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ROCKET NOISE FILTERING SYSTEM USING DIGITAL FILTERS

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ABSTRACT

A set of digital filters is designed to filter rocket noise to various bandwidths. The filters are designed to have constant group delay and are implemented in software on a general purpose computer. The Parks-McClellan algorithm is used. Preliminary tests are performed to verify the design and implementation. An analog filter which was previously employed is also simulated.

SUMMARY

Acoustic data is collected by a field of sensors during launch. Although data is collected which contains valid data up to about 2 kHz, not all users require full bandwidth data. Filtering the data to remove spectral components which are not of interest results in savings in storage volume and processing time. For some applications it is important to maintain constant group delay. A set of digital filters has been designed and implemented which provide constant delay, very sharp roll off, and large stop band attenuation. Complete processing time for a typical field of sensors is less than 30 hours and could readily be reduced to less than 8 hours. A program was written to simulate a single analog filter which had previously been used for this application was written but not tested because of time limitations. The digital design eliminates data manipulation which is time consuming and potentially error prone while providing performance which is much superior to analog filtering.

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SECTION I

INTRODUCTION

1.1 THE CURRENT ROCKET NOISE DATA PROCESSING SYSTEM

The input to the current rocket noise filtering system is an analog signal proportional to absolute pressure. The present system utilizes a 5th order analog Butterworth filter having a 3 dB bandwidth of 2 kHz for its anti-aliasing filter. The data is then sampled at a rate of 9091 samples per second (sps) and converted to digital form by employing a 10 bit analog to digital (A/D) converter. This resolution is equivalent to 1024 cells and corresponds to a dynamic range of about 60 dB. The data is then stored in digital form on magnetic tapes with four decimal place precision. Four decimal place precision provides about 80 dB of dynamic range.

When it is necessary to reduce the bandwidth of the signal for some applications and generate samples at a lower sampling rate (down sampling), the digital data is played back and fed to a digital to analog (D/A) converter. The converter output is then filtered by an appropriate low pass filter and re-sampled at the lower rate. If the data is then to be processed digitally, it is re-digitized and stored.

One typical down sampling operation employs a 5th order analog Butterworth filter which has a bandwidth of 1 kHz. The 9091 sps digital data is pulled from tape, converted to analog form and filtered. It is then sampled at one-half of the original sampling rate, 4545.5 sps.

1.2 DISADVANTAGES OF CURRENT SYSTEM.

The data playback, reconstruction, and resampling process exhibits several shortcomings.

1. The process is time consuming since the original data tapes must be obtained and remounted. The process is repeated for each data pull.

2. The analog filter must be redesigned and rebuilt or, at a minimum, reconfigured whenever a new bandwidth is desired.

3. The Butterworth filter is not very selective so that either more aliasing noise must be accepted or higher sampling rates must be used for a given bandwidth.

4. The Butterworth filters intrinsically generate time delay distortion.

5. There are many opportunities for error in the procedure. Since the tape playback is repeated for each data pull, there is opportunity for the introduction of extraneous noise each time. The tape heads may be dirty or misaligned. The hardware may be incorrect. The wrong analog filter or the wrong sampling rate may be employed.

6. Errors may be insidious. Since they tend to add noise to the noise already present, they may be very difficult to detect. (Interestingly, there was noise in the first set of processed

data that was provided as test data. Figure 1-1a depicts histograms of the filtered and unfiltered data for sensor #1. The cell width for the histograms is 0.2 pounds per square inch (psi). Comparison of the original 9 ksps data with the down sampled data reveals that the mean of the filtered data differs from that of the unfiltered data. This indicates an error since the mean value should propagate through the filter without change. Histograms for sensor #2, shown in figure 1-1b, do not exhibit this anomaly.)

It is not necessary that the signal be reconstructed in analog form, refiltered, and re-sampled. Once the data has been acquired in digital form all processing can be done digitally. Once operational, digital processing will be reliable and repeatable and overcome the disadvantages cited above.

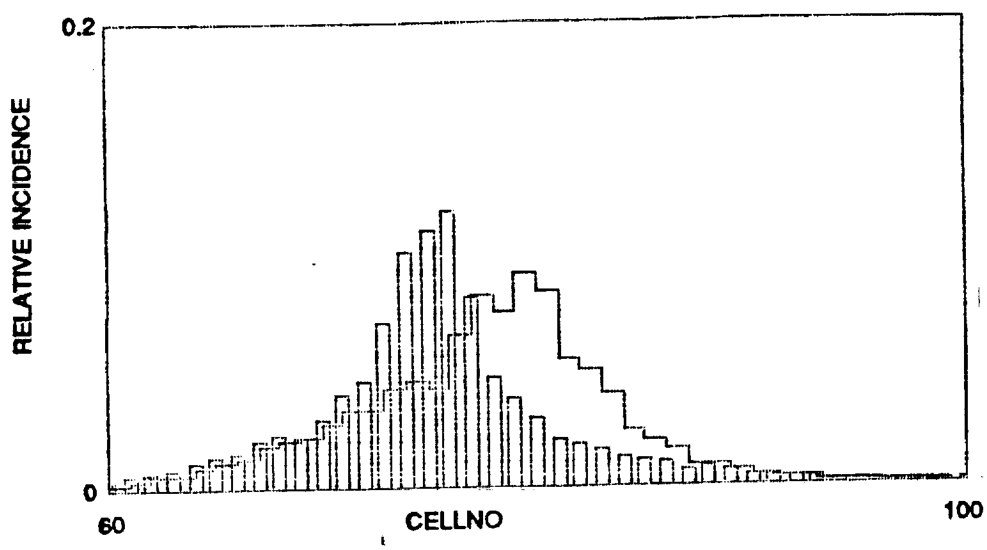


Figure 1-1a. Histogram of Filtered and Unfiltered Data for Sensor #1
Unfiltered Data drawn with vertical bars.
Cell width = 0.2 psi

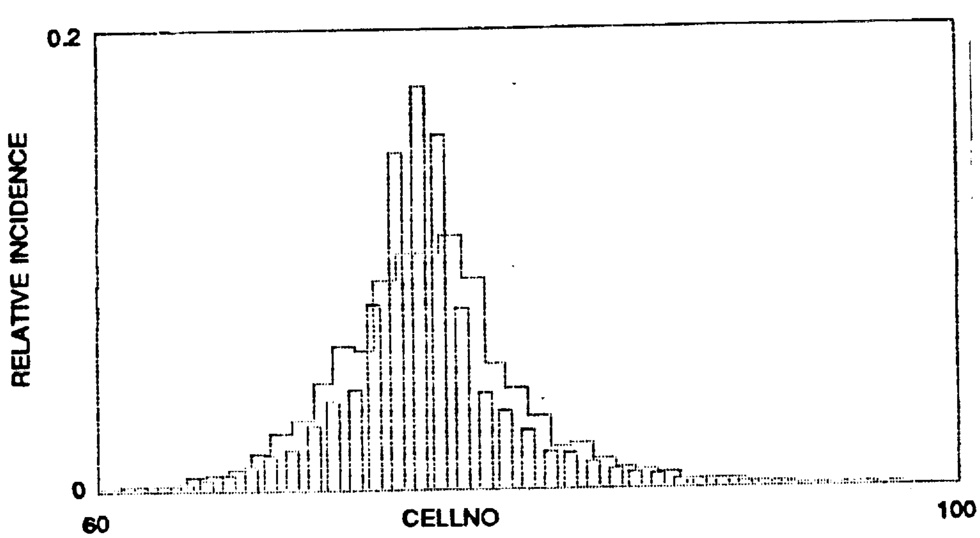


Figure 1-1b. Histogram of Filtered and Unfiltered Data for Sensor #2
Unfiltered data drawn with vertical bars.
Cell width = 0.2 psi

