N O T I C E

THIS DOCUMENT HAS BEEN REPRODUCED FROM
MICROFICHE. ALTHOUGH IT IS RECOGNIZED THAT
CERTAIN PORTIONS ARE ILLEGIBLE, IT IS BEING RELEASED
IN THE INTEREST OF MAKING AVAILABLE AS MUCH
INFORMATION AS POSSIBLE

NASA

Technical Memorandum 80593

On the Performance of the Martin Digital Filter for High- and Low-Pass Applications

Charles R. McClain and Harvey Walden

OCTOBER 1979

ON THE PERFORMANCE OF THE MARTIN DIGITAL FILTER FOR

HIGH- AND LOW-PASS APPLICATIONS

Charles R. McClain

Harvey Walden

October 1979

ON THE PERFORMANCE OF THE MARTIN DIGITAL FILTER FOR HIGH- AND LOW-PASS APPLICATIONS

Charles R. McClain

Harvey Walden

ABSTRACT

The Martin digital filter is a nonrecursive numerical filter in which the weighting sequence is optimized by minimizing the excursion from the ideal rectangular filter in a least-squares sense over the entire domain of normalized frequency. Additional corrections to the weights in order to reduce overshoot oscillations (Gibbs phenomenon) and to insure unity gain at zero frequency for the low-pass filter are incorporated. The filter is characterized by a zero phase shift for all frequencies (due to a symmetric weighting sequence), a finite memory and stability, and it may readily be transformed to a high-pass filter. Equations for the filter weights and the frequency response function are presented, and applications to high- and low-pass filtering are examined. A discussion of optimization of high-pass filter parameters for a rather stringent response requirement is given, in an application to the removal of aircraft low-frequency oscillations superimposed on remotely-sensed ocean surface profiles. Several frequency response functions are displayed, both in normalized frequency space and in period space. A comparison of the performance of the Martin filter with some other commonly used low-pass digital filters is provided in an application to oceanographic data.

iii

CONTENTS

LIST OF ILLUSTRATIONS

LIST OF ILLUSTRATIONS (continued)

ON THE PERFORMANCE OF THE MARTIN DIGITAL FILTER FOR

HIGH- AND LOW-PASS APPLICATIONS

INTRODUCTION

With the development of high-speed digital computers, many concepts in analogue electronic

signal processing have been adapted for application to time series of discretely-sampled data, usually

equally spaced. During the past twenty years, a great deal of literature has been generated on such

topics as digital or numerical filters [Gold and Rader, 1969; Cadzow, 1973] and on spectral analysis

of discrete finite-duration time series [Jenkins and Watts, 1968; Brillinger, 1975; Harris, 1978].

Spectral analysis is the most powerful tool available for scrutinizing physical processes, and pre-

filtering or prewhitening the raw data prior to the analysis is often required for trend removal or for

handling so-called "red noise" [Davis, 1974]. Failure to prewhiten can cause severe distortion of

the computed spectra because of spectral leakage via the sidelobes of the window function. In this

paper, we will discuss a particular digital filter which we feel has many merits and will provide ex-

amples of its use for both high-pass and low-pass applications. The filter was developed by Martin

[1957; 1959] and is discussed by Linnette [1961] and briefly by Kaiser [Kuo and Kaiser, 1966],

and is applied by Davis [1974].

Before proceeding to a discussion of the Martin filter, a few remarks regarding filters in general

are in order. Linear numerical filters such as the Martin filter and most commonly used discrete sys-

tems are mathematically implemented by deriving output values via a linear difference equation,

$$y(nT) = \sum_{k=-M}^{M'} h_k \, x(nT + kT) + \sum_{k=1}^{N'} p_k \, y(nT - kT), \qquad (1)$$

where h_k and p_k are the filter (constant) coefficients and n, M, M', and N' are all non-negative inte-

gers. Equation (1) reflects the fact that the output y can be a linear function of future, present, and

past inputs x, as well as past outputs. Usually only causal systems (where M' = 0) are considered,

but there is no analytical reason for this restriction. The first term on the right side of Equation (1)

represents the convolution of a weighting sequence h_k with the input sequence, while the second

1

term determines whether the filter is recursive or nonrecursive. For a nonrecursive system ($p_k = 0$ for all k), the sum ($M + M'$) can be quite large in order to obtain a sharp filter response with minimum oscillation (this latter property is known as the Gibbs phenomenon). The sequence h_k for a nonrecursive filter is called the impulse response function, while the impulse response function for a recursive filter is the inverse z-transform of the transfer function of system (1). The design of recursive systems relies heavily on the z-transform and complex analysis, as discussed in detail by Cadzow [1973].

The existence of feedback allows the realization of a desired filter response which requires fewer calculations per output value. This savings in computational time does have its price, however. Using feedback, the filter must be designed so as to insure stability, and the phasing between component frequencies is not preserved. Also, such a filter has "infinite memory," which means that its impulse response spans the entire data set, and each data input influences all subsequent outputs. On the other hand, nonrecursive filters have a finite memory ($M + M' + 1$ points), are necessarily stable, and, if h_k is a symmetric sequence about k = 0 with an odd number of points, phase is preserved. Therefore, the best reason for applying recursive filters is for processing efficiency. However, techniques for implementing nonrecursive filters via fast Fourier transforms can make them competitive [Gold and Rader, 1969, Chapter 7]. An additional comment is that the transfer function for the Martin filter is not particularly simple mathematically. In some situations, great care must be taken in determining the filter parameter values which produce a particular desired response, and optimization of these parameters may be a trial and error procedure. Indeed, it appears to the authors that this fact has not always been appreciated, and this will be indicated later in this paper.

THE MARTIN FILTER

For discrete systems, the time lag T between samples is naturally quite important. Since we are primarily concerned with the frequency response of a system, the constant quantity $f_s = 1/T$ assumes a very significant role. The sampling frequency f_s is used as the factor for frequency scaling, and the normalized frequency $\gamma = f/f_s$ becomes the independent variable. The sampling theorem

2

for the time domain [Goldman, 1953, pp. 67–71] states that if a function g(t) contains no frequency components higher than f_{max}, then g(t) can be completely determined by sampling at a rate of $2 f_{max}$; otherwise, high frequency components are not uniquely determined and Nyquist folding occurs. Therefore, the range of normalized frequency γ is [−0.5, 0.5].

The Martin filter incorporates the following features:

(1) The phase shift for all frequencies is zero, i.e., h_k is a symmetric sequence, where $h_k = h_{-k}$.

(2) The weights are optimized by minimizing the error in a least-squares sense between the realizable gain function and an ideal gain function (a step function of unit amplitude at the desired cut-off γ_c) over the entire range of γ.

(3) The weights are corrected to minimize the overshoot oscillations (Gibbs phenomenon) in the gain function associated with sharp cut-off.

(4) The weights are corrected to insure unity gain at zero frequency.

The third feature above is accomplished by inserting a sine termination to the gain function. The width of this termination is adjustable and associated with the parameter b, the "slope of weights." The smaller b is chosen, the steeper the gain function will become and the more pronounced the over-shoot. With the corrections to the gain function, the filter weights are given by:

$$h_k = L_k + \alpha/(2N + 1), \text{ for } k = 0, 1, 2, \ldots, N, \qquad (2)$$

where

$$L_k = \begin{cases} \left[\dfrac{\cos(2\pi kb)}{1 - 16 k^2 b^2}\right]\left[\dfrac{\sin[2\pi k(\gamma_c + b)]}{\pi k}\right], & \text{for } k = 1, 2, \ldots, N \\[2em] 2(\gamma_c + b), & \text{for } k = 0 \end{cases}$$

and

$$\alpha = 1 - \left(L_0 + 2 \sum_{k=1}^{N} L_k\right).$$

3

The transfer or gain function (also called the frequency response function) is

$$H(\gamma) = h_0 + 2 \sum_{k=1}^{N} h_k \cos(2\pi k\gamma). \tag{3}$$

For $k = (4b)^{-1}$ or $b = (4k)^{-1}$, the indeterminacy in L_k is resolved by L'Hôpital's rule as:

$$L_k = b \cos\left(\frac{\pi\gamma_c}{2b}\right) = \frac{1}{4k} \cos(2\pi k\gamma_c). \tag{4}$$

In order to transform from low-pass (lp) filter weights to high-pass (hp), or vice versa:

$$h_k^{(hp)} = h_{-k}^{(hp)} = -h_k^{(lp)} = -h_{-k}^{(lp)} \text{ for } k = 1, 2, \ldots, N$$

and

$$h_0^{(hp)} = 1 - h_0^{(lp)},$$

where the filter weights given by Equation (2) are those for the low-pass case. Thus, the Martin filter is determined by three parameters: an integer N defined by the total number of weights $(2N + 1)$ in the weighting function, a parameter b which allows variation in the slope of the sine termination in the gain function, and $\gamma_c = f_c/f_s$, the normalized cut-off frequency for the ideal filter.

