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(54) **FILTER**

FILTER

FILTRE

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Description

Technical Field

5   **[0001]** The present invention relates to filters, and more specifically, to a finite impulse response (FIR) filter having a small number of taps and steep attenuation characteristics.

Background Art

10   **[0002]** With recent digitalization of multimedia information, the importance of arithmetic processing algorithms and arithmetic processing circuits that implement digital signal processing has been increasing. Especially, digital filters are used to eliminate noise, adjust frequency characteristics, separate signals, and so on. FIR filters having finite impulse response ensure stability always against bounded input. If the impulse response is symmetrical, the FIR filters can implement perfect linear phases. The linear phases are important characteristics in waveform transmission, measure-
15   ment, sound reproduction, and so on. However, because of the trade-off between precision required in the frequency characteristics of the filter and the scale of the filter, high-order transfer characteristics are required to obtain steeper attenuation characteristics, and the circuit requires a large number of multipliers and delay elements. To create a passband filter with a very narrow band, for example, a very high degree of transfer characteristics is required, and a large number of multipliers are required accordingly.

20   **[0003]** In that context, many propositions have been made. Non-patent literature 1 describes the Remez method as a conventional method that has been used usually. Non-patent literature 2 describes a method of configuring a low-sensitivity linear-phase FIR filter by cascade connection, and the cascade connection allows the number of multipliers to be reduced. Patent literature 1 describes an FIR filter provided to vary the delay of the filter output signal.
Moreover, FIR filters for various uses such as filters for separating acoustic signal bands and filters for eliminating noise
25   in image have been studied.
[0004] There are more conventional technologies. For example, non-patent literature 3 discloses the design and implementation of oversampled filter banks for multilevel FIR filters with low aliasing.
Non-patent literature 4 discloses a filter bank with desired frequency response and low aliasing noise.
Non-patent literature 5 discloses a realization of oversampled cosine modulated filter banks with perfect reconstruction.
30   Patent literature 2 discloses a filter bank and filtering method that can efficiently implement critically sampled filter banks for real-value signals and especially a cosine modulated filter bank.
[0005] Nakamura et al: "Design of FIR filter with small number of coefficients based on compactly supported fluency sampling function" (2003 IEEE Pacific Rim Conference on Communications, computers and signal processing) discloses the general concept of upsampling a basic low-pass and high-pass filter having an impulse response function configured
35   by a piecewise polynomial with a finite support and using a value of the impulse response function at a connection point between the polynomials as a filter coefficient.
[0006] EP 1 533 898 A1 discloses low-pass and high-pass filter units with a common center frequency designed on the basis of a basic unit filter having a predetermined basic numeric string as a filter coefficient. The low-pass and high-pass filter units are connected in cascade in order to design a band pass filter.

40

Prior Art References

Non-Patent Literature

45   **[0007]**

Non-patent literature 1: D. Shpak, "A generalized Remez method for the design of FIR digital filters," IEEE Trans. Circuit System, Vol. 37, No. 2, pp. 161-174, Feb. 1990.
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Non-patent literature 6: Nakamura, Koji, et al: "Design of FIR filter with small number of coefficients based on compactly supported fluency sampling function" (2003 IEEE Pacific Rim Conference on Communications, computers and signal processing)

Patent Literature

[0008]

Patent literature 1: Japanese Unexamined Patent Application Publication No. 2006-20191
Patent literature 2: Japanese Unexamined Patent Application Publication No. 2001-102931
Patent literature 3: European patent application EP 1 533 898 A1

Disclosure of Invention

Problems to be solved by the Invention

[0009] In any conventional filter, however, the number of multipliers (number of multiplications in calculation by software) and the number of delay elements (number of memories in processing by software) have not been reduced dramatically. For example, according to non-patent literature 1, extensive numbers of multipliers and delay elements are required to implement a filter having steep attenuation characteristics. According to non-patent literature 2, low-sensitivity filters can be realized, but the technology is not suitable for filters having steep attenuation characteristics.
[0010] A high-performance filter having steep attenuation characteristics configured by any conventional method requires extensive numbers of input taps, delay circuits, and multipliers, and it has taken very long until filter output with predetermined characteristics becomes stable.
Music, video, and other multimedia data distributed recently in MP3 (audio compression), MPEG (video compression), and other forms involves problems such as the deluge of information and quality lowered by jaggies. The compression processing includes band separation in the frequency domain, which requires excessive numbers of multipliers and delay elements.
[0011] Since the conventional REMEZ filter uses the SINC function (function over the infinite interval) as the basic function, reduction to a finite interval is needed in the actual circuit and is known as one possible cause of noise.
In view of the problems described above, it is an object of the present invention to provide a filter that can exhibit designated characteristics with reduced numbers of input taps, delay circuits, and multipliers, can improve the response, and can lower the cost.
[0012] Another object of the present invention is to provide a low-cost FIR filter with high noise elimination characteristics.
A further object of the present invention is to provide an FIR filter that does not generate jaggies and is suitable for signal processing.
FIR filters can be used in a variety of apparatuses such as audio apparatuses including amplifiers, image apparatuses for moving or still image processing, communication apparatuses including mobile phones, control apparatuses, PCs and other computers.

Means for Solving the Problems

[0013] Since one main feature of the present invention is that the basic function is an impulse response function expressed by an n-th-degree piecewise polynomial having a finite support, and nodal values of the function are used as filter coefficients, a filter configured according to the present invention samples the input signal at a smaller number of points, in comparison with a filter using the conventional SINC function as the basic function, and can exhibit frequency characteristics without support reduction error or aliasing error.
Another feature of the present invention is that the n-th-degree piecewise polynomial needs to be a quadratic to quartic function, and the frequency characteristics in the passband are flat with reduced ripples and have linear phases.
[0014] Another feature of the present invention is that scaling the basic filters in frequency allows the passband to be narrowed in accordance with the scaling value and increases the number of zero points, consequently increasing the attenuation degree in the stopband.
A further feature of the present invention is the cascade connection of scaled filters, which allows a high-performance filter to be configured with a reduced number of multipliers. An additional feature of the present invention is that the cascade connection of the scaled filters means the combination of low-pass and high-pass scaled filters having a desired passband or a wider passband and ensures desired cutoff frequencies.
[0015] A further additional feature of the present invention is that since scaled filters are selected depending on the

attenuation degree in the stopband, a very high attenuation degree can be provided in the stopband.

[0016] According to the present invention, a filter having desired passband characteristics is provided, in particular a desired cut-off frequency, a desired passband attenuation degree and a desired frequency width of the passband, comprising

-) a basic low-pass filter with passband width and having an impulse response function configured by a piecewise polynomial with a finite support and using values of the impulse response function at connection points between the polynomials as filter coefficients c(k),

-) a basic high-pass filter with passband width and having filter coefficients b(k) obtained by inverting the sign of the filter coefficients c(k) alternately, and

-) upsampled low-pass filters and upsampled high-pass filters combined in a cascade connection, wherein

-) the upsampled low-pass filters are formed by upsampling the basic low-pass filter by all values of p (p=1,2,...P-1), with p being an upsampling factor in the time domain and P-1 being smaller than M, and

-) the upsampled high-pass filters are formed by upsampling the basic high-pass filter by all values of q (q=1,2,... Q-1), with q being an upsampling factor in the time domain and Q-1 being smaller than M,

[0017] with the maximum upsampling factor M+1 being determined by the ratio of the passband width of the basic filter to the desired passband width.