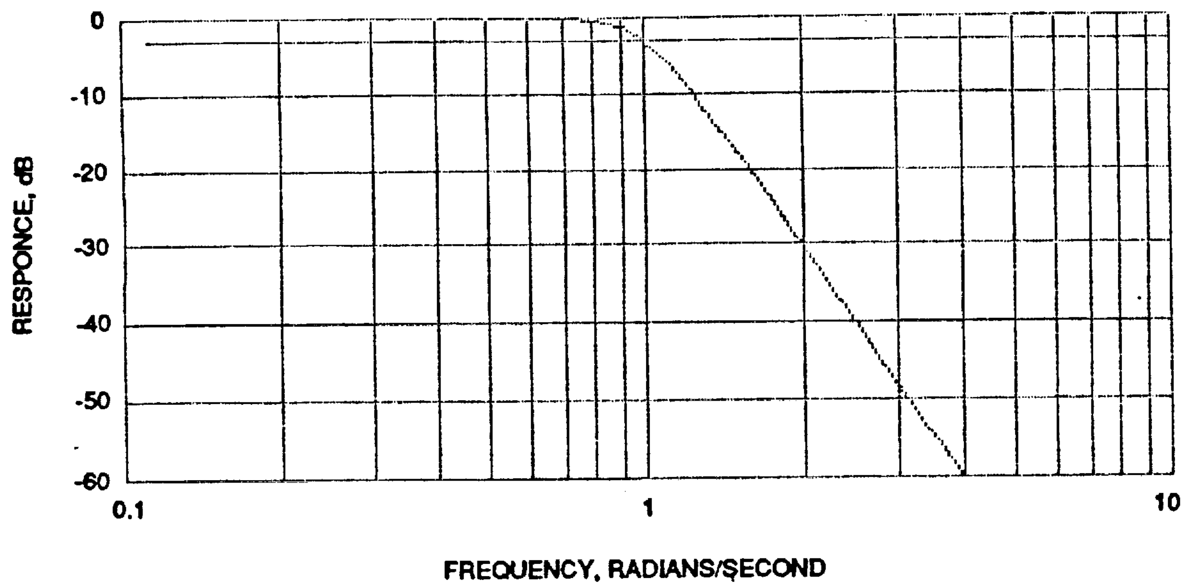


Figure 2-1. Bode Plot Response of 5th Order Butterworth Analog Filter
Normalized to a corner frequency of 1.
horizontal line is -3 dB.

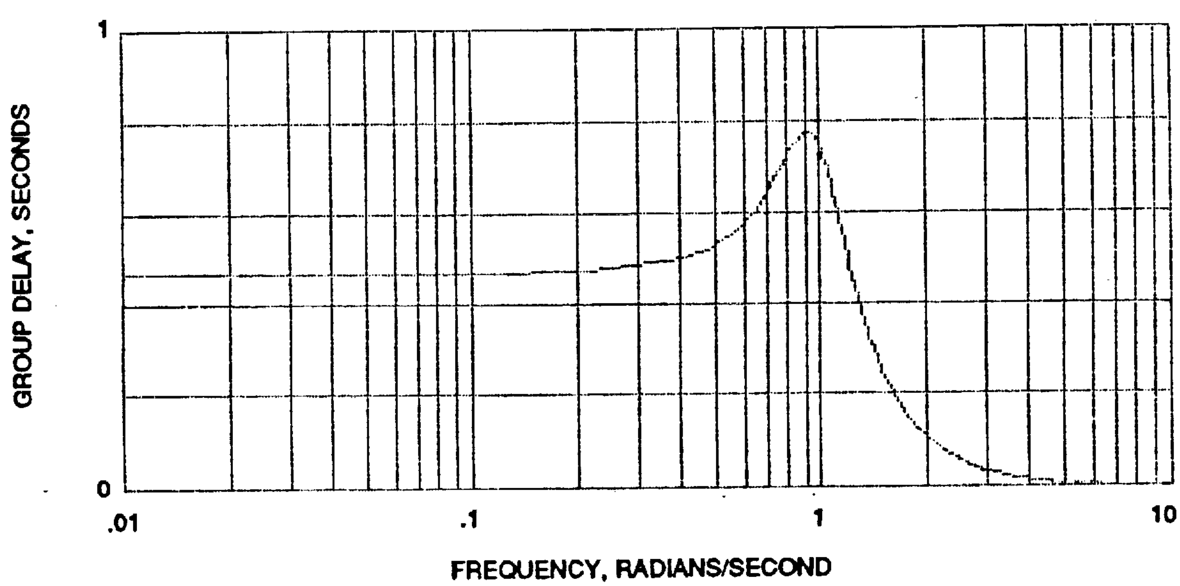


Figure 2-2. Group Delay of 5th Order Butterworth Filter
(for a 1 radian/second bandwidth filter)

SECTION II

ANALOG AND DIGITAL FILTERS

2.1 SIMULATION OF THE CURRENT ANALOG FILTER

It was decided to simulate the operation of the analog filter and the half rate down sampler which is often used. This would provide continuity with previous operations, and affords the opportunity to cross check results and detect errors. It was anticipated that this work could be done along with the digital filter development effort within the allotted ten week interval.

The analog filter used in the 2:1 down sampling operation has a 5th order Butterworth response, and a corner frequency of 1000 Hz. The poles of this filter lie on a circle of radius 2000*pi radians/second in the complex plane, with one pole on the negative real axis and angular spacings between successive poles of 36 degrees. The poles are therefore at -2000*pi, -1618*pi +/- j*1176*pi, and -618*pi +/- j*1902*pi. Figure 2-1 is a Bode plot of the 5th order Butterworth response, normalized to a cut off frequency of 1 radian/second. The group delay of this filter is shown in figure 2-2. The time delay is approximately constant for low frequencies (far below the cut off frequency) but rises rapidly to a maximum in the vicinity of the cut off frequency.

2.1.1 The Impulse Response of the 5th Order Butterworth Filter

The impulse response, h(t), for the filter will be used in the simulation of the analog filter, and may be determined by taking the inverse Laplace transform of the transfer function of the filter, which can be written in the form

$$B5(s) = \frac{Ao * B1 * B2}{(s + Ao) * (s^2 + A1 * s + B1) * (s^2 + A2 * s + B2)} \qquad (1)$$

where the name B5 was used to indicate that this is the transfer function of a 5th order Butterworth filter. A partial fraction expansion of this transfer function may be made in the form

$$B5(s) = \frac{Co}{(s+Ao)} + \frac{C1 * s + D1}{(s^2 + A1 * s + B1)} + \frac{C2 * s + D2}{(s^2 + A2 * s + B2)} \qquad (2)$$

The constants Co, C1, C2, D1, and D2 of this expansion may be determined by expressing the partial fraction expansion expression as a single term, over a common denominator and equating the coefficients of the powers of s. The resulting equations may be written in matrix form as

$$[C] * [U] = [K] \tag{4}$$

where the coefficient matrix [C] is

$$[C] = \begin{bmatrix} 1 & 1 & 1 & 0 & 0 \\ (A1+A2) & (A0+A2) & (A0+A1) & 1 & 1 \\ (B1+B2+A1A2) & (B2+A0A2) & (B1+A0A1) & (A0+A2) & (A0+A1) \\ (A1B2+A2B1) & A0B2 & A0B1 & (B2+A0A2) & (B1+A0A1) \\ B1B2 & 0 & 0 & A0B2 & A0B1 \end{bmatrix} \tag{5}$$

and the unknown matrix [U] and constant matrix [K] are

$$[U] = \begin{bmatrix} C0 \\ C1 \\ C2 \\ D1 \\ D2 \end{bmatrix} \qquad [K] = \begin{bmatrix} 0 \\ 0 \\ 0 \\ 0 \\ A0B1B2 \end{bmatrix} \tag{6}$$

The unknowns are therefore

$$[U] = [C]^{-1} * [K] \tag{7}$$

Equation (2) may be inverted on a term by term basis. The inverse transform of the first term of (2), defined as h1(t), is

$$h1(t) = C0 * \exp\{-A0*t\} \tag{8}$$

The second term of (2) is

$$\frac{C1 * s + D1}{(s^2 + A1 * s + B1)} \tag{9}$$

which may be inverted to yield h2(t)

$$h2(t) = \exp\{-A1*t/2\} * [C1 * \cos(wd1*t) + CF1 * \sin(wd1*t)] \quad (10)$$

where

$$wd1 = (B1 - A1^2/4)^{.5}$$

and

$$CF1 = (D1 - A1*C1/2)/wd1$$

Since the form of the third term of (2) is the same as that of the second term, its inverse transform, h3(t) has the same form as h2(t) except that 2 replaces 1 in the definition of the constants C1, D1, A1, B1, wd1, and CF1. The total impulse response is the sum of y1(t), y2(t) and y3(t).

The calculations to generate the constants and evaluate and graph h(t) have been performed in program B5HOFT.MCD. This program stores sample values of h(t) in a disk file. A related program is GB5CONST.MCD, which evaluates the constants A0, A1, A2, B1, B2, C0, C1, C2, D1, and D2 and defines the maximum significant duration time of h(t), MAXTH and stores this data in file B5H1K.CON for convenient use in the filtering program.

2.1.2 The Analog Filter Simulation Program

The analog filter simulation program B5A1K19.BAS simulates the 5th order Butterworth analog filter with a 1 kHz cut off frequency that is used as an anti-aliasing filter for the 2:1 down sampling system currently in use. This program may be used either to cross check the data produced by the traditional down sampling process or in lieu of it if so desired.