APPLICATIONS

A. High-pass Filters

The example selected to illustrate a high-pass filter application is the removal of relatively low-frequency oscillations (due to aircraft motion) superimposed on remotely-sensed profiles of the ocean surface. The application of numerical filters to this problem is discussed in Barnett and Wilkerson [1967], Ross et al. [1970], and McClain et al. [1979]. This problem is very similar to that of "red noise" presented in Davis [1974]. There are some subtleties, however, which, if not appreciated, can cause severe difficulties. In order to sample the surface adequately at the relatively high speed of an aircraft (75 to 140 m/sec), a sampling rate of 45 to 90 Hz is normally used. Horizontal sampling every two to three meters allows the resolution of wave components over the entire equilibrium range. This avoids Nyquist folding problems, because there is little energy at such wavelengths. Usual aircraft motion periodicities are of the order of 30 seconds or more, but during

4

flights at low altitudes and high winds, significant noise may be present at higher frequencies. In such cases, vertical acceleration should be monitored and included in the analysis. Also, many situations arise where the flight track may not be perpendicular to major wave components. This will occur when wind waves and swell superimpose or when tracks are determined from other considerations. In these situations, the aliased surface wavelengths may merge monotonically into the aircraft motion's frequency domain, and any filtering is detrimental. Under this circumstance, only vertical acceleration may be used to remove the aircraft motion in the time domain. An algorithm such as that presented by McClain et al. [1979] cannot deal with this situation, since it utilizes frequency domain techniques after time domain filtering.

With these considerations in mind, the design of the filter is attempted. The filter must allow all feasible aliased ocean wave components to pass, at least for reasonable angles of encounter, i.e., 0 to 60 degrees [Hammond and McClain, 1979], yet remove the more energetic aircraft-motion components. If we assume an aircraft speed of 100 m/sec, then a filter with a gain which begins to attenuate at about 6 or 7 seconds allows wavelengths of encounter up to 600 or 700 meters to pass. Using the P-M spectrum [Pierson and Moskowitz, 1964] to approximate the range of wavelengths we expect to sample, this length should be adequate. Of course, at high angles of encounter, it may not be possible to derive meaningful power spectra unless the wave field is highly directional with a known heading. However, the significant wave height $H_{1/3}$ and higher statistical moments, which do not require Doppler shifting for their determination, may be obtainable. Using the values of cut-off frequency f_c and $f_s \cong 90\,Hz$ implied above, the value of γ_c is approximately 0.002. The aircraft motion under smooth conditions would begin around $\gamma_c = 0.004$. Needless to say, these are quite small values of γ, and very close attention must be given to the filter design, since the actual frequency response is more sensitive to the parameters b and N than to γ_c over such a narrow range of γ.

As an aid to visualizing the problem of optimizing the filter parameters, graphs of the frequency response function are plotted in both γ-space and period τ-space (for $f_s = 90\,Hz$, the period τ in

seconds is given by $\tau = (90\gamma)^{-1}$.). The utility of τ-space is that by multiplying τ by the aircraft speed, one can see the attenuation in terms of distance or wavelength. Figure 1 provides a parametric study of the effects of various values of γ_c (with corresponding ideal cut-off periods τ_c of 5, 10, 20, and 100 seconds) on the response function $H(\tau)$. Figures 2 and 3 illustrate the effects of varying the parametric values of N and b, respectively, on the response function in τ-space. The digital filter used in McClain et al. [1979] is given in Figures 4 and 5, plotted as a function of τ and γ, respectively. In order to display the characteristics of the frequency response function in γ-space adequately, it is necessary to expand the graphical scale substantially, as in Figures 6 and 7, which demonstrate the initial rapid rise in the gain and the high-frequency Gibbs oscillations about unity, respectively, for the same filter as in Figures 4 and 5. The nature of the Gibbs phenomenon, as a function of N, is illustrated further in Figure 8, which utilizes, in γ-space, the same parameter values as in Figure 2. In particular, note how the value of N influences the effective cut-off and the frequency of the Gibbs oscillations. By further experimentation in the choice of filter parameters, an improved filter has been developed (Figure 9). This filter exhibits the desirable characteristics of a smaller amplitude overshoot, a sharper cut-off, and near-zero gain over almost all of the aircraft motion domain, with a slight compromise on the effective cut-off (which can be defined quantitatively as the point at which 0.9 gain is achieved).

B. Low-pass Filters

As an example of the performance of the Martin filter in the low-pass mode, a comparison with the results presented in Roberts and Roberts [1978] will be offered. In that work, the Butterworth, cosine-Lanczos, Gaussian, and ideal (or rectangular) filters were applied to sea level data sampled each hour at two points in the Gulf of Alaska. The data record consisted of 512 points, and the objective of the low-pass filtering was to examine long-period fluctuations in a signal dominated by diurnal and semi-diurnal tides. Figure 10 provides the superposition of the Martin filter response onto their power gain functions for the four low-pass filters enumerated above [Roberts and Roberts, 1978, p. 5511, Figure 1]. Note that the power gains shown are the squares of the frequency response
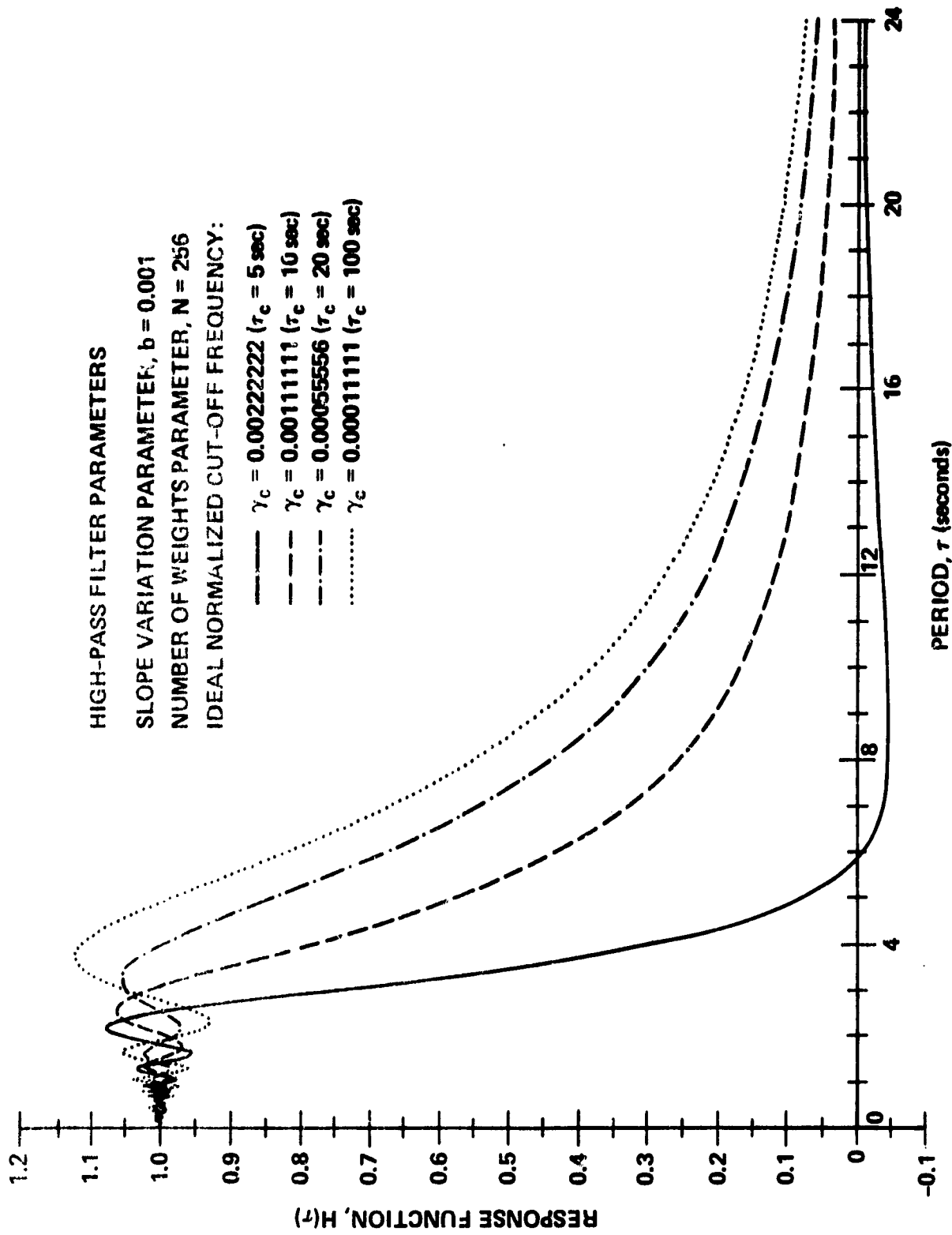
Figure 1. Variation of the response function $H(\tau)$ with the ideal normalized cut-off frequency γ_c for a given set of high-pass filter parameters b and N

