SINE SQUARED WINDOW'
    ELSE IF (WindowCode.EQ.2) THEN
      ! Set up an exponential window, increasing up to the centre of
      ! the window at NWIN/2+1, and decaying from then onwards. The
      ! damping is specified by GAMMA as % of critical.
      GAMMA = 0.03
      OMEGA = TWOPI*FPB(1)
      DO I = 1,NWIN
        T = FLOAT(I-(NWIN/2+1))*DT
        IF (I.LE.NWIN/2+1) THEN
          WINDOW(I) = EXP(+GAMMA*OMEGA*T)
        ELSE
          WINDOW(I) = EXP(-GAMMA*OMEGA*T)
        ENDIF
      ENDDO
      WindowDesc = 'COSINE SEQUENCE WITH EXPONENTIAL WINDOW'
    ELSE
      ! Set up uniform window
      DO I = 1,NWIN
        WINDOW(I) = 1.0
      ENDDO
      WindowDesc = 'COSINE SEQUENCE WITH UNIFORM WINDOW'
    ENDIF

    ! Apply the window to the NWIN points in the data sequence

    DO I = 1,NWIN
      XDATA(I) = XDATA(I)*WINDOW(I)
    ENDDO

    ! Initialize filter

    XFILT(1) = FILT1(XDATA(1),XSAV,H,NFILT,IPOS,0)

    ! Filter the sequence

    DO I = 1,NPTS
      XFILT(I) = FILT1(XDATA(I),XSAV,H,NFILT,IPOS,1)
    ENDDO

    ! Get FFT of raw and filtered data for comparison

    DO I = 1,NFFT
      FFTDATA(I) = XDATA(I)
      FFTFILT(I) = XFILT(I)
    ENDDO

    CALL FAST(FFTDATA,NFFT)
    CALL FAST(FFTFILT,NFFT)

    WRITE(IOUT,*)
    WRITE(IOUT,*) '=====
    WRITE(IOUT,*) 'FAST FOURIER TRANSFORM OF RAW AND FILTERED DATA'
    WRITE(IOUT,*) '          FOR CHECKING Q OF FIR FILTER'
    WRITE(IOUT,*) '=====
    WRITE(IOUT,*)
    WRITE(IOUT,*(1X,I4,A)) NFFT,' POINT FAST FOURIER TRANSFORM'
    WRITE(IOUT,*)
    WRITE(IOUT,*) WindowDesc(1:LENSTR(WindowDesc))
    WRITE(IOUT,*)
    WRITE(IOUT,*) '
    & WRITE(IOUT,*) '          UNFILTERED    FILTERED '//
    & WRITE(IOUT,*) '          UNFILTERED    FILTERED '
    & WRITE(IOUT,*) 'POINT    FREQUENCY    FFT MAG    FFT MAG '//
    & WRITE(IOUT,*) '          PSD M^2/Hz    PSD M^2/Hz'
    & WRITE(IOUT,*) '~~~~~ ~~~~~~ ~~~~~~ ~~~~~~'//
    & WRITE(IOUT,*) '          ~~~~~~ ~~~~~~'

    DO I = 1,NFFT/2+1
      J = 2*I
      FREQ = FLOAT(I-1)/FLOAT(NFFT)*SRATE
      MFFTDATA = ( CABS( CMPLX(FFTDATA(J-1),FFTDATA(J)) ) )
      MFFTFILT = ( CABS( CMPLX(FFTFILT(J-1),FFTFILT(J)) ) )
      ! Display FFT results as well as PSD results
      WRITE(IOUT,100) I,FREQ,MFFTDATA,MFFTFILT,
    & MFFTDATA**2/SRATE,MFFTFILT**2/SRATE

```


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