The program reads the constants which define the impulse response of the filter from file B5H1K.CON and defines the function H(T). It then loads the data from file ZMIPH1_9.DAT which contains zero mean pressure data which has been scaled by a factor of 100 and stored in zero mean integer form. This data originated from sensor #1 and was acquired at 9091 sps. (It was more efficient to work with scaled integer values of the pressure data.) The data is read into arrays which are stored in ephemeral memory (RAM) in arrays IDH19A and IDH19B. Two arrays were necessary because of array size limitations in Quick Basic 4.5. The filter output is estimated by forming a numerical approximation for the convolution of the input data with the impulse response for every multiple of the read out time interval, i.e. the reciprocal of the output sampling rate. There were 42,976 samples in the input file. The first 32,000 points are from array IDH19A, the rest from IDH19B. The progress of the program is reported to the console by printing the output point number and the time for points 1, 40, 50, 1000, and 2000. This information may then used to estimate the time to completion. The progress information may be omitted if so desired, but the savings in execution time will not be great. The filter output is then stored in file YF1910KA.DAT. This file is nominally half as large as the input file because of the 2:1 down sampling.

2.2 THE DIGITAL FILTER

Digital filters have the potential to outperform their analog counterparts in many respects. Since they may be implemented as computer programs they can be of relatively high order with essentially no increase in complexity, thus leading to steep selectivity skirts. Since no hardware is involved, they may be readily and quickly changed. They can be designed to have phase characteristics which are exactly linear. The group delay will then be constant for all frequencies. They are completely stable with respect to environmental factors such as temperature and humidity. Component aging is not a factor. Once operational, they are reliable. Should a system fail, it will typically fail completely so that there is no uncertainty with respect to the occurrence.

2.2.1 Selection of the Digital Filter Type

Digital filters may be classified as having either infinite impulse response (IIR) or finite impulse response (FIR). The output of an IIR filter may extend to infinity because samples of the output are fed back through the system. The impulse response of FIR filters must be zero after the last non-zero input has propagated through the system. The choice of the filter type depends upon the nature of the application and circumstance.

Although IIR filters may be unstable, FIR filters are always absolutely stable; with no feedback there is no possibility of unbounded oscillation. Closed form design equations for IIR filters exist for many filters, whereas there is no analogous set of design equations for FIR filters. For similar levels of performance, a FIR filter tends to be of higher order than an IIR filter. This leads to reduced hardware requirements and faster execution times for IIR implementations. A FIR filter may be designed to have exactly linear phase so that the time delay of the filter can be constant for all frequencies. A more extensive comparison of the relative differences between FIR and IIR filters is contained in reference 1.

Execution time is not of great consequence for our application since filtering need not be done on a real time basis. Also, since the filter will be implemented on a general purpose computer, hardware complexity is not a factor. The advantage of constant time delay afforded by FIR filters is highly desirable for our application; therefore our choice is to use a FIR filter.

2.2.2 Linear Phase FIR Filters

It can be shown that a sufficient condition for linear phase response is that the unit sample response of the system, h(n), be even symmetric about its midpoint. See, for example, reference 2.

2.2.3 Design Algorithms

Although no general closed form design algorithm exists for FIR filters, there are known design procedures. Impulse invariance techniques are the simple, easy to employ and allow translation of analog filter designs to digital designs, but exhibit aliasing problems and are not usually optimum. Modifications may include the use of weighting functions (windowing) to yield improved response. Bilinear transformation can also afford a means by which analog filters may be translated to digital filters. These eliminate the aliasing problem associated with the previous technique, but the mapping involved distorts the frequency scale. (Pre-warping can be employed to produce acceptable results in the case of filters having piecewise constant transfer functions.) The design techniques cited are derivatives of analog filter designs. As such, they lose much of the potential advantage of digital filters.

Techniques analogous to impulse invariance exist in the frequency domain. The unit sample response of a digital filter may be obtained by taking the inverse discrete transform of samples of the desired response in the frequency domain.

Direct approaches which do not rely on prior analog filter designs have also been developed. Consider the design of a low pass filter having equal ripple in the pass band and in the stop band. The parameters of interest are the order of the filter, the frequency of the upper edge of the pass band, the frequency of the lower edge of the stop band, the ripple in the pass band and the ripple in the stop band. These parameters are interrelated; they can not be chosen independently. Although the problem has been formulated with many choices for the independent variables, Parks and McClellan have developed the mathematical conditions and a computer program which employs iterative techniques to design linear phase filters when the order of the filter, the edge of the pass band and the edge of the stop band are given. Please refer to references 3 and 4. The program minimizes the maximum error. The resulting filters show nearly equal ripple throughout the band. Even high order filters may be designed relatively quickly, although it may be desirable to repeat the design process to minimize the width of the transition band.

2.2.4 Digital Filter Design

The performance of digital filters of various orders was explored empirically using the Parks and McClellan design algorithm. Filters of order 128 were selected and used exclusively because of their performance. Although such high order filters increase computation time, this was considered to be a relatively insignificant for this application; the rapid transition from pass band to stop band and large attenuation in the stop band were considered to be more desirable than reduced computation time.

Figure 2-3 indicates the topology for the filter, while figure 2-4 shows the performance of a filter designed to have a cut off frequency of 1 kHz, the same as the fifth order analog Butterworth filter used in the current down sampling process. At the corner frequency the analog filter is down by 3 dB, as for any order Butterworth filter. The digital filter output is down only a fraction of a dB at the corner frequency. The original analog filter response is down about 30 dB one octave above the cut off frequency; the digital filter is down about 92 dB for any frequency above 1.12 kHz.

2.2.5 Filter Bank Definition

It was decided that a set of filters be designed and implemented on a general purpose computer. Each filter output would produce a separate file. The set of bandwidths chosen for the filters was 2.0 kHz, 1.5 kHz, 1.0 kHz, 800 Hz, 600 Hz, 400 Hz, 300 Hz, 200 Hz, 150 Hz, 100 Hz, 60 Hz, 40 Hz, 30 Hz, 20 Hz, 15 Hz, and 10 Hz. The ratio of successive filter bandwidths was chosen to be no less than 0.6. The set of filter bandwidths selected is felt to be reasonably complete, although filters having different bandwidths may be designed.