HIGH-PASS FILTER PARAMETERS

IDEAL NORMALIZED CUT-OFF FREQUENCY.
$\gamma_c = 0.00123457$ ($\tau_c = 9$ sec)
SLOPE VARIATION PARAMETER, $b = 0.0001$
NUMBER OF WEIGHTS PARAMETER:

N = 16
N = 32
N = 64
N = 128

Figure 2. Variation of the response function H(τ) with the number of weights parameter N for a given set of high-pass filter parameters γ_c and b

RESPONSE FUNCTION, H(τ)

PERIOD, τ (seconds)

8
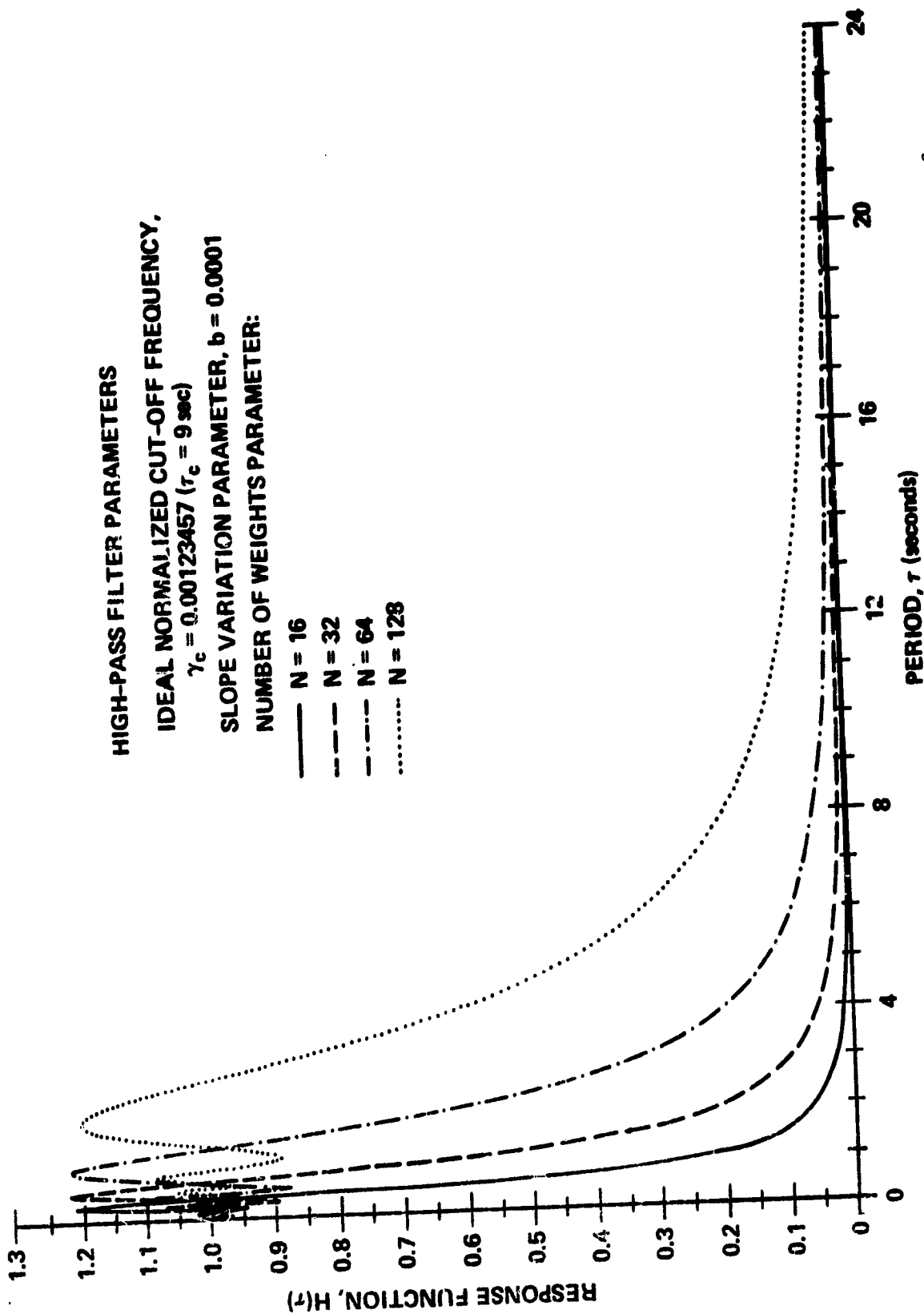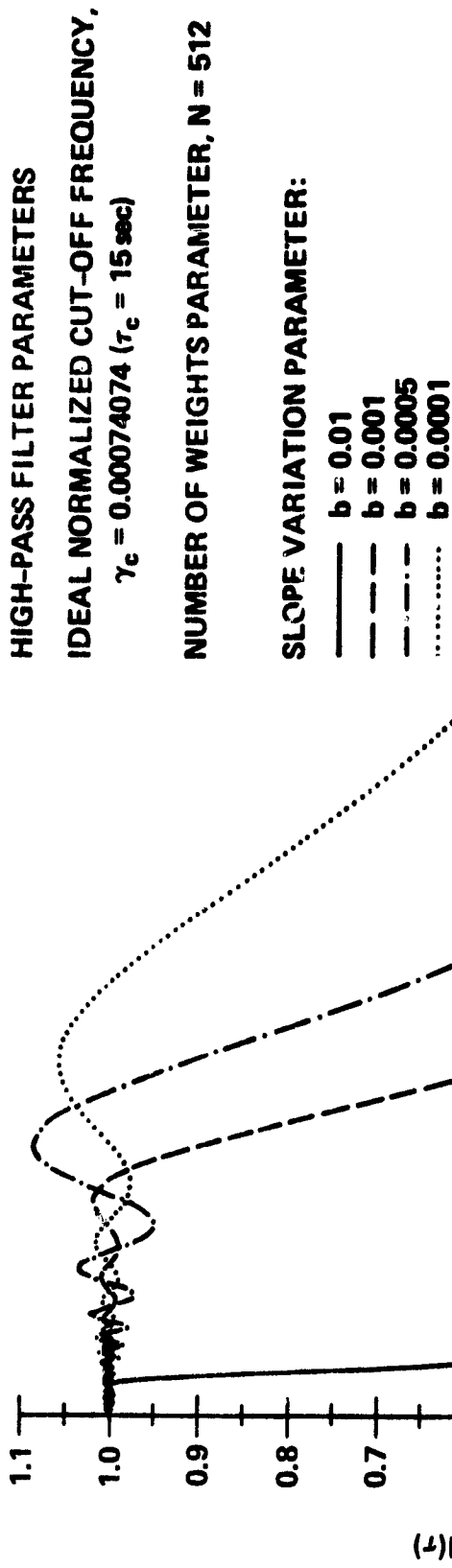
Figure 3. Variation of the response function H(τ) with the slope variation parameter b for a given set of high-pass filter paramete~ , and N

Figure 4. Response function H(τ), given in period space, for the high-pass digital filter used in the analysis of laser profilometer data in the Gulf Stream Ground Truth Project

HIGH-PASS FILTER PARAMETERS

IDEAL NORMALIZED CUT-OFF FREQUENCY, γ_c = 0.0012346 (τ_c = 9 sec)
SLOPE VARIATION PARAMETER, b = 0.0001
NUMBER OF WEIGHTS PARAMETER, N = 512