[0018] According to a further embodiment of the present invention, a filter providing desired characteristics is provided wherein the upsampling factor P is the minimum value of p that satisfies the following equation:

$$f_3^{(p)} = f_3^{(0)}/(1+p) < f_3$$

with $f_3^{(p)}$ being the cut-off frequency of the upsampled low-pass filter (L_P) defined by a passband attenuation degree of -3dB, $f_3^{(0)}$ being the cut-off frequency of the basic low-pass filter (L_0) defined by a passband attenuation degree of -3dB and f_3 being the desired cut-off frequency (f_1) of the cascade connection stage defined by a desired passband attenuation degree (A) of -3dB.

[0019] According to the present invention, a method is provided for configuring an FIR filter to provide desired passband characteristics, in particular a desired passband width Δfb(M), the filter comprising a cascade connection of

-) a basic low-pass filter with a passband width Δfb(0) and having an impulse response function configured by a piecewise polynomial with a finite support and using a value of the impulse response function at a connection point between the polynomials as filter coefficients c(k),

-) a basic high-pass filter with a passband width Δfb(0) and having filter coefficients b(k) obtained by inverting the sign of the filter coefficients c(k) alternately, and

-) upsampled low-pass filters (L_p) and upsampled high-pass filters (H_q),

the method comprising the step of inputting a desired center frequency, a desired cut-off frequency, a desired stopband frequency, a desired passband attenuation degree at the cut-off frequency, a desired stopband attenuation degree at the stopband frequency, and a sampling frequency,

wherein the method further comprises the steps of

-) calculating the desired passband width Δfb (M) as (f_1-f_C)*2 and obtaining the maximum upsampling factor M+1 by calculating the ratio of the passband width Δfb(0) of the basic filter to the desired passband width Δfb(M),

-) sequentially selecting a number of α_p (α_p =0,1,2,...) of the upsampled low-pass filters formed by upsampling the basic low-pass filter by all values of p (p=1,2,...P-1), with p being an upsampling factor in the time domain and P-1 being smaller than M,

-) and sequentially selecting a number of β_q (β_q =0,1,2,...) of the upsampled high-pass filters formed by upsampling the basic high-pass filter by all values of q (q=1,2,... Q-1), with q being an upsampling factor in the time domain and Q-1 being smaller than M, and

-) determining whether a cascade connection of the selected upsampled low-pass filters and upsampled high-pass filters conforms to the desired passband attenuation degree at the cut-off frequency and to the desired stopband attenuation degree at the stopband frequency.

[0020] According to an embodiment of the present invention, the selection of upsampled low-pass filters and upsampled high-pass filters is made depending on whether the attenuation degree at the stopband frequency is lower than or equal to the desired stopband attenuation degree.

[0021] According to a further embodiment of the present invention, the selection of upsampled low-pass filters and upsampled high-pass filters is made depending on whether the attenuation degree at the cut-off frequency is higher than or equal to a predetermined permissible level below the desired passband attenuation degree.

[0022] According to a further embodiment of the present invention an optimum combination of p, q, α_p, and β_q is selected by selecting α_p and β_q in descending order from stored maximum values and selecting p and q in descending order from the maximum upsampling factor M.

[0023] According to a further embodiment of the present invention, α_p and β_q are increased, if this causes a filter gain at the desired stopband frequency of the filter to decrease.

Advantages of the Invention

[0024] According to the present invention, low-pass filters, high-pass filters, and band-pass filters that have reduced numbers of taps, delay circuits, and multipliers and exhibit excellent frequency separation can be implemented because the filters are configured by scaled filters formed by scaling basic filters with steep attenuation characteristics. Since the attenuation degree in the stopband can be increased, a filter with a high noise reduction capability can also be implemented according to the present invention. Since the number of taps (multipliers) can be reduced greatly in comparison with conventional filters, a low-cost filter can be provided according to the present invention.

Brief Description of the Drawings

[0025]

Fig. 1 is a structural view of a filter.
Fig. 2 is a structural view of a scaled element filter.
Fig. 3 is a schematic illustration of filter processing.
Fig. 4 is a structural view of a general filter.
Fig. 5 is a view showing the waveform of a conventional impulse response function.
Fig. 6 is a view showing the waveform of an impulse response function.
Fig. 7 is an illustration of a discrete impulse response and filter coefficient values.
Fig. 8 is a view showing an example of the frequency response of a low-pass filter.
Fig. 9 is a structural view of a basic low-pass filter.
Fig. 10 is an illustration of the discrete impulse response and filter coefficient values of a basic high-pass filter.
Fig. 11 is a view showing an example of the frequency response of the basic high-pass filter.
Fig. 12 is an illustration of scaling by M.
Fig. 13 is an illustration of a scaled filter.
Fig. 14 is a view illustrating a cascade connection configuration and showing an example of its frequency response.
Fig. 15 is an illustration of a band-pass filter.
Fig. 16 is an illustration of a frequency shift of the band-pass filter.
Fig. 17 is an illustration of the relationship between a passband frequency width and scaling factor in the embodiment.
Fig. 18 is a flowchart illustrating a processing procedure in the embodiment.
Fig. 19 is a view showing an example configuration of the band-pass filter according to the embodiment.
Fig. 20 is a structural view of a scaling filter module of the band-pass filter.
Fig. 21 is an illustration of the band-pass filter.
Fig. 22 is a structural view of a basic low-pass filter $L_{0B}(z)$.
Fig. 23 is a structural view of a low-pass filter $L_{pB}(z)$ formed by scaling by p + 1.
Fig. 24 is a structural view of a basic high-pass filter $H_{0B}(z)$.
Fig. 25 is a structural view of a high-pass filter $H_{pB}(z)$ formed by scaling by p + 1.
Fig. 26 is a structural view of a high-pass filter $H_{qB}(z)$ formed by scaling by p + 1.
Fig. 27 shows a filter structural view.
Fig. 28 shows another filter structural view.
Fig. 29 is a frequency characteristics diagram of a filter designed with theoretical values by each method.
Fig. 30 is a frequency characteristics diagram of a filter designed by each method after coefficient quantization.
Fig. 31 shows the evaluation of a filter according to the present invention in comparison with required specifications.
Fig. 32 is a structural view of the hardware of a design system according to the embodiment.
Fig. 33 is a circuit diagram of a filter G.

Embodiments for Carrying Out the Invention

[0026] To give a better understanding of the present invention, an embodiment will be described.

1. Filter configuration

[0027] Fig. 3 is an outlined illustration of filter processing.

[0028] A non-recursive digital filter is also called a finite impulse response (FIR) filter since the impulse response is expressed by a finite number of pulses.

[0029] Generally, an FIR filter can be obtained from an input-output relationship as shown below, by multiplying the past and current input signals X(m) by a predetermined coefficient (filter coefficient) C(k), as shown in Fig. 3 , where X(m) is an input signal and Y(m) is an output signal.

[Expression 4]

$$Y(m) = \sum_{k=0}^{P} C(k)x(m-k) = \sum_{k=0}^{P} C(k)X(m)z^{-k}$$

here,

$$L(z) = \sum_{k=0}^{P} C_k z^{-k}$$

The second expression is referred to as an FIR low-pass filter, where $C_k = C(k)$.

[0030] Fig. 4 is a structural view of the filter.

[0031] The filter includes delay elements 31-m to 31-(m-p), multipliers 32-0 to 32-p having filter coefficients C_0 to C_p, respectively, and an adder 33.

[0032] A high-pass filter is expressed by the expression below.

[Expression 5]

$$H(z) = \sum_{k=-\infty}^{\infty} C_k (-z)^k$$

[0033] The filter coefficients C_k of the low-pass filter and high-pass filter given above are derived from the impulse response function.