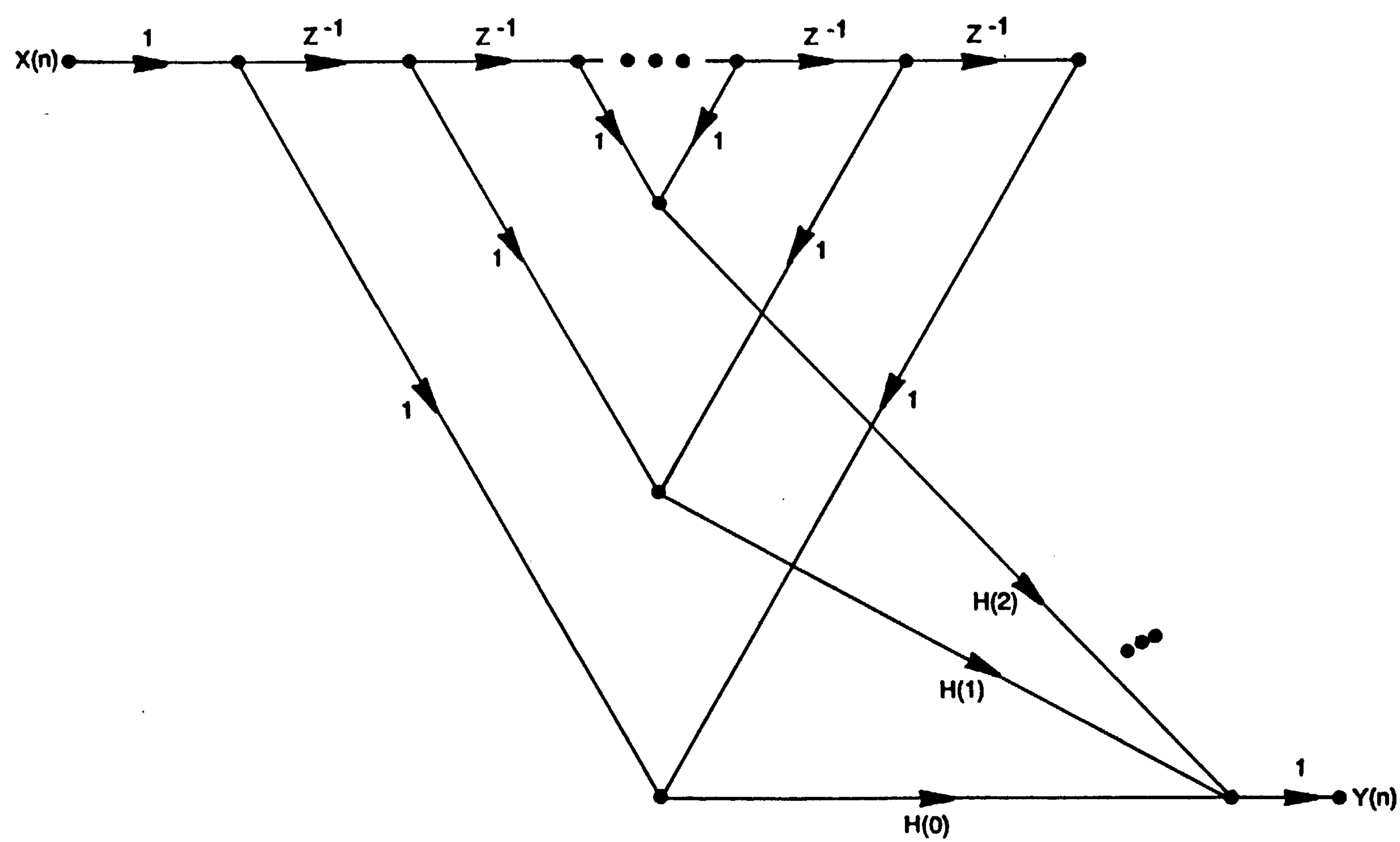


Figure 2-3. Topology of Digital Filter.

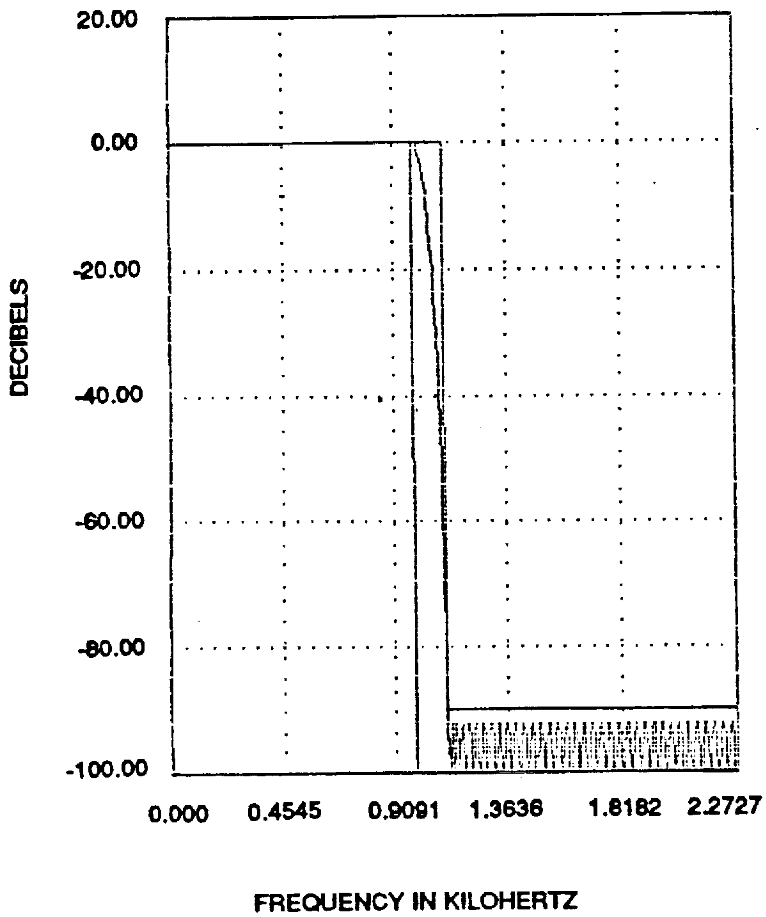


Figure 2-4. Performance of a 1 kHz Digital Filter.

Not all filters have been designed with the same sampling rate. As the spectral width of the data is reduced by filtering to less than one-fourth the input data sampling rate, it is possible to reduce the sampling rate by a factor of two and still meet the Nyquist sampling rate criterion. This is done whenever possible to reduce the volume of stored data. This also enables us to maintain the sharp selectivity skirt of the ensuing filter. Finally, the computation time is correspondingly reduced with no loss of information.

2.2.6 Block Diagram of Filter Bank and Designations

A block diagram of the filter bank which has been implemented is shown in figure 2-5. Low pass filters are designated by a lead L followed by a number indicating their bandwidth. Output files are designated by a lead Y followed by a number indicating the data bandwidth and a letter specifying their input sampling rate. The convention used to define the sampling rate is as follows:

N - 9.091 ksps	W - 1.136 ksps	E - .142 ksps
F - 4.546 ksps	H - .568 ksps	S - .071 ksps
T - 2.273 ksps	Q - .284 ksps	D - .036 ksps

The logic behind these designators is that the rates are nominally Nine, Four, Two, Won, Half, Quarter, one Eighth, one Sixteenth, and one thirty seconD ksps. The use of "Won" allows us to avoid the use of O which might be confusing. The author apologizes for the misuse of won, but the reduced probability of error warrants the potentail wrath of grammarians. The other rather odd usage is the D for one thirty second, but occurs because T had already been used to designate two ksps.

The input data from file ZMIPH1_9.DAT came from sensor #1 at a 9091 sps rate. (This has been rounded to 9 ksps in the figure.) It contains data which is the integer part of 100 times the original pressure data minus the mean value. No information has been lost by this process. The data is filtered by the 2 kHz wide low pass filter, and data is stored in output file Y2_0N. The same input data is also filtered by the 1.5 kHz wide low pass filter, but since this is less than one fourth the sample frequency, it is only necessary to store alternate samples. The two to one down sampling operation is indicated in the diagram by a circle with an arrow pointing downward. Similar logic was used to define the rest of the filters and files.

2.2.7 Digital Filter Definition

The sixteen filters cited in the previous section may be defined by citing their unit sample responses. The array defining the unit sample responses of the filters is in disk file available upon request.

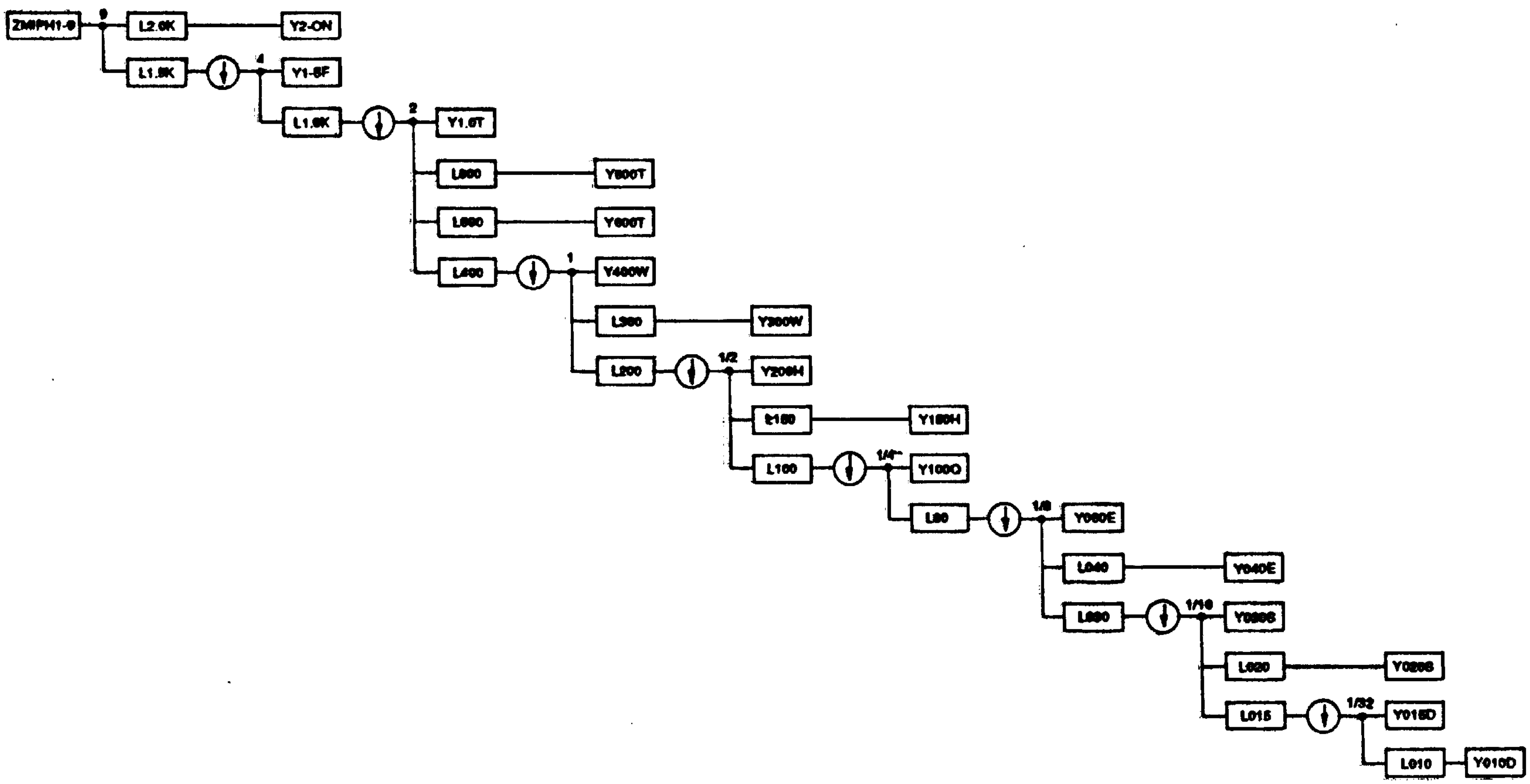


Figure 2-5. Block Diagram of Digital Filter Bank.