HIGH-PASS FILTER PARAMETERS

IDEAL NORMALIZED CUT-OFF FREQUENCY, γ_c = 0.0012346
SLOPE VARIATION PARAMETER, b = 0.0001
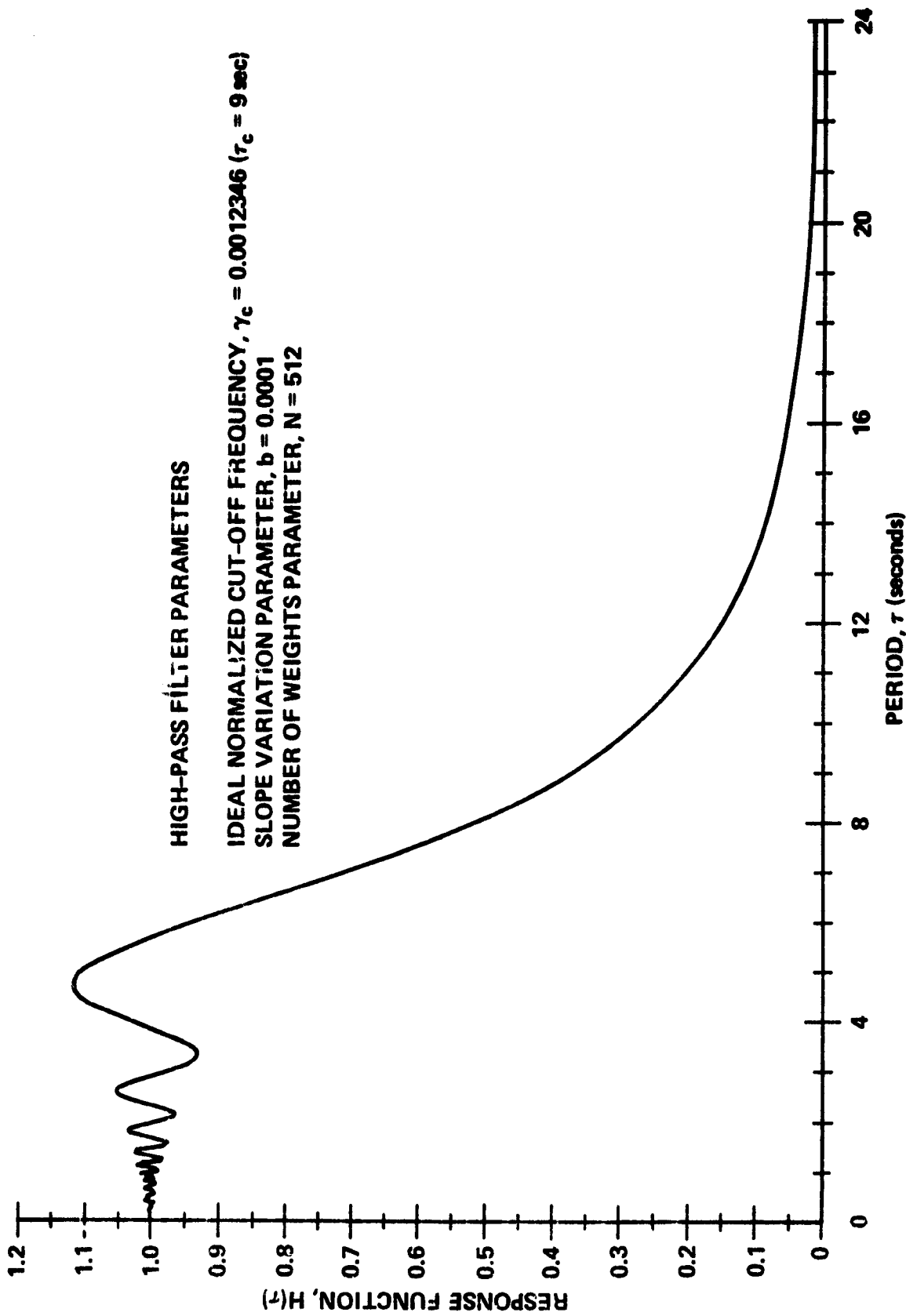NUMBER OF WEIGHTS PARAMETER, N = 512

FREQUENCY RESPONSE FUNCTION, H(γ)
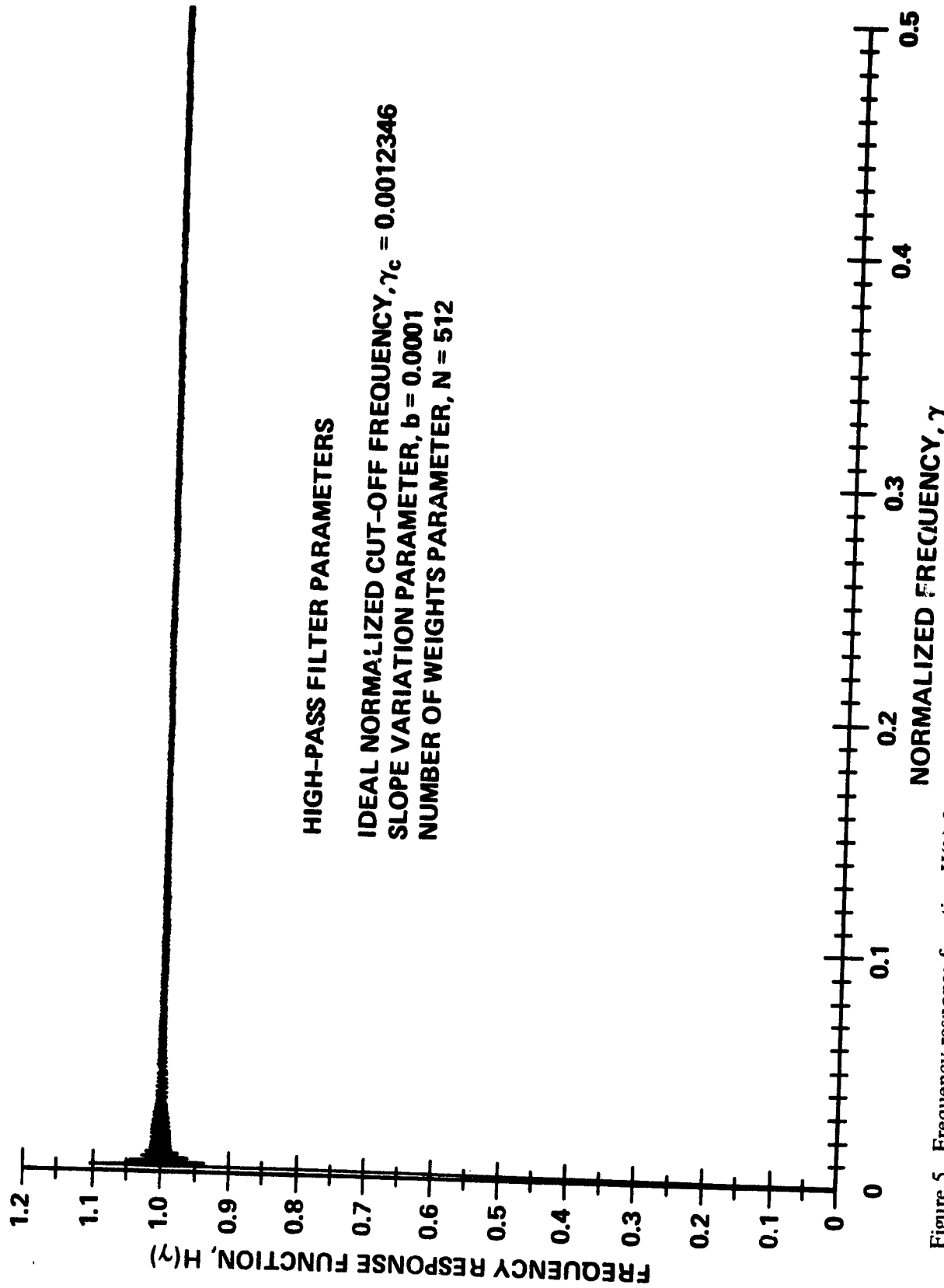
NORMALIZED FREQUENCY, γ

Figure 5. Frequency response function H(γ) for the high-pass digital filter used in the analysis of laser profilometer data in the Gulf Stream Ground Truth Project

11

HIGH-PASS FILTER PARAMETERS

IDEAL NORMALIZED CUT-OFF FREQUENCY, γ_c = 0.0012346

SLOPE VARIATION PARAMETER, b = 0.0001

NUMBER OF WEIGHTS PARAMETER, N = 512

Figure 6. Frequency response function $H(\gamma)$, for $0 \leq \gamma < 0.002$, for the high-pass digital filter used in the analysis of laser profilometer data in the Gulf Stream Ground Truth Project