[0034] Fig. 5 is a view showing the waveform of a conventional impulse response function.

[0035] A SINC function shown in Fig. 5 has been applied as the conventional impulse response function, and the function value at the middle point of the distance between samples has been utilized as a filter coefficient C_k. The SINC function, however, is a function of infinite interval, and an infinite number of filter coefficients C_k are required ideally. An

actual filter must have a finite number of coefficients, but a very large number of coefficients are required to ensure certain precision in spite of an error caused by the reduction of coefficients.

[0036] Fig. 6 is a view showing the waveform of an impulse response function according to an example.

[0037] This example uses an impulse response function having a finite support as shown in Fig. 6. Given below is an example of the function obtained by shifting the B-spline function at intervals of a half of the distance between sample points and performing synthesized approximation.

[Expression 6]

$$\psi(t) = -\frac{h}{2}\phi\left(t + \frac{h}{2}\right) + 2h\phi(t) - \frac{h}{2}\phi\left(t - \frac{h}{2}\right)$$

$$\phi(t) \triangleq \int_{-\infty}^{\infty} \left\{\frac{\sin(\pi f h)}{\pi f h}\right\}^3 e^{j2\pi f t} df$$

where

    h: Sampling time interval
    f: Frequency

[0038] According to Japanese Unexamined Patent Application Publication No. 2001-084244, the function given above can be approximated by piecewise polynomials shown in the equation given below.

[Expression 7]

$$\phi(t) = \begin{cases} -\dfrac{t^2}{4} - t - 1 & -2 \le t \le -\dfrac{3}{2} \\[2mm] \dfrac{3t^2}{4} + 2t + \dfrac{5}{4} & -\dfrac{3}{2} \le t \le -1 \\[2mm] \dfrac{5t^2}{4} + 3t + \dfrac{7}{4} & -1 \le t \le -\dfrac{1}{2} \\[2mm] -\dfrac{7t^2}{4} + 1 & -\dfrac{1}{2} \le t \le \dfrac{1}{2} \\[2mm] \dfrac{5t^2}{4} - 3t + \dfrac{7}{4} & \dfrac{1}{2} \le t \le 1 \\[2mm] \dfrac{3t^2}{4} - 2t + \dfrac{5}{4} & 1 \le t \le \dfrac{3}{2} \\[2mm] -\dfrac{t^2}{4} + t - 1 & \dfrac{3}{2} \le t \le 2 \end{cases}$$

[0039] Fig. 7 is an illustration of a discrete impulse response and filter coefficient value of the example.

[0040] A black dot in the figure indicates a sample point (singular point) in the impulse response function with finite support. When the shown function $\psi(t)$ is given as the waveform of impulse response having a finite support, if the value of the black dot on the vertical axis is given as the filter coefficient (tap coefficient) C(k) with the filter delay time treated as a half of the sampling time width h, the waveform of the lowest-degree impulse response is reproduced. A filter having this filter coefficient C(k) is called a basic low-pass filter L_0.

[0041] With the shown function ψ, the basic low-pass filter is expressed as follows.

[0042] In the equation given above, the value of $\psi(t)$ at connection points at intervals of h/2 in each section as shown in Fig. 7 is taken as the coefficient C(k), with the following settings being taken into account

$$C(0) = \psi(-1.5) = -1/16$$

$$C(1) = \psi(-1.0) = 0$$

$$C(2) = \psi(-0.5) = 9/16$$

$$C(3) = \psi(0) = 1$$

$$C(4) = \psi(0.5) = 9/16$$

$$C(5) = \psi(1.0) = 0$$

$$C(6) = \psi(1.5) = -1/16$$

The basic low-pass filter $L_0(z)$ is expressed by the following equation.

[Expression 8]

$$L_0(z) = \sum_{k=0}^{6} C(k)z^{-k}$$
$$C(k) = \psi((k-3)\tau)$$
$$\tau = h/2$$

From the definition given above, $L_0(z)$ is expressed as follows.

[Expression 9]

$$L_0(z) = \sum_{k=0}^{6} C(k)z^{-k} = C(0) + C(1)z^{-1} + C(2)z^{-2} + C(3)z^{-3} + C(4)z^{-4} + C(5)z^{-5} + C(6)z^{-6}$$

$$= -\frac{1}{16} + \frac{9}{16}z^{-2} + z^{-3} + \frac{9}{16}z^{-4} - \frac{1}{16}z^{-6}$$

The frequency characteristics of the basic low-pass filter are expressed by the equation given below.

[Expression 10]

$$L_0(f) = e^{-j6\pi f}\left(\frac{C(3)}{2} + \sum_{k=0}^{2} C(k)\cos(2k\pi f)\right)$$

[0043]   Fig. 8 is a view showing an example of the frequency response of the low-pass filter of the embodiment. The figure shows the characteristics expressed by the equation given above.
[0044]   Fig. 9 is a structural view of the basic low-pass filter of the embodiment.
[0045]   An example of the arithmetic circuit of the basic low-pass filter L_0 is configured as shown in Fig. 9.
[0046]   The basic low-pass filter includes delay elements 91-1, 91-2, 91-3, 91-4, 91-5, and 91-6, multipliers 92-1, 92-2, and 92-3, and adders 93-1, 93-2, 93-3, and 93-4. The filter coefficients of the multipliers 92-1, 92-2, and 92-3 are - 1/16, 9/16, and 1, respectively. The multiplier 92-3 can be omitted because the coefficient is 1.
[0047]   Based on the low-pass filter $L_0(z)$, a basic high-pass filter $H_0(z)$ can be expressed by the following equation.

[Expression 11]

$$H_0(z) = -L_0(-z) = -\sum_{k=0}^{6} b(k)z^{-k}$$
$$b(k) = (-1)^k C(k)$$

[0048] Fig. 10 is an illustration of the discrete impulse response and filter coefficient value of the basic high-pass filter of the example.

[0049] The coefficient b(k) in the equation given above is obtained by inverting the sign of the coefficient C(k) alternately as shown in Fig. 10.

[0050] Fig. 11 is a view showing an example of the frequency response of the basic high-pass filter.

[0051] The high-pass filter $H_0(z)$ has frequency characteristics shown in Fig. 11 and can be expressed by the equation given below.

[Expression 12]

$$H_0(f) = e^{-j6\pi f}\left(\frac{C(3)}{2} - \sum_{k=0}^{2}(-1)^k C(k)\cos(2k\pi f)\right)$$

[0052] The reason for using the impulse response functions such as the Fluency DA function is that the local support allows filter coefficients to be used in the range of finite numbers and that the even function (linear phase) can implement a linear phase. In addition, the maximally flat characteristics can solve a ripple problem in the passband.

2. Scaled filter

(1) Scaling

[0053] In comparison with the basic filters described above, a filter with a time base multiplied by M + 1, or a filter having a delay time interval multiplied by 1/(M + 1) will now be considered. In other words, a filter scaled in the frequency domain as given by the equation given later is defined. $L_M(z)$ and $H_M(z)$ are obtained by scaling $L_0(z)$ and $H_0(z)$, respectively by M + 1, where M is referred to as a scaling factor.

[0054] On the basis of the basic low-pass filter $L_0(z)$ and basic high-pass filter $H_0(z)$, described above, filters scaled in the frequency domain will be defined next.

[0055] A low-pass filter $L_M(z)$ and a high-pass filter $H_M(z)$ obtained by scaling the basic low-pass filter $L_0(z)$ and the basic high-pass filter $H_0(z)$ by M + 1 can be expressed respectively by the following equations.