SECTION III

RESULTS AND DISCUSSION

3.1 PERFORMANCE OF FILTERS

The performance of each of the sixteen filters is similar to that of the 1 kHz digital filter which was cited previously, so Bode plots for each individual filter have not been included. Table 3-1, below, summarizes the performance of each filter more compactly. The column labeled Bandwidth is the width of the passband of the filter. The Stopband is the frequency of the lower edge of the stopband. The % BW is the width of the transition band measured as a percentage of the pass band width. Fsample is the sampling rate of the input data. Ripple is the peak value occurring within the passband. Attenuation is the minimum value of loss occurring within the stop band.

Bandwidth (Hz)	Stopband (Hz)	% BW	Fsample (Hz)	Ripple (dB)	Atten. (dB)
2000	2400	20	9091	0.005	> 100
1500	1800	20	9091	0.05	> 100
1000	1120	12	4546	0.15	92
800	860	8	2273	0.14	92
600	660	10	2273	0.15	92
400	460	15	2273	0.15	92
300	330	10	1136	0.15	92
200	230	15	1136	0.15	92
150	165	10	568	0.15	92
100	115	15	568	0.15	92
60	68	13	284	0.15	92
40	47	19	284	0.15	91
30	34	12	142	0.19	90
20	22	9	71	0.18	90
15	17	12	71	0.19	90
10	11	9	36	0.18	90

Table 3-1. Summary of Filter Performance

3.2 DIGITAL FILTER BANK IMPLEMENTATION

The digital filter band program is named DFILTER. The program was written in BASIC because this language is familiar and ubiquitous. Current versions of the language have extensive capability and very convenient debug and test features. A brief discussion of the operation of the program is included in this section of the report.

3.2.1 The Digital Filtering Program

DFILTER first reads the constants which define the unit sample response of each of the sixteen digital filters into a two dimensional array, H(K,FILTNO). The source data is contained in file HKFILTNO.FDT which was produced by a utility program which was written to extract the values of H(K) from the design program DFDP. (DFDP, Digital Filter Design Program, is part of a digital filter design program which is commercially available.)

After initializing the output point number (OUTPTNO&), and defining the length of the filter (FLTLEN, which is 128 for all the filters at present), the filter length plus 1 (FLPLUS1), the filter length divided by 2 (FLDIVBY2), and the number of filters (NOOFFILT), the input array (X) is dimensioned to the length of the filter and the output array (Y) is dimensioned to the number of filters. The constants FLPLUS1 and FLDIVBY2 are evaluated outside the ensuing loops to eliminate unnecessary repetitive calculation.

The file to be processed is then opened for input. In the listing, the file ' being processed, ZMIPH1_9.DAT, is from sensor #1. File ZMIPH2_9.DAT or any other data file sampled at 9091 sps could be processed. The filters having cut off frequencies of 2.0 kHz and 1.5 kHz are designed to operate at this sampling rate. The files for the output (Y2_0N.DAT and Y1_5F.DAT) are then opened.

Next the input array (X) is loaded with the first 128 samples of the input file. Pairs of input values which share the same unit sample weighting are summed, multiplied by the appropriate value of H(K) and accumulated for the 2.0 kHz cut off frequency filter which is defined to be filter #1. Since the 1.5 kHz filter, #2, also operates on input data having the same sample rate, similar processing is done for this filter. However, the spectral content of its output is low enough that it may be down sampled; this is done by calculating its output for only odd numbered output points. This also reduces computation time. Having completed calculations for the filter outputs, the input array is shifted by one sample and a new input sample is read into the first cell of the input array. This process is repeated until the input data array is exhausted. The filtering process is then repeated for the remaining filters with the source file updated to provide data at the correct sampling rate.

The last portion of the program evaluates the mean squared value of the data in the output arrays. This is the AC power of the input array since the mean value has already been subtracted. The value is scaled by 10,000 because of the conversion to integer format, which included multiplication by a factor of 100.

At present, any input file will produce the output files cited. It is suggested that the output files be renamed so that they are not inadvertently overwritten. The primary names of the output files are only 5 places long, so there are 3 additional places which could be used to designate the sensor number and launch.

3.2.2 Initial Filter Program Tests

Preliminary tests have been run on the digital filter bank program to verify its operation. Tests included sinusoidal and noise excitation.

3.2.2.1 Sinusoidal Excitation Test

Sums of unit sine waves at various frequencies were sampled at 9091 sps and 100,000 data points were stored. The frequencies were chosen so that one additional sine wave would be included in each successively wider filter. A few sine waves were then added near the corner frequency of the widest (2.0 kHz) filter and above its cut off frequency. The power calculations should then show a decrease of 1/2 for each successively narrower band, except that the widest filter (2 kHz) would partially respond to the additional sine waves around its cut off frequency. Specifically, the unit sine waves were at frequencies of 5, 12, 18, 28, 38, 58, 90, 140, 190, 290, 390, 590, 790, 990, 1490, 1990, 2100, 2200, 2500, and 3000 Hz. Table 3-2 contains data for theoretical and calculated filter output power when the filter bank is excited by the sum of these sine waves. The ripple in the pass band has been ignored.

File Source	Theoretical Power	Calculated Power	Percent Error
Input File	10.00	9.99941	-0.006
2.0 kHz	8.35	8.33643	-0.163
1.5 kHz	7.50	7.50658	0.088
1.0 kHz	7.00	7.02102	0.300
800 Hz	6.50	6.52702	0.416
600 Hz	6.00	6.02446	0.408
400 Hz	5.50	5.55879	1.069
300 Hz	5.00	5.04134	0.827
200 Hz	4.50	4.50858	0.191
150 Hz	4.00	4.07841	1.960
100 Hz	3.50	3.49527	-0.135
60 Hz	3.00	2.93645	-2.118
40 Hz	2.50	2.47886	-0.846
30 Hz	2.00	1.99582	-0.209
20 Hz	1.50	1.49261	-0.493
15 Hz	1.00	1.01627	1.627
10 Hz	0.50	0.99467	98.934

Table 3-2. Comparison of Theoretical and Calculated Power Outputs

The theoretical power calculation includes observable effects of the components at 2.1 and 2.2 kHz in the output of the 2.0 kHz wide filter. Despite ignoring the ripple in the pass band, the error build up from a large number of calculations and variations due to sampling, the maximum error is only 2.118% and the mean absolute error is about .68 % excluding the results from the 10 Hz filter. It is felt that there is an error in this filter. The origin of this error was not pursued since this is the least important filter and because of time limitations.

3.2.2 White Noise Excitation Test

An alternate test was also performed to increase confidence in the digital filtering program. The pseudo-random number generator available in the BASIC language was used to provide samples which were interpreted as originating from bandlimited white noise. The bandwidth of the source was assumed to be 4545.5 Hz and the sample rate was defined to be 9091 in order to be compatible with the filter sample rate requirement. The amplitude of the samples was scaled to provide a noise power spectral density of 1 volt squared per Hz, and 100,000 samples were generated and stored in a file.

The file of noise samples was then fed to the filter bank program as input data. The power of the data in the input file and in the data of each output file were estimated. The power in the output files should be proportional to the equivalent noise bandwidth. Since the spectral density was scaled to unity, we expect the output power to be numerically equal to the equivalent noise bandwidth of the filter in a statistical sense. The equivalent noise bandwidth of the filters was not calculated, but may reasonably expected to be limited to a value greater than the pass band width but less than the edge of the stop band. These limits are only valid in a statistical sense, and only one random noise sample file was run. The number of samples in outputs of the wider filters is quite large, and the variability of the power should be relatively small. Table 3-3 is a tabulation of the power in the input file, the power in each of filter outputs, and the limits cited.