HIGH-PASS FILTER PARAMETERS

IDEAL NORMALIZED CUT-OFF FREQUENCY, $\gamma_c = 0.0012346$

SLOPE VARIATION PARAMETER, $b = 0.0001$

NUMBER OF WEIGHTS PARAMETER, $N = 512$

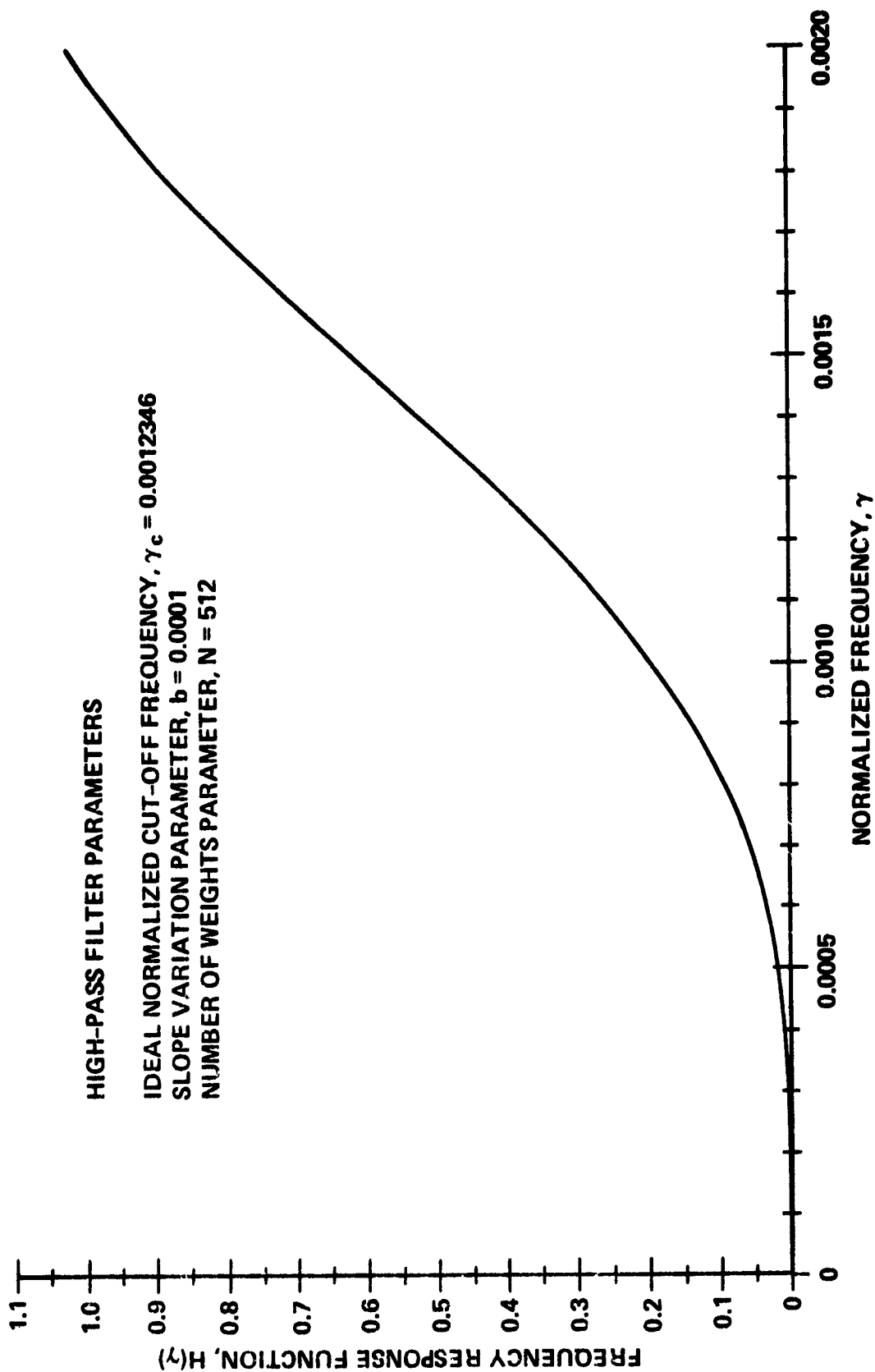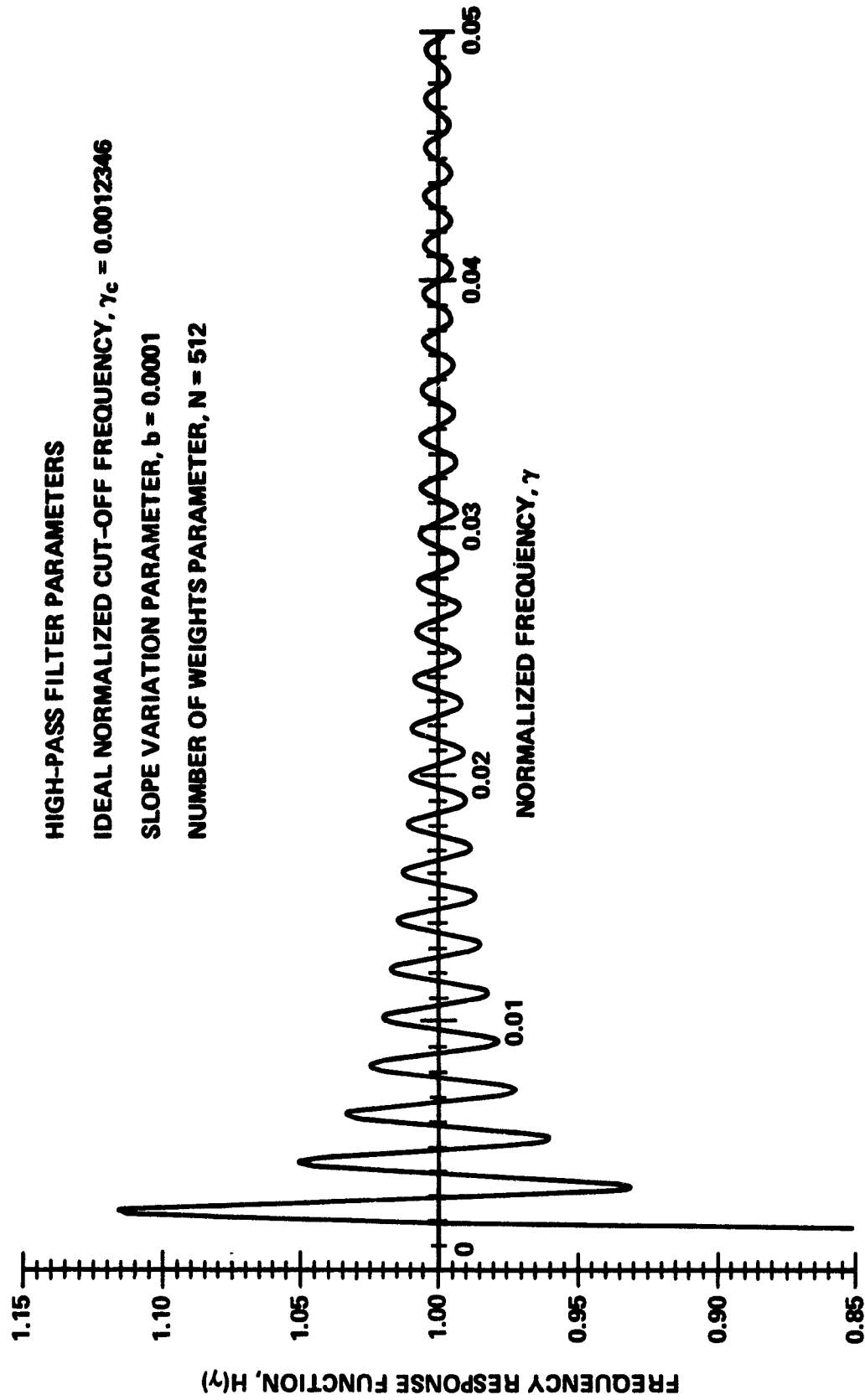Figure 7. Frequency response function $H(\gamma)$, for $0 < \gamma \leq 0.05$, for the high-pass digital filter used in the analysis of laser profilometer data in the Gulf Stream Ground Truth Project

13

Figure 8. Variation of the frequency response function H(γ), for $0 < \gamma \leq 0.10$, with the number of weights parameter N for a given set of high–pass filter parameters γ_c and b