[Expression 13]

$$L_M(z) = L_0(z^{M+1})$$
$$H_M(z) = H_0(z^{M+1})$$

[0056] Fig. 12 is an illustration of scaling by M

[0057] As the equations given above indicate, scaling in the frequency domain corresponds to upsampling in the time domain.

[0058] The low-pass filter $L_M(z)$ and high-pass filter $H_M(z)$ are obtained by upsampling M times in the time domain as shown in Fig. 12, which corresponds to insertion of M zeros between sample points.

[0059] Fig. 13 is an illustration of scaled filters.

[0060] Examples of the frequency characteristics of the low-pass filter $L_M(z)$ are shown in Fig. 13.

[0061] The figure shows the characteristics of filters L_1 and L_2 obtained by scaling the basic low-pass filter L_0 by 2 and 3, respectively.

[0062] The basic filter L_0 is scaled to the filter L_1 as shown in the figure, and the filter L_1 is scaled to the filter L_2 as shown in the figure. A filter L_3 and subsequent filters are obtained likewise by scaling.

[0063] The configuration of an example arithmetic circuit of $L_M(z)$ will now be described. A filter $L_p(z)$ formed by scaling by p + 1 will be described below.

[0064] Fig. 2 is a structural view of a scaled element filter. The figure shows a filter formed by scaling the basic filter

by p + 1.

[0065] The basic low-pass filter has delay elements 21-1, 21-2, 21-3, 21-4, 21-5, and 21-6 that cause a delay of p + 1 samples, multipliers 22-1, 22-2, and 22-3, and adders 23-1, 23-2, 23-3, and 23-4. The filter coefficients of the multipliers 22-1, 22-2, and 22-3 are -1/16, 9/16, and 1, respectively. The multiplier 22-3 can be omitted because the coefficient is 1.

[0066] This filter has an arithmetic circuit implementing the following equation.

[Expression 14]

$$Lp(z) = \sum_{k=0}^{6} C(k)(z^{p+1})^{-k} = \sum_{k=0}^{6} C(k)z^{-k(p+1)}$$

3. Cascade connection of filters

[0067] Fig. 1 is a structural view of an example filter having a cascade connection of low-pass filters.

[0068] The filter has n low-pass filters $L_{pk}(z)$ connected in cascade. The low-pass filter $L_{pk}(z)$ is an element filter formed by scaling the basic low-pass filter $L_0(z)$. Scaling is to define a filter by multiplying the basic filter by pk + 1 in the frequency domain or time domain.

[0069] The shown configuration includes just scaled filters based on the low-pass filter. However, a configuration that includes high-pass filters alone or both high-pass filters and low-pass filters, which will be described later, is also possible.

[0070] Fig. 14 is an illustration of variations in frequency characteristics caused by cascade connections of the basic filters.

[0071] By connecting scaled low-pass filters $L_{pk}(z)$ in cascade as shown in Fig. 1, filters having the frequency characteristics shown in Fig. 14, for example, can be implemented. The cascade connection of scaled filters makes it possible to narrow the passband width and to lower the stopband sufficiently and creates other new characteristics.

[0072] In the figure, the filter of a single stage is the basic filter L_0; the filter of two stages includes the basic filter L_0 and the filter L_1 connected in cascade; the filter of three stages includes the basic filter L_0, the filter L_1, and the filter L_2 connected in cascade. As shown in the figure, the cascade connection of scaled filters can narrow the passband width and can also lower the stopband.

[0073] The examples described above include just low-pass filters, but generally, a variety of filters X(z) can be configured by combining low-pass filters and high-pass filters. The filter X(z) can be expressed generally by the equation below.

[Expression 15]

$$X(z) = X_0(z)\prod_{p=1}^{P} L_p^{\alpha_r}(z) \cdot \prod_{q=1}^{Q} H_q^{\beta_q}(z), \, \alpha_p, \beta_q \in N_0$$

$$L_p(z) = L_0(z^{p+1}), \quad H_q(z) = H_0(z^{q+1})$$

[0074] L_0 is the basic low-pass filter (mother filter).

[0075] H_0 is the basic high-pass filter (mother filter).

[0076] p and q are scaling values (scaling factors). In other words, a scaled filter is formed by scaling the frequency by p + 1 or q + 1.

[0077] α_p is the number of times an identical scaled filter is connected, or the number of connected scaled low-pass filters formed by scaling by p + 1 (exponential value).

[0078] β_q is the number of times an identical scaled filter is connected, or the number of connected scaled high-pass filters formed by scaling by q + 1 (exponential value).

[0079] $X_0(z)$ is a highest-order filter $L_P(z) = L_0(z^{P+1})$ or $H_Q(z) = H_0(z^{Q+1})$ that has a target bandwidth (frequency width up to the attenuation of -3 dB, for example). The highest degrees p and q are indicated as P and Q.

[0080] The relationship between the target bandwidth f_p and the bandwidth f_0 of the mother filter L_0 or H_0 is expressed by the following equation.

$$f_p = f_0/(p + 1)$$

[0081] Advantages of the cascade connection include the reduction of the bandwidth in the passband and the attenuation of an unnecessary-signal band in the stopband. Making use of these advantages, the embodiment configures a filter with a reduced number of taps.

4. Filter configuration method

[0082] An example filter configuration method will be described next.
[0083] Fig. 15 is an illustration of a band-pass filter. The figure shows the characteristics of the filter to be implemented as a design target.
[0084] The requirements of the filter to be implemented include the following items. The corresponding symbols are shown in Fig. 15.

> (1) Sampling frequency fs
> (2) Center frequency fc
> (3) Cutoff frequencies (frequencies at which the attenuation is 3 dB) f_1, f_{-1}
> (4) Passband frequency width (width of -3 dB band) Δfb
> (5) Stopband frequencies f_2, f_{-2}
> (6) Stopband attenuation Ad (dB)

[0085] In a signal pass filter, the sampling frequency fs is the repetition frequency of the signal input to the filter, and the center frequency fc is the frequency at the center of the filter, extracted from the frequency components included in the signal. Generally, fc/fs < 0.5, according to the Nyquist theorem. The cutoff frequency f_1 is the highest frequency at which the signal passes. The symmetrical frequency f_{-1} about the center frequency fc is the lowest frequency at which the signal passes. In the example described here, the attenuation of the passing signal at the cutoff frequencies f_1 and f_{-1} is -3 dB, in comparison with the original signal. An appropriate value other than -3 dB can be specified as the reference value. The attenuation A is defined with the magnitude X of the input signal and the magnitude Y of the filter output signal at the cutoff frequency, as indicated in the equation given below.

[Expression 16]

$$A = 20 * \log_{10} \frac{|Y|}{|X|} = -3 (dB)$$

$$\frac{|Y|}{|X|} = 10^{-3/20} \cong 0.708 \cong \frac{1}{\sqrt{2}}$$

[0086] The input signal is attenuated to about 70.8% (1/√2) at the cutoff frequencies f_1 and f_{-1}, and the result is output. This means that the amplitude energy of the signal is attenuated to a half.
[0087] The signal passband is the range between the cutoff frequencies f_1 and f_{-1}, and the frequency width Δfb(m) thereof is given by the following equation.

$$\Delta fb(m) = f_1 - f_{-1} = (f_1 - fc) * 2$$

where Δfb(m) is the passband of the filter formed by scaling by m + 1.
[0088] The stopband frequencies f_2 and f_{-2} indicate that the frequency components above f_2 or below f_{-2} of the input signal are cut off. With the corresponding attenuation degree Ad (such as -80 dB) at the stopband frequencies, the transition band attenuation characteristics from the attenuation level (such as -3 dB) at the cutoff frequencies f_1 and f_{-1} to the attenuation degree Ad will be given. An appropriate value other than -80 dB can be specified as the reference value. Ideally speaking, the cutoff frequency should be equal to the stopband frequency. Since that is difficult to realize in actual filters, a transition band is provided. When the attenuation degree Ad in the stopband is -80 dB, the ratio between the input signal X and the filter output signal in magnitude is 0.01% as given below.