File Name	Power Estimated	Power Limits Lower	Power Limits Upper
INPUT	4553.666	4546	4546
Y2_0NWN	2126.869	2000	2400
Y1_5FWN	1583.643	1500	1800
Y1_0TWN	1032.652	1000	1120
Y800TWN	820.728	800	860
Y600TWN	623.858	600	660
Y400WWN	420.640	400	460
Y300WWN	316.196	300	330
Y200HWN	214.496	200	230
Y150HWN	156.262	150	165
Y100QWN	106.328	100	115
Y060EWN	61.536	60	68
Y040EWN	40.286	40	47
Y030SWN	31.766	30	34
Y020SWN	19.942	20	22
Y015DWN	14.566	15	17
Y010DWN	13.660	10	11

Table 3-3. Power in Volts squared for the input and output files.

Agreement is excellent for nearly all the files, although the power in the filters having bandwidths of 20 Hz and 15 Hz is very slightly low, this may be a consequence of statistical variation and the relatively small sample size available at these low frequencies. The only questionable result is once again that from the 10 Hz filter. Its output appears to be unduly high. This result again makes this filter implementation suspect.

3.3 EXECUTION TIME AND DATA VOLUME

A complete run, including filter output power calculations in BASIC, for the white noise test sample, which consisted of 100,000 points of data, took less than three hours to complete on a computer having a 80386 CPU and a 80387 co-processor chip running at a speed of 16 MHz. This volume of data is equivalent to the output of one sensor for 11 seconds, so the execution time is about .27 hours/sensor-second. A set of 10 sensors collecting data for 11 seconds each would lead to a total processing time of about 30 hours under the conditions cited. Since this is a non-recurrent operation for a given launch, the current execution time may be acceptable.

3.3.1 Reduction of Execution Time

Run time could be reduced by using a faster machine, say a 33 MHz system, thus speeding execution by a factor of 2. It would be possible to use one of the new 80486 systems, which are reputed to be 2 to 4 times faster than the 80386 machines. The program could also be run on a mini, mainframe, or other faster machines. A compiled language program would also execute more rapidly. Additionally, the program could be modified to reduce computation time. For example, the shift register operation, which mimics a hardware implemented shift register could be replaced by a functionally equivalent system in which the data is loaded into an array in RAM and accessed by pointers. This implementation would most likely be quicker. It is also possible that transform techniques may be faster.

3.3.2 Data Volume

The data volume may be reduced with no loss of information by storing only the sampling interval and a sequential set of pressure samples. The pressure samples may be stored in integer format rather than floating point. Data compression techniques, which are currently in use, should be continued for archival purposes.

SECTION 4

CONCLUSIONS

A filter bank of 15 digital filters has been designed and implemented in software for this project and appears to be functioning correctly, although additional verification work would further increase confidence.

The all-digital filtering program is far more versatile than the previous system. It is much more extensive.

Should different applications require different bandwidths, a new set of digital filters could be quickly, easily, and inexpensively designed and implemented.

The performance attained by the digital filters far exceeds that of their analog predecessor both in roll off rate and attenuation. Aliasing errors are correspondingly reduced, assuming the adequacy of the analog anti-aliasing filter.

The time delay for the digital filters is constant for all frequencies, so there is no distortion caused by relative time shifts between the spectral components of a signal.

The time to process 11 seconds of data from 10 sensors may be filtered and the power in the filter outputs calculated in less than 30 hours. The processing time could be easily reduced to less than 8 hours.

Once operational, the digital filtering system is reliable and error free. No analog signal reconstruction, filtering and resampling operations are necessary, thus eliminating many opportunities for error.

The volume of data could be further reduced and subsequent processing time decreased if the sampling rate were made close to the Nyquist rate for each filter bandwidth. This would require non-integer changes in the sampling frequency. It is possible, through a process of upsampling and down sampling, to produce sampling rates which related by any rational number.

The analog filter simulation was written but not verified due to time limitations. It was a secondary goal, intended to provide cross comparison and verification of the current processing system. It was therefore given lower priority than the digital effort.

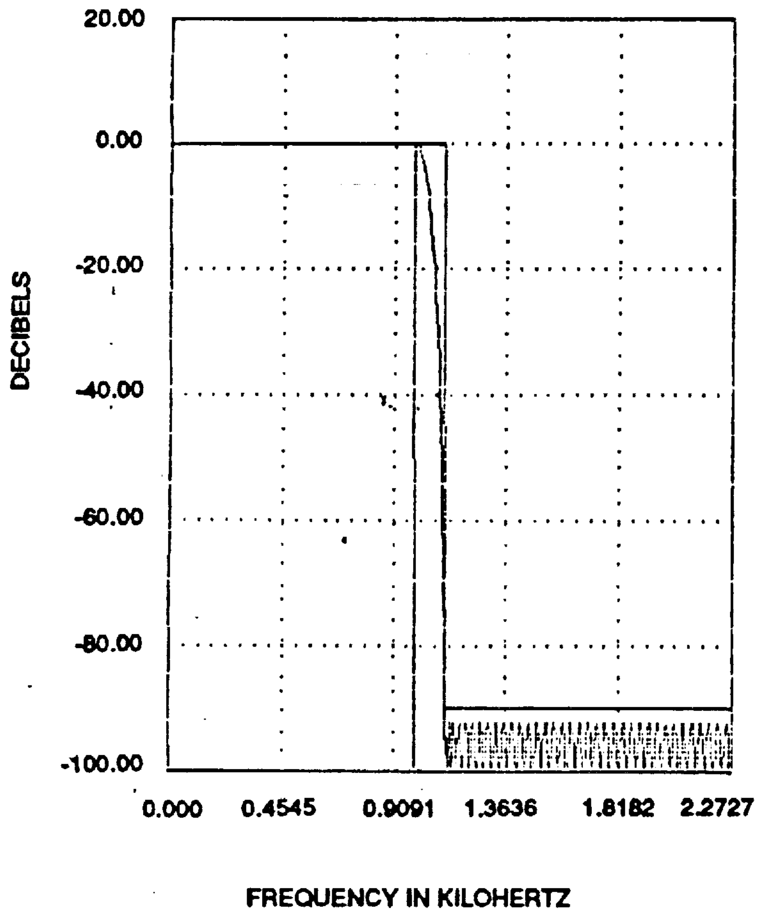
References

1. L. R. Rabiner, J. F. Kaiser, O. Herrmann, and M. T. Dolan, "Some Comparisons between FIR and IIR Digital Filters," Bell System Technical Journal, Volume 53, Number 2, February, 1974, pages 305 - 331.

2. Alan V. Oppenheim and Ronald W. Schafer, "Digital Signal Processing," Prentice-Hall, 1975.

3. T. W. Parks and J. H. McClellan, "Chebyshev Approximation for Non-recursive Digital Filters with Linear Phase," IEEE Transactions Circuit Theory, Volume CT-19, March, 1972, pages 189 - 194.

4. T. W. Parks and J. H. McClellan, "A Program for the Design of Linear Phase Finite Impulse Response Filters," IEEE Transactions Audio Electroacoustics, Volume AU-20, Number 3, August, 1972, pages 195 - 199.



Log Magnitude Response

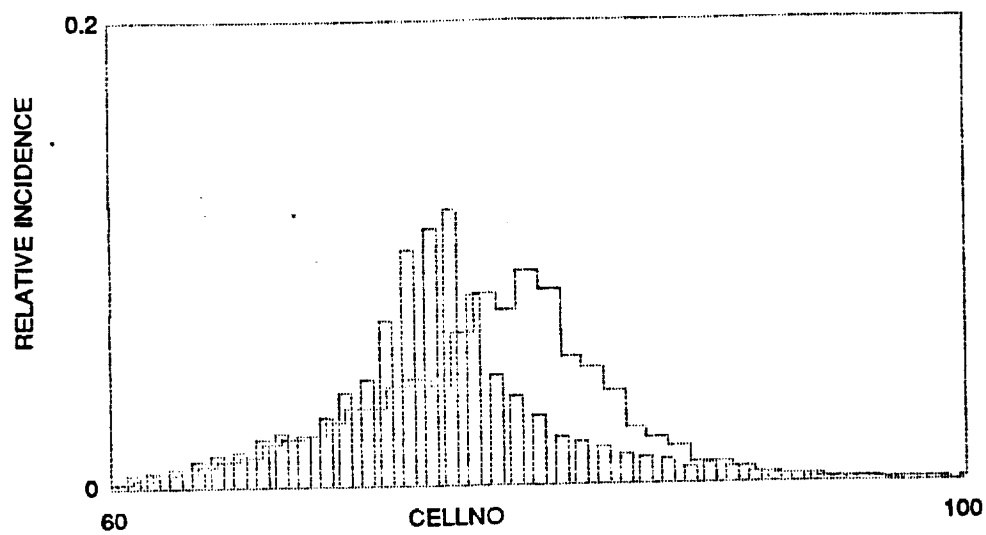


Figure 1-1a. Histogram of Filtered and Unfiltered Data for Sensor #1
Unfiltered Data drawn with vertical bars.
Cell width = 0.2 psi