14
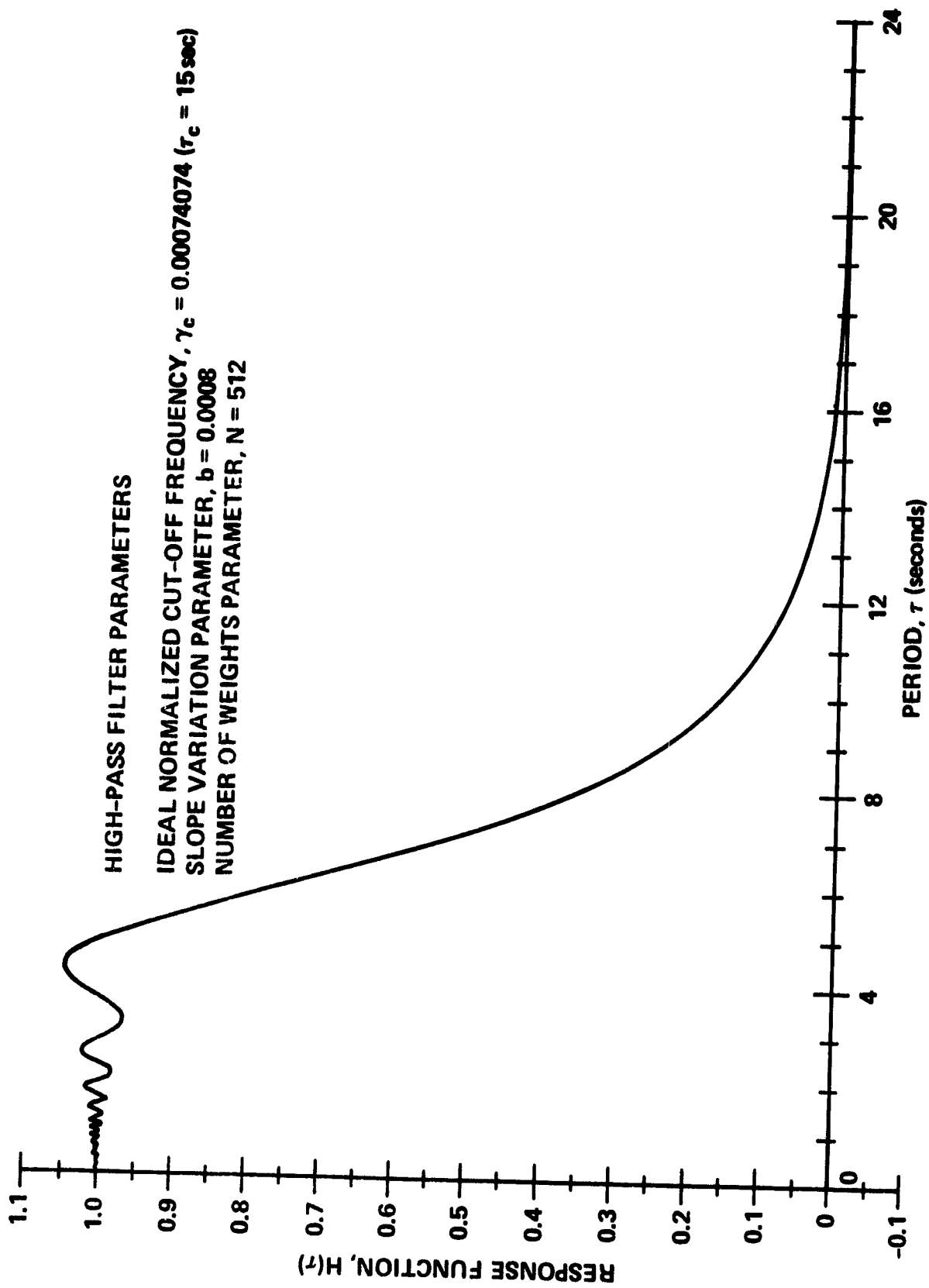
Figure 9. Response function H(τ), given in period space, for an improved high–pass digital filter

HIGH-PASS FILTER PARAMETERS

IDEAL NORMALIZED CUT-OFF FREQUENCY, γ_c = 0.00074074 (τ_c = 15 sec)
SLOPE VARIATION PARAMETER, b = 0.0008
NUMBER OF WEIGHTS PARAMETER, N = 512

PERIOD, τ (seconds)

RESPONSE FUNCTION, H(τ)

15

Figure 10. Power gain functions for five low-pass filters with ideal cut-off frequency at 1/(40 hours). The cosine-Lanczos and Gaussian filters each were given 60 weights, and the number of weights parameter N = 60 for the Martin filter

function, because of the fact that since the Butterworth filter is recursive, the signal must be filtered forward and then backward in time through the same filter in order to eliminate the phase shift that would otherwise result. The ideal cut-off frequency is taken as $(40 hr)^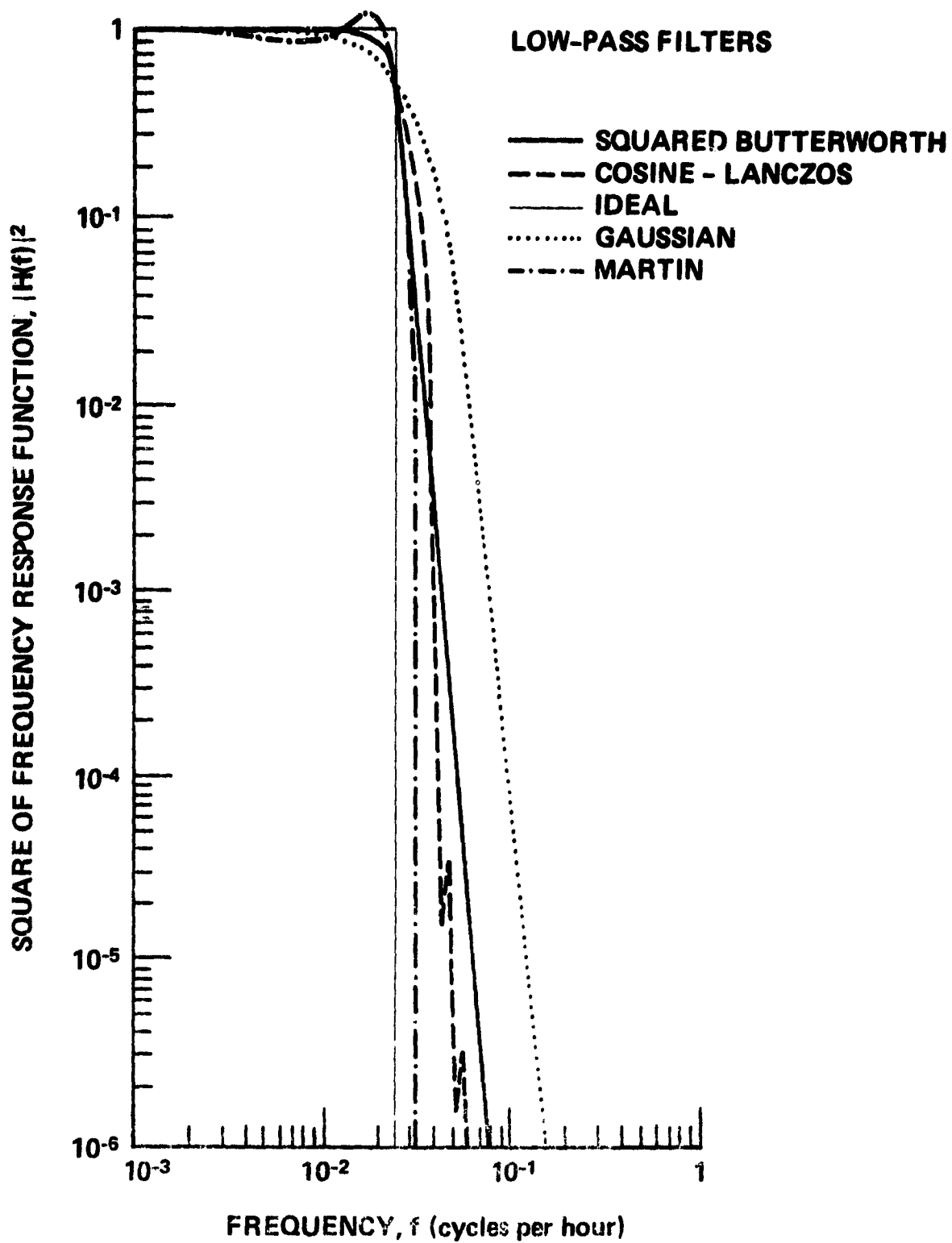{-1} = 0.025$ cycles/hr for all the filters, and for the Martin filter, the slope variation parameter $b = 0.0001$. Since the sampling frequency $f_s = 1 hr^{-1}$, the frequency f in cycles per hour is equivalent numerically to the normalized frequency γ. For proper comparisons, the number of terms in the weighting sequence h_k must be determined, but it is not clear from the Roberts and Roberts paper what value for $(2N + 1)$ should be chosen. That paper states that "60 weights were used for both the cosine-Lanczos and the Gaussian filters," in which case h_k might be presumed to consist of 60 terms (i.e., $N \cong 30$); however, the first and last 60 hours of the data output from the Butterworth filter are discarded "for ease of comparison with other low-pass filters," indicating that h_k contains 121 terms (i.e., $N = 60$). It should be noted that the first and last N points of a set of input data cannot be processed by the Martin filter (or any similar nonrecursive and phase-preserving filter), since, for these "boundary" data points, N past input or N future input points are not available.