[Expression 17]

$$Ad = 20 * \log_{10} \frac{|Y|}{|X|} = -80(dB)$$

$$\frac{|Y|}{|X|} = 10^{-80/20} \cong 1/10000$$

[0089] Based on the preconditions described above, the filter configuration in this embodiment will be described next.

[0090] Fig. 16 is an illustration of a frequency shift of a band-pass filter.

[0091] To simplify the design, a filter B(z) obtained by shifting a band-pass filter B_B(z) with the center frequency fc (2πfc = ωc) to the point of frequency zero will be considered, as shown in Fig. 16.

[Expression 18]

$$B_B(\omega) = B(\omega - \omega_c) + B(\omega + \omega_c)$$

here,

$$B_B(z) = \sum h_B(k) z^{-k}$$
$$B(z) = \sum h(k) z^{-k}$$

The frequency shift theorem gives the following.

[Expression 19]

$$h_B(k) = h(k) e^{j\omega_c k} + h(k) e^{-j\omega_c k}$$
$$= h(k) \left(e^{j\omega_c k} + e^{-j\omega_c k} \right) = 2h(k) \cos(\omega_c k)$$

The filter B_B(z) is converted by multiplying the coefficient of the filter B(z) by $2\cos(\omega_c k)$.

[0092] This means that when the filter B(z) is designed, the requirements should be converted as given below.

[0093] Fig. 21 is an illustration of the band-pass filter of the embodiment. The figure shows the characteristics of the target filter to be implemented.

    (1) Sampling frequency fs
    (2) Center frequency 0
    (3) Cutoff frequencies (frequencies at which the attenuation is 3 dB, for example) fa = Δfb/2, -Δfb/2 = -fa
    (4) Passband frequency width (width of -3 dB band, for example) Δfb
    (5) Stopband frequencies fb = $(f_2 - f_{-2})/2$, $-(f_2 - f_{-2})/2$ = -fb
    (6) Stopband attenuation Ad (dB) (-80 dB, for example)

[0094] The cascade connection of filters will be described next.

[0095] The general expression is the equation given below.

[Expression 20]

$$B(z) = B_0(z) \prod_{p=0}^{P} L_p^{\alpha_p}(z) \cdot \prod_{q=0}^{Q} H_q^{\beta_q}(z), \quad \alpha_p, \beta_q \in N_0$$
$$B_0(z) = L_M(z) \quad or \quad H_M(z)$$
$$L_p(z) = L_0\left(z^{p+1}\right), H_q(z) = H_0\left(z^{q+1}\right)$$

[0096] $L_p(z)$ is a low-pass filter formed by scaling the basic low-pass filter $L_0(z)$ by p + 1.

[0097] $H_q(z)$ is a high-pass filter formed by scaling the basic high-pass filter $H_0(z)$ by q + 1.

[0098] The maximum scaling value M is determined by the frequency width of the passband, and the scaling values P and Q are smaller than M. α_p and β_q are identical $L_p(z)$ and $H_q(z)$ raised to some power and indicate cascade connections of α_p and β_q filters.

[0099] A filter $B_0(z)$ of low-pass filter type is $L_M(z)$, and a filter $B_0(z)$ of high-pass filter type is $H_M(z)$.

[0100] Fig. 17 is an illustration of the relationship between the passband frequency width and scaling factor in this embodiment.

[0101] The maximum scaling value M is determined by a required passband frequency width Δfb. The passband width Δfb narrows with an increase in scaling value, as clearly indicated in Fig. 13, and the relationship between Δfb and the scaling value p (or q) is as shown in Fig. 17. The attenuation degree is the ratio of the output Y(m) to the input x(m), or Y(m)/x(m) = Lp(z). The attenuation degree is calculated in decibels (dB) as follows. The Δfb-p relationship satisfies the following.

[Expression 21]

$$Y(m) = \sum_{k=0}^{P} C(k)x(m-k) = \sum_{k=0}^{P} C(k)x(m)z^{-k}$$
$$= L_p(z)x(m)$$

$$A = 20\log_{10}\left|\frac{Y(m)}{x(m)}\right| = 20\log_{10}\left|L_p(\Delta f_b)\right| = -3[dB]$$

The relationship between the passband width Δf_{b0} (= $\Delta f_{b0}(0)$) of the basic low-pass filter $L_0(z)$ and the passband width $\Delta f_b(p)$ of a low-pass filter formed by scaling by p + 1 can be expressed by the following equations.

[Expression 22]

$$p = \frac{\Delta f_{b0}}{\Delta f_b(p)} - 1$$
$$\Delta f_{b0} = \Delta f_b(0)$$

[0102] In this embodiment, a low-pass filter $L_M(z)$ or a high-pass filter $H_M(z)$, where p, q = M, is used as an initial value, for example. Then, p or q is selected in descending order, for example; an optimum combination of p, q, α_p, and β_q is selected; and B(z) is designed.

[0103] Lp(z) or Hp(z) is selected in accordance with whether the attenuation degree at the stopband frequency fb is equal to or smaller than the required value Ad (such as -80 dB). In addition, a filter with the attenuation degree required at the cutoff frequency fa much lower than the value (-3 dB, for example) will not be selected.

[0104] The filter $B_B(z)$ to be selected according to the criteria given above is expressed as follows.

[Expression 23]

$$B_B(z) = B_0(z)\prod_{p=0}^{P} L_{pB}^{\alpha_p}(z) \cdot \prod_{q=0}^{Q} H_{qB}^{\beta_q}(z) = \sum_{k=1}^{N} h_B(k)z^{-k}$$
$$h_B(k) = 2h(k)\cos(\omega_c k)$$
$$B(z) = \sum_{k=1}^{N} h(k)z^{-k}$$

[0105] By multiplying the coefficients of the filters $L_p(z)$ and $H_q(z)$ by $2\cos(\omega_c k)$, $L_{pB}(z)$ and $H_{qB}(z)$, which form $B_B(z)$, are obtained.

[0106] Fig. 19 is a view showing an example configuration of the band-pass filter according to the embodiment.

[0107] The band-pass filter is designed in the method described above in accordance with the configuration shown in the figure.

[0108] The filter includes a filter $B_0(z)$ having the maximum scaling value, a filter containing low-pass filters L_{0B} to L_{pB} formed by scaling by 0B to p_B, respectively, connected in α_p stages in cascade, and a filter containing high-pass filters H_{0B} to H_{qB} formed by scaling by 0B to q_B, respectively, connected in β_q stages.

[0109] Fig. 20 is a structural view of a low-pass filter $L_{pB}(z)$ formed by scaling by p + 1.

[0110] The low-pass filter includes delay elements 201-1, 201-2, 201-3, 201-4, 201-5, and 201-6, which cause a delay of p + 1 samples, multipliers 202-1, 202-2, and 202-3, and adders 203-1, 203-2, and 203-3, and multipliers 205-1, 205-2, and 205-3. The filter coefficients of the multipliers 202-1, 202-2, and 202-3 are -1/16, 9/16, and 1, respectively. The coefficients of the multipliers 205-1, 205-2, 205-3 are $2(1 + \cos(6\omega_c))$, $2(\cos(2\omega_c) + \cos(4\omega_c))$, and $2\cos(3\omega_c)$, respectively. The multiplier 202-3 can be omitted because the coefficient is 1. Pairs of multipliers 202-1 and 205-1, multipliers 202-2 and 205-2, multipliers 203-3 and 205-3 can be configured as integrated multipliers.