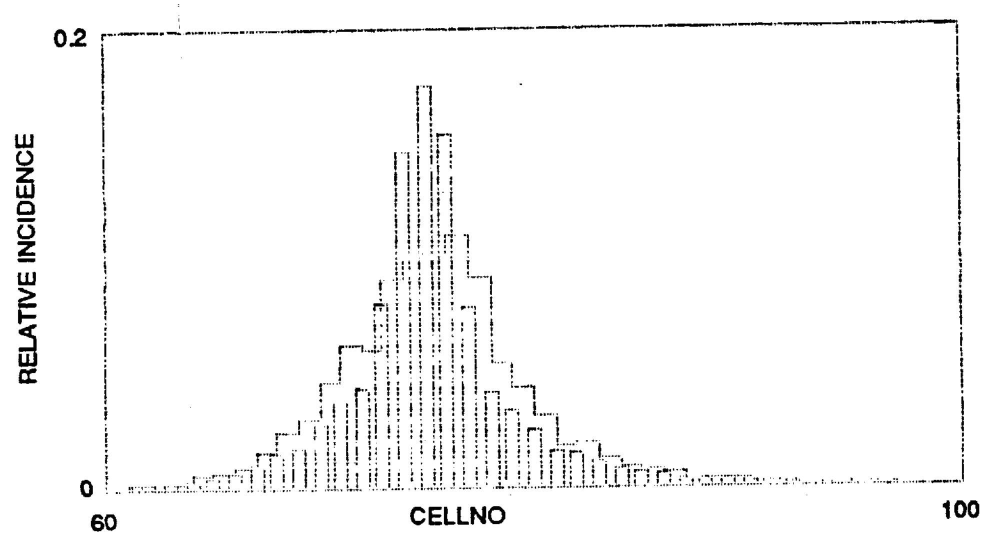


Figure 1-1b. Histogram of Filtered and Unfiltered Data for Sensor #2
Unfiltered data drawn with vertical bars.
Cell width = 0.2 psi

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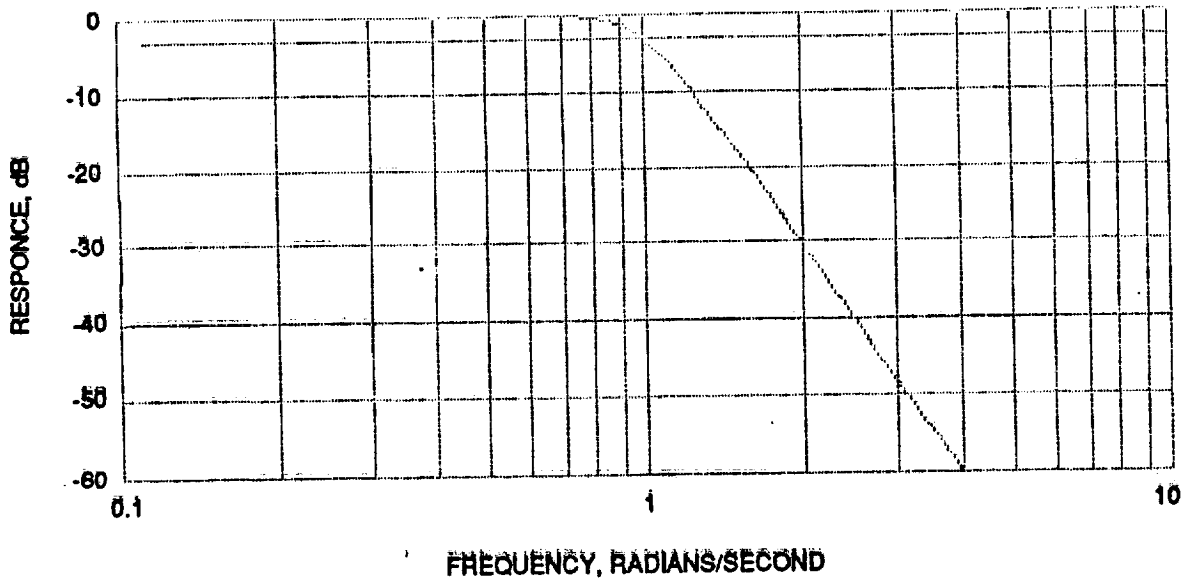

Figure 2-1. Bode Plot Response of 5th Order Butterworth Analog Filter
Normalized to a corner frequency of 1.
horizontal line is -3 dB.

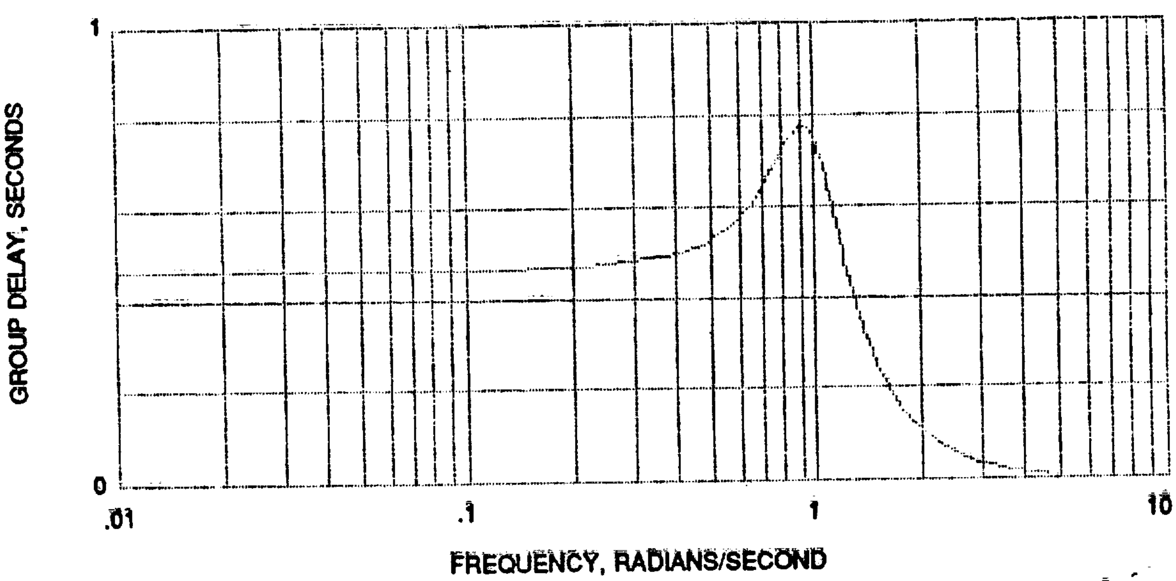

Figure 2-2. Group Delay of 5th Order Butterworth Filter
(for a 1 radian/second bandwidth filter)