In order to demonstrate the efficacy of the Martin filter, a comparison is presented in Figure 11 in which the superposed Martin filter assumes $N = 30$, or one-half the number of weights that the other filters presumably incorporate, with the parameters γ_c and b unchanged from the previous figure. Note that the Gibbs oscillations about unity for the Martin filter are greatly diminished, and, although the response attenuation is not quite as sharp as for the $N = 60$ case, the gain remains comparatively superior to the other filters displayed. For the sake of completeness, the impulse response function, or the weighting sequence h_k, for the Martin filter of Figure 10 where $N = 60$ is given in Figure 12. The impulse response function for the Martin filter of Figure 11 where $N = 30$ would virtually coincide with this curve but, of course, be truncated at $k = \pm 30$.

CONCLUSIONS

Many disciplines utilize time series analysis and associated data processing operations for the interpretation of data and the validation of theoretical models. As an example, the use of digital
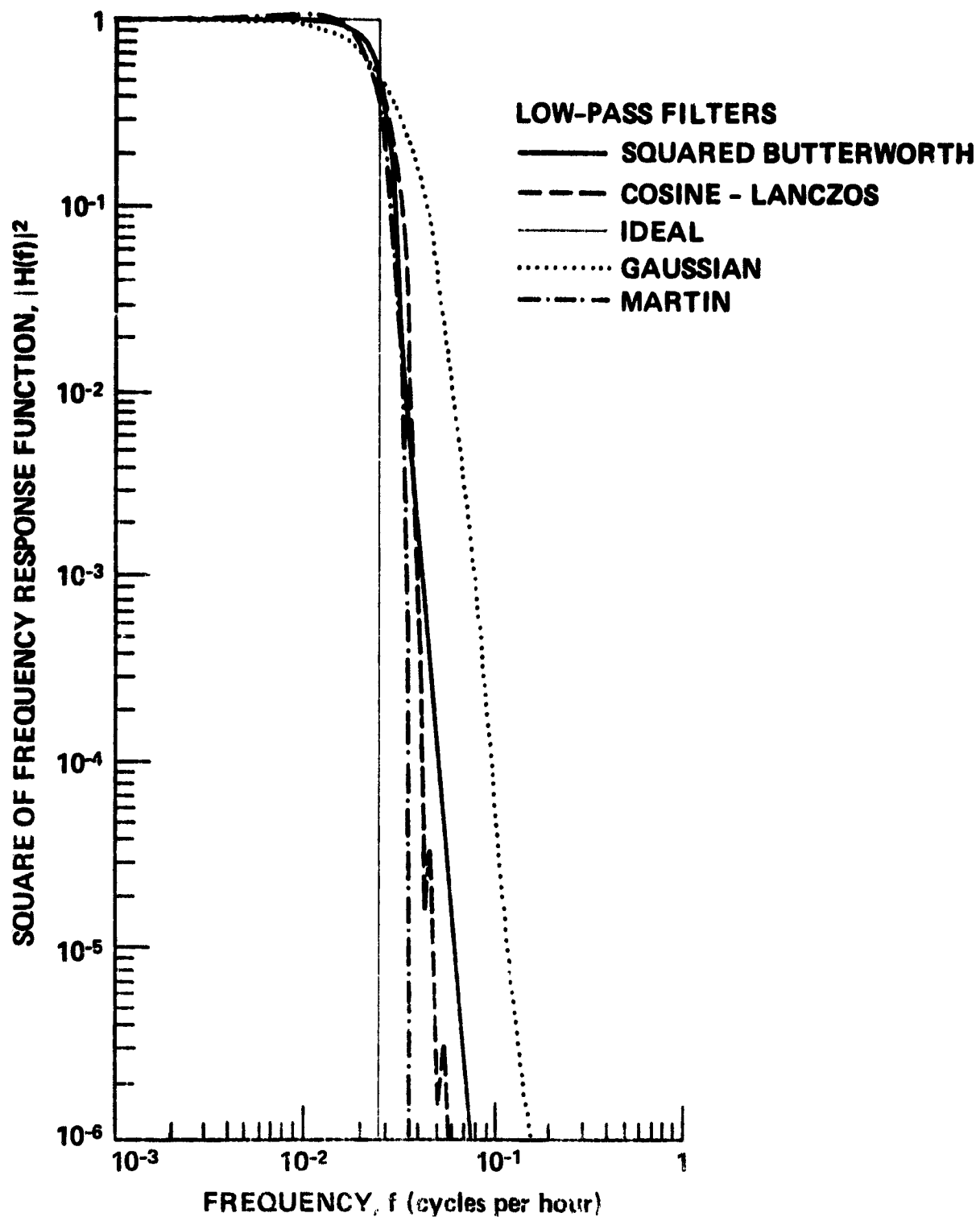
Figure 11. Power gain functions for five low-pass filters with ideal cut-off frequency at 1/(40 hours). The cosine-Lanczos and Gaussian filters each were given 60 weights, and the number of weights parameter N = 30 for the Martin filter
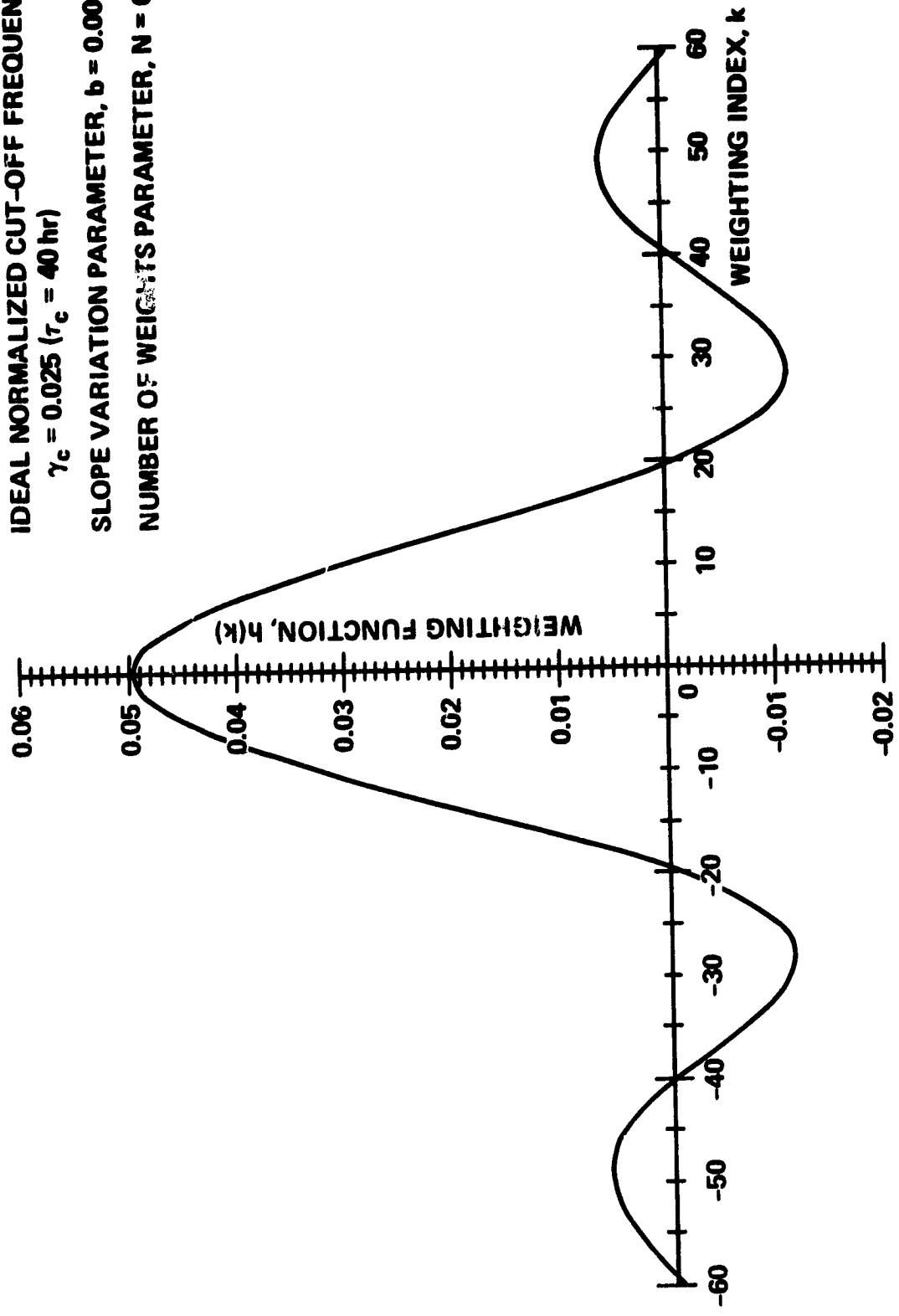
18