[0111] Fig. 26 is a structural view of a high-pass filter $H_{qB}(z)$ formed by scaling by q + 1.

[0112] The high-pass filter includes delay elements 201-1, 201-2, 201-3, 201-4, 201-5, and 201-6, which cause a delay of q + 1 samples, multipliers 202-1, 202-2, and 202-3, adders 203-1, 203-2, and 203-3, and multipliers 205-1, 205-2, and 205-3. The filter coefficients of the multipliers 202-1, 202-2, and 202-3 are 1/16, -9/16, and 1, respectively. The coefficients of the multipliers 205-1, 205-2, and 205-3 are $2(1 + \cos(6\omega_c))$, $2(\cos(2\omega_c) + \cos(4\omega_c))$, and $2\cos(3\omega_c)$, respectively. The multiplier 202-3 can be omitted because the coefficient is 1. Pairs of multipliers 202-1 and 205-1, multipliers 202-2 and 205-2, multipliers 203-3 and 205-3 can be configured as integrated multipliers.

5. Filter circuit configuration

[0113] Other filter circuit configurations will be described next.

[0114] Fig. 22 is a structural view of the basic low-pass filter $L_{0B}(z)$.

[0115] The basic low-pass filter $L_{0B}(z)$ expressed by the equation given above has the shown circuit configuration.

[0116] The basic low-pass filter includes delay elements 21-1, 21-2, 21-3, 21-4, 21-5, and 21-6, which cause a delay of a single sample, multipliers 22-1, 22-2, 22-3, 22-4, and 22-5, and adders 23-1, 23-2, 23-3, and 23-4. The filter coefficients of the multipliers 22-1, 22-2, 22-3, 22-4, and 22-5 are -1/8, $9\cos(2\omega_c)/8$, $2\cos(3\omega_c)$, $9\cos(4\omega_c)/8$, and $-\cos(6\omega_c)/8$, respectively.

[0117] Fig. 23 is a structural view of the low-pass filter $L_{pB}(z)$ formed by scaling by p + 1.

[0118] As shown in the figure, the low-pass filter $L_{pB}(z)$ formed by scaling by p + 1 has the same structure as L_{0B}, but Z^{-1} is replaced with $Z^{-(p+1)}$.

[0119] The low-pass filter formed by scaling by p + 1 includes blocks 31-1, 31-2, 31-3, 31-4, 31-5, and 31-6 having p + 1 delay elements, multipliers 32-1, 32-2, 32-3, 32-4, and 32-5, and adders 33-1, 33-2, 33-3, and 33-4. The filter coefficients of the multipliers 32-1, 32-2, 32-3, 32-4, and 32-5 are -1/8, $9\cos(2\omega_c)/8$, $2\cos(3\omega_c)$, $9\cos(4\omega_c)/8$, and $-\cos(6\omega_c)/8$, respectively.

[0120] In the figure, the delay elements 31-1 to 31-6 each have p + 1 delay elements. Each delay element block may have any number of delay elements, if they cause a delay of p + 1 samples.

[0121] Fig. 24 is a structural view of the basic high-pass filter $H_{0B}(z)$.

[0122] The basic high-pass filter $H_{0B}(z)$ has the shown circuit configuration.

[0123] The basic high-pass filter has delay elements 41-1, 41-2, 41-3, 41-4, 41-5, and 41-6 that cause a delay of a single sample, multipliers 42-1, 42-2, 42-3, 42-4, and 42-5, and adders 43-1, 43-2, 43-3, and 43-4. The filter coefficients of the multipliers 42-1, 42-2, 42-3, 42-4, and 42-5 are 1/8, $-9\cos(2\omega_c)/8$, $2\cos(3\omega_c)$, $-9\cos(4\omega_c)/8$, and $\cos(6\omega_c)/8$, respectively.

[0124] Fig. 25 is a structural view of the high-pass filter $H_{qB}(z)$ formed by scaling by q + 1.

[0125] The high-pass filter $H_{qB}(z)$ formed by scaling by q + 1 has the same structure as the filter H_0 as shown in the figure, but Z^{-1} is replaced with $Z^{-(q+1)}$.

[0126] The high-pass filter formed by scaling by q + 1 includes blocks 51-1, 51-2, 51-3, 51-4, 51-5, and 51-6 that have q + 1 delay elements, multipliers 52-1, 52-2, 52-3, 52-4, and 52-5, and adders 53-1, 53-2, 53-3, and 53-4. The filter coefficients of the multipliers 52-1, 52-2, 52-3, 52-4, and 52-5 are 1/8, $-9\cos(2\omega_c)/8$, $2\cos(3\omega_c)$, $-9\cos(4\omega_c)/8$, and $\cos(6\omega_c)/8$, respectively.

[0127] In the figure, the delay elements 51-1 to 51-6 each have q + 1 delay elements. Each delay element block may have any number of delay elements, if they cause a delay of q + 1 samples.

6. Hardware

[0128] Fig. 32 is a structural view of the hardware of a design system.

[0129] The design system includes a processing unit 1, which is a central processing unit (CPU), an input unit 2, an

output interface unit 3, a display unit 4, a storage unit 5, and a filter circuit 6. The processing unit 1, input unit 2, output interface unit 3, display unit 4, and storage unit 5 are connected by a star, bus, or another connection means.

[0130] The processing unit 1 accesses the storage unit 5 and executes filter design processing.

[0131] The storage unit 5 includes a filter configuration file 51 that stores design specifications and requirements (such as conditions (range, value, ratio, etc) of the passband and stopband, upper limits of the numbers α_p and β_q of filters connected in cascade), a basic filter characteristics file 52 that stores the characteristics of the basic low-pass filter, the low-pass filters L_0 to L_p formed by scaling the basic low-pass filter, the low-pass filter L_{pB} formed by multiplying the filter coefficient of the filter L_p by $2\cos(\omega_c k)$, the basic high-pass filter, the high-pass filters H_0 to H_q formed by scaling the basic high-pass filter, and the high-pass filter H_{qB} formed by multiplying the filter coefficients of filter H_q by $2\cos(\omega_c k)$, and a filter output file 53 that stores data and parameters (such as the shift amount B, maximum scaling value M, filter coefficients p, q, α_p, β_q of multipliers) for determining a designed filter configuration and intermediate results, final results, and so on by the processing unit 1, such as designed filter characteristics. The processing unit 1 can output these data items (for example, data and parameters for determining the filter configuration, such as the filter coefficients of multipliers) through the output interface unit 3 to the filter circuit 6. The filter circuit 6 is provided by software or hardware and implements a filter having the specified characteristics in accordance with the data items set up through the output interface unit 3 by the processing unit 1.

7. Design flowchart

[0132] Fig. 18 is a flowchart illustrating a processing procedure in the embodiment.

[0133] The design system configures a filter through steps S01 to S07 described below.

(1) Step S01

[0134] The processing unit 1 specifies beforehand desired characteristics data to be given as design specifications and requirements, including the center frequency fc, cutoff frequency f1, stopband frequency f2, passband attenuation degree A, stopband attenuation degree Ad, and sampling frequency fs, for example, by storing them in the storage unit 5 (filter configuration file 51) through operations from the input unit 2 or by reading the values stored in the storage unit 5 (filter configuration file 51) beforehand.