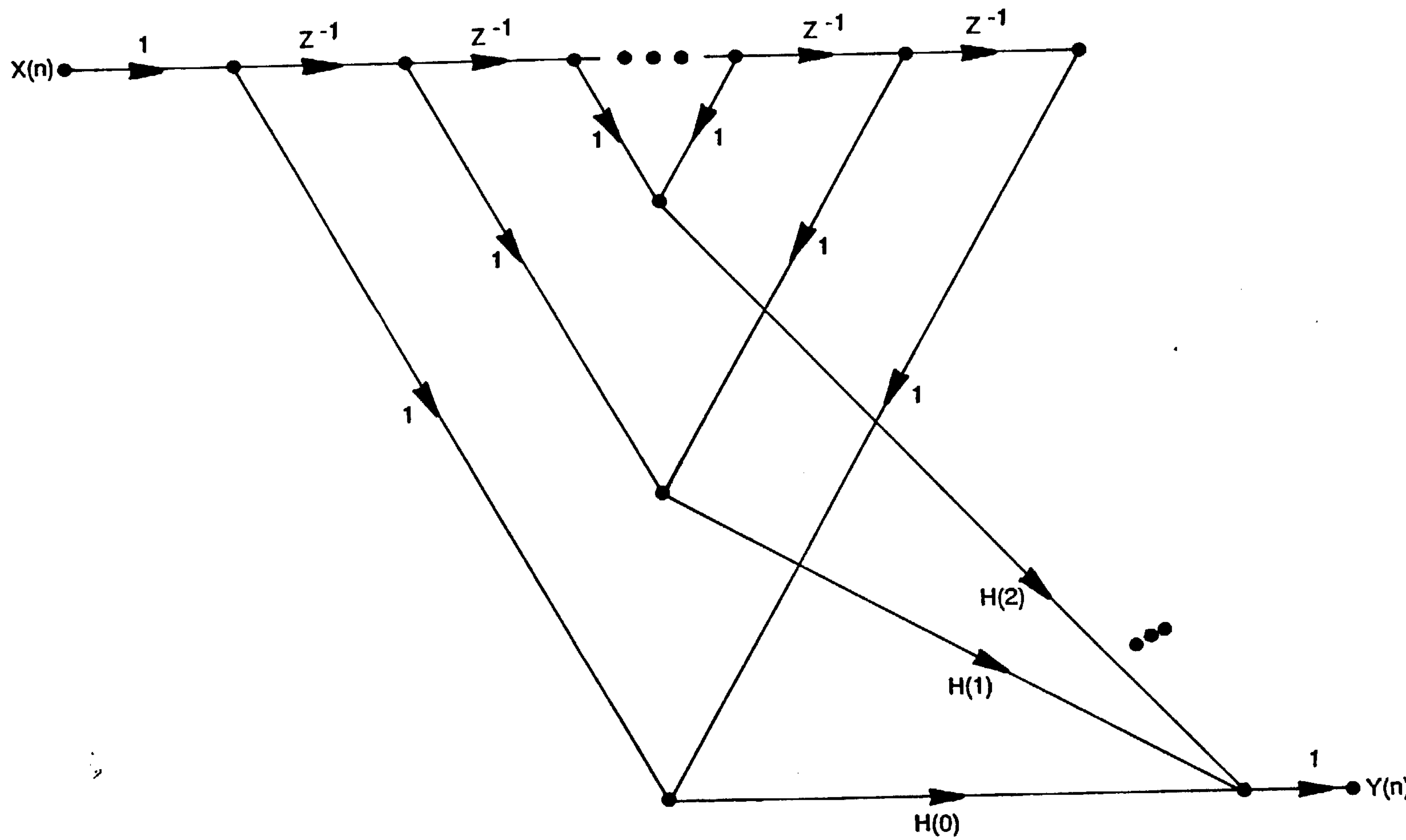

Figure 2-3. Topology of Digital Filter.

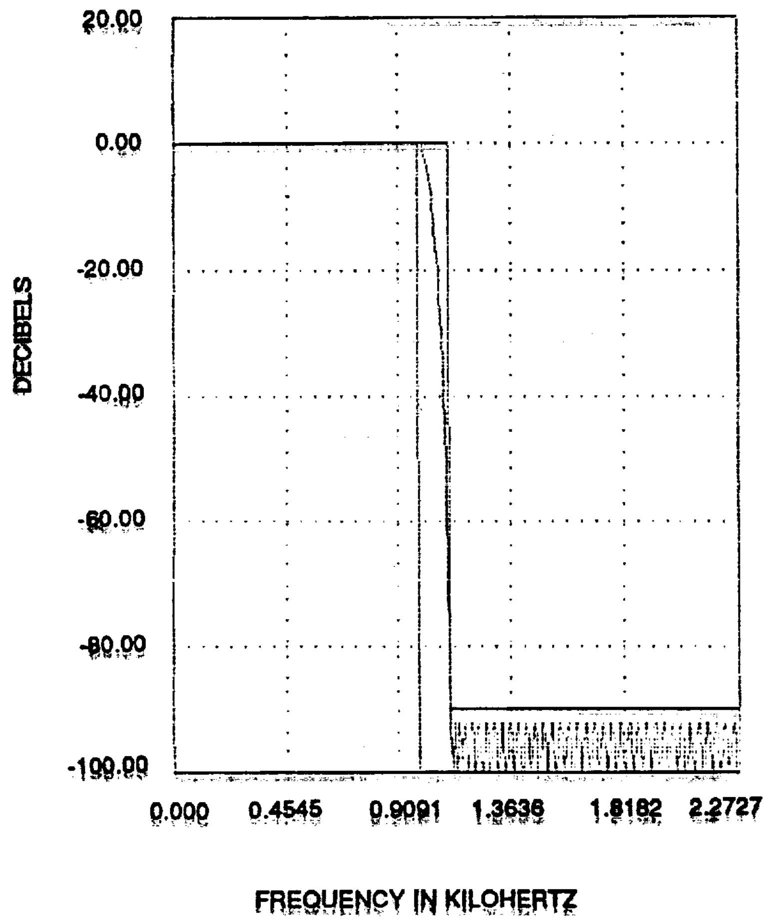

Figure 2-4. Performance of a 1 kHz Digital Filter.

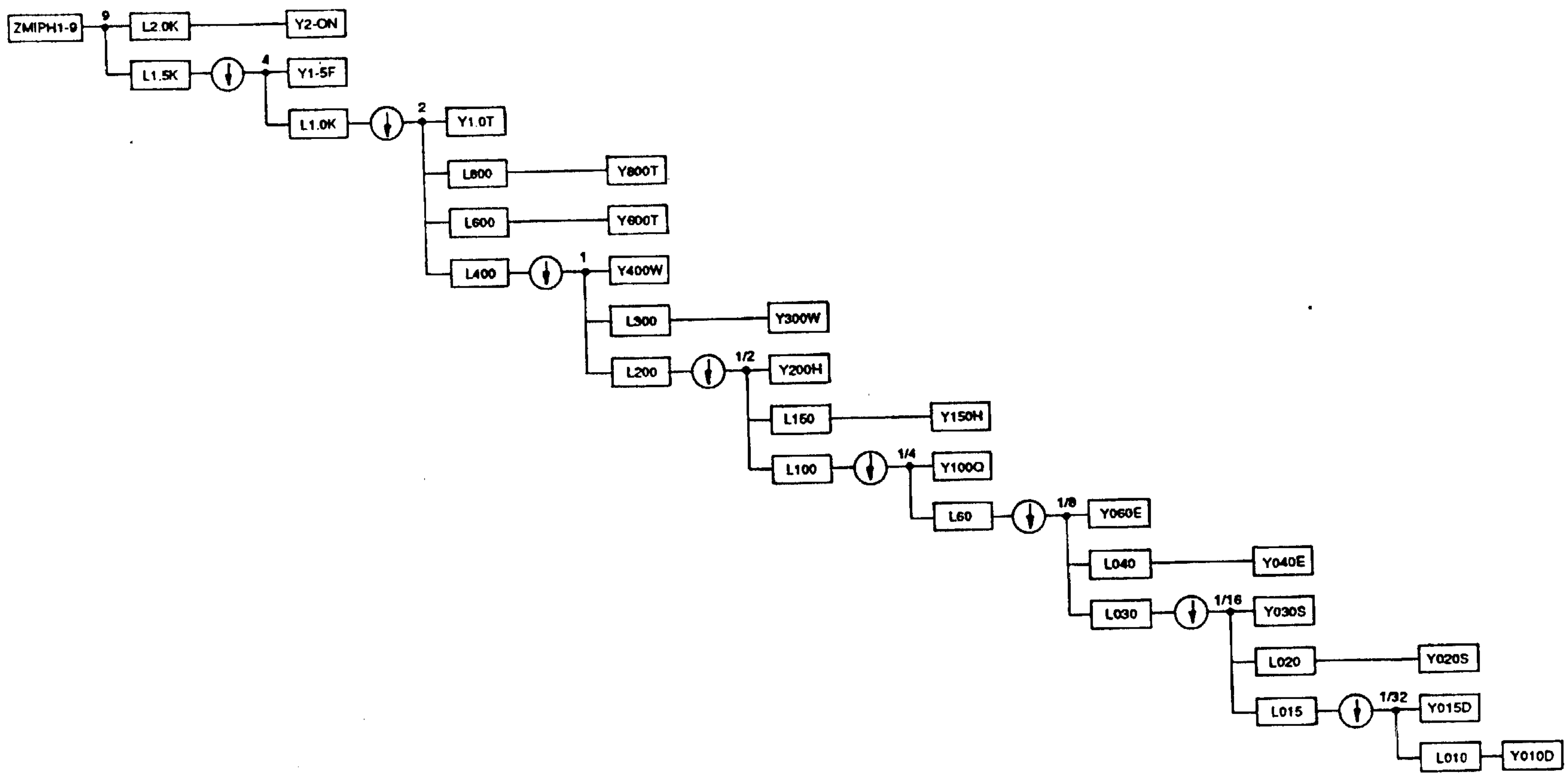

Figure 2-5. Block Diagram of Digital Filter Bank.

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