Figure 12. Impulse response function, or the weighting sequence, for the low-pass Martin filter

19

filters for pre-whitening data is central to the problem of deriving valid spectra. In general, filtering techniques are useful to effect the efficient separation of various frequencies imbedded in the sampled data, thereby separating the physical phenomenon of interest from measurement and transmission errors. Furthermore, the application of filtering offers a systematic and efficient means of performing this separation of frequencies on large volumes of similar data by digital computers. A single simple computer program may perform both high–pass and low–pass filtering; only the proper filter parameters must be chosen for a given application. As has been indicated in this paper, the selection of the proper filter parameters can require considerable and careful effort in the optimization of the response function.

The Martin filter has been shown to possess many desirable characteristics, including stability, finite memory, and phase preservation. In addition, it can perform competitively with some of the most commonly used low–pass digital filters, in terms of providing a sharp response function with minimal oscillations and requiring fewer computations when a decreased number of weights will suffice. With the steep response function attained with the Martin low–pass filter, an adjustment in the value of the ideal normalized cut–off frequency γ_c can yield a response nearly coincident with that of the ideal filter over most of the ordinate values. To decrease the magnitude of the overshoot or Gibbs phenomenon, an appropriate combination of decreasing the number of weights parameter N or increasing the slope variation parameter b is generally effective. Of course, there are trade–offs in the selection of the proper filter parameters. For instance, by increasing the value of b, the sharpness of the desired cut–off is diminished, and by decreasing the value of N, the maximum absolute error in the gain (as compared to the ideal filter) is increased. One additional parameter that influences the filter performance is the sampling frequency f_s, inasmuch as this is used as a normalization factor in defining the independent frequency variable γ. For instance, if the value of f_s is decreased from the nominal value of 90 Hz assumed in this paper, then the effective cut–off in τ–space is more sensitive to the value of γ_c and a sharp cut–off is more difficult to achieve with an acceptable overshoot.

20

In sum, the investigator has considerable flexibility through the choice of filter parameters in the design of a Martin digital filter appropriate for a given application. It is felt that the benefits offered by the Martin filter more than compensate for the expenditure required to design it properly, especially in light of the common availability of small calculators with peripheral plotters which readily permit one to adjust the filter parameters until the desired response is obtained.

REFERENCES

Barnett, T. P. and J. C. Wilkerson, "On the generation of ocean wind waves as inferred from airborne radar measurements of fetch-limited spectrum," J. Mar. Res., vol. 25, pp. 292-328, 1967.

Brillinger, D. R. Time Series Data Analysis and Theory, Holt, Rinehart and Winston, New York, 500 pp., 1975.

Cadzow, James A., Discrete-Time Systems: An Introduction with Interdisciplinary Applications, Prentice-Hall, Englewood Cliffs, N. J., 440 pp., 1973.

Davis, T. M., "Theory and practice of geophysical survey design," Naval Oceanographic Office Reference Publication 13, 137 pp., 1974.

Gold, Bernard and Charles M. Rader, Digital Processing of Signals, McGraw-Hill, New York, 269 pp., 1969.

Goldman, Stanford, Information Theory, Prentice-Hall, New York, 385 pp., 1953.

Hammond, D. L. and C. R. McClain, "Spectral distortion inherent in airborne profilometer measurements of ocean wave heights," Ocean Engineering, 1979 (accepted for publication).

Harris, Fredric J., "On the use of windows for harmonic analysis with the discrete Fourier transform," Proc. IEEE, vol. 66, no. 1, pp. 51-83, January 1978.

Jenkins, G. M. and D. G. Watts, Spectral Analysis and Its Applications, Holden-Day, San Francisco, 525 pp., 1968.

Kuo, F. F. and J. F. Kaiser, System Analysis by Digital Computer, John Wiley and Sons, New York, 438 pp., 1966.

Linnette, H. M., "Statistical filters for smoothing and filtering equally spaced data," U.S. Navy Electronics Laboratory (San Diego) Research Report 1049, 126 pp., July 1961.

Martin, Marcel A., "Frequency domain applications in data processing," General Electric Co., Missile and Ordnance Systems Dept. (Philadelphia) Document No. 57SD340, 128 pp., May 1957.

Martin, Marcel A., "Frequency domain applications to data processing," IRE Trans. on Space Electronics and Telemetry, pp. 33–41, March 1959.

McClain, C. R., D. T. Chen and D. L. Hammond, "Gulf Stream ground truth project: Results of the NRL airborne sensors," Ocean Engineering, 1979 (accepted for publication).

Pierson, W. J., Jr. and L. Moskowitz, "A proposed spectral form for fully developed wind seas based on the similarity theory of S. A. Kitaigorodskii," J. Geophys. Res., vol. 69, no. 24, pp. 5181–5190, 1964.

Roberts, Jo and Thomas D. Roberts, "Use of the Butterworth low–pass filter for oceanographic data," J. Geophys. Res., vol. 83, no. C11, pp. 5510–5514, November 1978.

Ross, D. B., V. J. Cardone, and J. W. Conaway, Jr., "Laser and microwave observations of sea–surface condition for fetch–limited 17– to 25–m/s winds," IEEE Trans. on Geoscience Electronics, vol. GE–8, no. 4, pp. 326–336, 1970.

BIBLIOGRAPHIC DATA SHEET

1. Report No. TM 80593	2. Government Accession No.	3. Recipient's Catalog No.
4. Title and Subtitle ON THE PERFORMANCE OF THE MARTIN DIGITAL FILTER FOR HIGH- AND LOW-PASS APPLICATIONS		5. Report Date October 1979
		6. Performing Organization Code 941
7. Author(s) Charles R. McClain and Harvey Walden		8. Performing Organization Report No.
9. Performing Organization Name and Address NASA/Goddard Space Flight Center Greenbelt, Maryland		10. Work Unit No.
		11. Contract or Grant No.
12. Sponsoring Agency Name and Address National Aeronautics and Space Administration Washington, D.C.		13. Type of Report and Period Covered Technical Memorandum
		14. Sponsoring Agency Code

15. Supplementary Notes

16. Abstract

The Martin digital filter is a nonrecursive numerical filter in which the weighting sequence is optimized by minimizing the excursion from the ideal rectangular filter in a least-squares sense over the entire domain of normalized frequency. Additional corrections to the weights in order to reduce overshoot oscillations (Gibbs phenomenon) and to insure unity gain at zero frequency for the low-pass filter are incorporated. The filter is characterized by a zero phase shift for all frequencies (due to a symmetric weighting sequence), a finite memory and stability, and it may readily be transformed to a high-pass filter. Equations for the filter weights and the frequency response function are presented, and applications to high- and low-pass filtering are examined. A discussion of optimization of high-pass filter parameters for a rather stringent response requirement is given, in an application to the removal of aircraft low-frequency oscillations superimposed on remotely-sensed ocean surface profiles. Several frequency response functions are displayed, both in normalized frequency space and in period space. A comparison of the performance of the Martin filter with some other commonly used low-pass digital filters is provided in an application to oceanographic data.

17. Key Words (Selected by Author(s)) Digital filtering Frequency response High-pass, low-pass	18. Distribution Statement Unclassified–Unlimited		
19. Security Classif. (of this report) Unclassified	20. Security Classif. (of this page) Unclassified	21. No. of Pages 29	22. Price*

*For sale by the National Technical Information Service, Springfield, Virginia 22151.

GSFC 25-44 (10/77)