(2) Step S03

[0135] The processing unit 1 reads the values of characteristics data specified in step S01, from the storage unit 5 (filter configuration file 51), and calculates the shift amount B, which shifts the center frequency fc to the origin O, by the equation given below, in accordance with the input characteristics data.

[0136] B = [fc/fs] (The square brackets indicate that fractions are rounded off to the whole numbers.)

[0137] The processing unit 1 stores the obtained shift amount B in the storage unit 5 (filter output file 53).

[0138] The processing unit 1 also converts the center frequency fc, cutoff frequency f1, and stopband frequency f2 of the input characteristics data to the center frequency 0, cutoff frequency f_a, and stopband frequency f_b, respectively, in accordance with the shift amount B, and stores the values in the storage unit 5 (filter output file 53).

[0139] The processing unit 1 reads the values of the characteristics data specified in step S01, from the storage unit 5 (filter configuration file 51), and calculates the scaling value (scaling factor) M of the highest degree on the basis of the input characteristics data.

[0140] To obtain the scaling value M of the highest-degree (maximum scaling value), the processing unit calculates the passband width Δfb (M) of the filter formed by scaling by M + 1, as given below, for example.

$$\texttt{Passband width } \Delta\texttt{fb(M)} = \texttt{(f1 - fc)*2}$$

[0141] Then, the processing unit 1 obtains the maximum scaling value M from the passband width Δfb (M) by referencing a scaling curve stored in advance in the storage unit 5 (basic filter characteristics file 52) or by calculating the equation of a scaling curve.

$$\texttt{M = P = f}_{-3}\texttt{(}\Delta\texttt{fb(M)) = [}\Delta\texttt{fb(0)/}\Delta\texttt{fb(M) - 1]}$$

$f_{-3}(\Delta fb(M))$: Scaling curve that causes an attenuation of 3 dB

[ ]: The square brackets indicate that fractions are rounded off to the whole numbers.

Δfb(0): Passband width of the basic filter

Δfb(M): Passband width of the filter formed by scaling by M + 1

[0142]   The scaling curve of -3 dB is shown in this example, but any other scaling curve of an appropriate value may be determined in advance with respect to the cutoff frequency or the passband frequency width or may be calculated and specified by the processing unit 1.

[0143]   The processing unit 1 may allow the maximum scaling value M to be determined beforehand by the input unit 2 or storage unit 5.

[0144]   The processing unit 1 stores the calculated maximum scaling value M in the storage unit 5 (filter output file 53) .

(3) Step S05

[0145]   The processing unit 1 reads from the storage unit 5 (filter configuration file 51 and filter output file 53) the values specified in step S01 and the maximum scaling value M calculated in step S03. The processing unit 1 also selects from the storage unit 5 an optimum combination of p, q, α_p, and β_q in accordance with the filter combination as given by the equation below.

[Expression 24]

$$B(z) = B_0(z)\prod_{p=0}^{P} L_p^{\alpha_p}(z) \cdot \prod_{q=0}^{Q} H_q^{\beta_q}(z), \qquad \alpha_p, \beta_q \in N_0$$

$$B_0(z) = L_M(z) \quad or \quad H_M(z)$$

$$L_p(z) = L_0(z^{p+1}), H_q(z) = H_0(z^{q+1})$$

[0146]   The processing unit 1 reads from the storage unit 5 (basic filter characteristics file 52) the characteristics of the low-pass filter L_p and high-pass filter H_q determined by p and q, calculates the characteristics of B(z) on the basis of the equation given above, and selects values conforming to the passband and stopband design specifications on a round-robin basis. The processing unit 1 may further select values that minimize the number of multipliers.

[0147]   The selection of $L_p(z)$ and $H_p(z)$ is made, depending on whether the attenuation degree at the stopband frequency fb is lower than or equal to the required value Ad (for example, -80 dB), and, at the same time, depending on it that a filter with an attenuation degree much lower than the required value (for example, -3 dB) at the cutoff frequency fa will not be selected (for example, a filter below a predetermined permissible level will not be selected). Thus, the processing unit 1 selects an optimum combination of p, q, α_p, and β_q.

[0148]   The maximum values of α_p and β_q can be specified beforehand in the storage unit 5 (filter configuration file 51) or the like, and the processing unit 1 can select an optimum combination of p, q, α_p, and β_q by selecting α_p and β_q in descending order from the stored maximum values and selecting p and q in descending order from the maximum scaling value M.

[0149]   Here, the processing unit may make the following judgment or processing.

[0150]   (The maximum attenuation is judged by Xd = X(z) at f = fs/ (p + 1): When Xd decreases at f = f2, α and β are increased; When Xd decreases at f = f1, added Lp and Hp are deleted.)

[0151]   In the description given above, the characteristics of filters formed by scaling the basic low-pass filter and basic high-pass filter are stored beforehand in the storage unit 5 (basic filter characteristics file 52). However, the characteristics of just the basic low-pass filter and the basic high-pass filter may be stored in the basic filter characteristics file 52, and the processing unit 1 may obtain the characteristics of a desired scaled filter through scaling based on the stored characteristics as necessary and use those characteristics.

[0152]   The processing unit 1 stores the obtained p, q, α_p, and β_q in the storage unit 5 (filter output file 53).

(4) Step 07

[0153]   The processing unit 1 reads from the storage unit 5 (filter output file 53) the specification data and parameters for configuring the filter, such as p, q, α_p, and β_q, reads further from the storage unit 5 (basic filter characteristics file 52) the characteristics of the filter L_{0M} or filter H_{0M} corresponding to $B_0(z)$, according to the low-pass filter or the high-pass filter, and the characteristics of the filter L_{pB} and filter H_{qB}, forms the filter $B_B(z)$ in accordance with the equation given below, and displays on the display unit 4 the filter characteristics and the parameters and data that are used or stores

them in the storage unit 5 (filter output file 53), or both displays and stores them.

[Expression 25]

$$B_B(z) = B_0(z) \prod_{p=0}^{P} L_{pB}^{\alpha_p}(z) \cdot \prod_{q=0}^{Q} H_{qB}^{\beta_q}(z) = \sum_{k=1}^{N} h_B(k) z^{-k}$$

$$h_B(k) = 2h(k)\cos(\omega_c k)$$

$$B(z) = \sum_{k=1}^{N} h(k) z^{-k}$$

[0154] The processing unit 1 can also read from the storage unit 5 (filter output file 53) the specification data and parameters for configuring the filter, such as p, q, α_p, and β_q, read from the storage unit 5 (basic filter characteristics file 52) the characteristics of the filters L_{pB} and H_{qB}, and output the values through the output interface unit 3 to the filter circuit 6. The processing unit 1 may use the filter circuit 6 to form the filter $B_B(z)$ in accordance with the read values. The filter circuit 6 can form the filter $B_B(z)$ with a computer simulation or by software. The filter circuit 6 may be configured by hardware to form the filter $B_B(z)$ in accordance with the values output from the output interface unit 3.

8. Method of coefficient quantization expansion

[0155] A quantization expansion method for filter coefficients will be described next.
[0156] The quantization expansion method here means a method of implementing the same characteristics as the impulse response characteristics of the filter designed to include the cascade connection configuration, as described above, by a shown filter circuit configuration.
[0157] Since $L_p(z) = L_0(z^{p+1})$, an impulse signal is input to the filter designed to include the cascade connection given below to obtain the response.

[Expression 26]

$$X(z) = X_0(z) \prod_{p=0}^{P} L_p^{\alpha}(z) \prod_{q=1}^{Q} H_q^{\beta}(z)$$

The impulse response can be calculated by a simulation program such as the well-known MATLAB to obtain the waveform of the impulse response. Alternatively, an expansion of X(z) given above can be derived by a mathematical operation program such as the well-known MATHEMATHICA. The expansion is the impulse response function of X(z).

[Expression 27]

$$X(z) = X_0(z) \sum_{k=0}^{k=K} C_k z^{-k}$$

Therefore, the coefficient C_K of the impulse response can be used as filter coefficients when a filter is configured.
[0158] Fig. 27 shows a filter structural diagram.
[0159] The filter includes multipliers 72-0 to 72-k, delay elements 71-1 to 71-k that cause a delay of a single sample, and adders 73-1 to 73-k.
[0160] For example, the processing unit 1 reads data (filter coefficients, p, q, α_p, β_q, etc.) for configuring the filter from the storage unit 5 (filter output file 53), obtains the impulse response by using the filter circuit 6 or the like, and stores the coefficient C_k of the impulse response in the storage unit 5 (filter output file 53).
[0161] Since the processing unit 1 reads the filter coefficient from the storage unit 5 and uses the filter circuit 6, the coefficients C_k (k + 1 coefficients) can be used as the filter coefficients (tap coefficients) in the shown circuit configuration, and the filter can be configured by using k delay elements z^{-1}.
[0162] Fig. 28 shows another filter structural diagram.
[0163] The filter includes multipliers 82-0 to 82-M, delay elements 81-1 to 81-M, and an adder 83.
[0164] A filter having characteristics designed in accordance with the filter coefficient C_k obtained as described above can be configured as shown in the figure besides the filter configuration described earlier.

9. Comparison between this embodiment and a conventional method

[0165] The method according to this embodiment will be compared with a conventional method below.

[0166] The requirements in the embodiment are as listed below.

Sampling frequency: 1.8 MHz
Center frequency: 450 kHz
-3 dB bandwidth: 6 kHz
-80 dB bandwidth: 12 kHz
Quantization bit rate: 18 bits
Phase characteristics: Linear phase

[0167] Fig. 29 is a frequency characteristics diagram of filters designed with theoretical values by different methods.

[0168] As the left graph shows, the Remez method satisfies the requirements but generates ripples in the passband and stopband.

[0169] The right graph shows that the filter according to the present invention satisfies the specifications and makes a large attenuation in the stopband.

[0170] Fig. 30 is a frequency characteristics diagram of filters designed by the methods after coefficient quantization.

[0171] The left graph clearly shows that the Remez method produces errors throughout the stopband.

[0172] The right graph shows that the filter according to the present invention produces just a few errors in the transition band.

[0173] Fig. 31 shows the characteristics of the filter according to the present invention, evaluated against the requirements.

[0174] As shown in the figure, the filter according to the present invention can be implemented with a much smaller number of taps in comparison with the conventional method.

10. Supplement

[0175] A filter configuration (design) method or filter configuration (design) apparatus or system according to the present invention can be provided by a filter configuration (design) program for executing each procedure on a computer, a computer-readable recording medium having the filter configuration (design) program recorded on it, a program product that includes the filter configuration (design) program and can be loaded into the internal memory of the computer, a server or any other computer that includes the program, and the like.

11. Filters G_{pass}, G_{stop}, G

[0176] Fig. 33 is a circuit diagram of a filter G.

[0177] Fig. 34 is a circuit diagram of G_{pass}.

[0178] Figs. 35 and 36 are expanded views of G_{pass}. Figs. 35 shows a part related to a first term of G_{pass}. Fig. 36 shows a part related a second term of G_{pass}.

[0179] With L_m and H_m described above, the filter passband main part G_{pass} is formed as indicated below.

[0180] G_{pass} is expressed by the equation given below.

[Expression 28]

$$G_{pass} = L_{P,N_P^{(1)}}^{\alpha_P} \left[1 - H_{P,N_P^{(2)}} \right]^{\beta_P}$$

where the superscripts and subscripts of G_{pass} indicate the following:

ω_P, β_P: Exponential value (indicating that identical $L_{P,N1}$ or $[1 - H_{P,N2}]$ is connected α_P or β_P times. Here, $N1 = N_p^{(1)}$ and $N2 = N_p^{(2)}$ for the sake of convenience of description),
$N_p^{(1)}$: Number of H_P ladder connection stages in $L_{P,N1}$ (indicating the number of diagonal connection stages in Fig. 35),
$N_p^{(2)}$: Number of L_P ladder connection stages in $[1 - H_{P,N2}]$ (indicating the number of diagonal connection stages in Fig. 36),
P: Filter formed by scaling the frequency by P + 1.

[0181]   Scaling is expressed by the equation below.

[Expression 29]

$$L_P(f) = L_0((P+1)f), \quad H_P(f) = H_0((P+1)f)$$

[0182]   Fig. 37 is a circuit diagram of G_{stop}.
[0183]   With L_m and H_m described above, the stopband main part G_{stop} is also formed as indicated below.
[0184] G_{stop} is expressed by the equation given below.

[Expression 30]

$$G_{stop} = \left[\prod_{k-1}^{K} L_{p_k, N_k}^{\alpha_k} \right] \cdot \prod_{l-1}^{L} \left[1 - H_{q_l, N_l} \right]^{\beta_l}$$

where

α_k, β_1: Exponential value (indicating that identical $L_{pk,Nk}$ or [1 - $H_{q1,N1}$] is connected α_k or β_1 times),
N_k: Number of H_p ladder connection stages in $L_{pk,Nk}$ (indicating the number of diagonal connection stages (corresponding to $N_p^{(1)}$) in Fig. 35)
N_1: Number of L_p ladder connection stages in [1 - $H_{q1,N1}$] (indicating the number of diagonal connection stages (corresponding to $N_p^{(2)}$) in Fig. 36)
p_k, q_1 : Filter formed by scaling the frequency by p_k + 1 or by q_1 + 1.

[0185]   The modules $L_{pk,Nk}$ and [1 - $H_{q1,N1}$] in the high-pass filter have the same structures as the modules

[Expression 31]

$$L_{P, N_P^{(1)}} \quad \text{or} \quad 1 - H_{P, N_P^{(2)}}$$

in the low-pass filter.
[0186]   Fig. 38 is a structural view of hardware .
[0187]   The hardware includes a processing unit 1, which is a central processing unit (CPU), an input unit 2, an output interface unit 3, a display unit 4, a storage unit 5, and a filter circuit 6. The processing unit 1, input unit 2, output interface unit 3, display unit 4, and storage unit 5 are connected by a star, bus, or another connection means.
[0188]   The storage unit 5 includes a filter configuration file 51 that stores design specifications and requirements (such as conditions (range, value, ratio, etc) of the passband and stopband, the upper limit N_{pass} of the number of multipliers, etc.), a basic filter characteristics file 52 that stores the characteristics of the basic low-pass filter, the scaled low-pass filters (L_0, L_1, ... L_m, ...), the basic high-pass filter, and the scaled high-pass filters (H_0, H_1, ..., H_m, ...), and a filter output file 53 that stores data and parameters (such as the filter coefficients N_{pass}, $N_p^{(1)}$, $N_p^{(2)}$ α_p, β_p, P, N_k, N_1, α_{pk}, β_{p1}, p_k, q_1 of multipliers, etc.) for determining a designed filter configuration, and intermediate results, final results, and so on by the processing unit 1, such as the designed filter characteristics. The processing unit 1 can output these data items (for example, data and parameters for determining the filter configuration, such as the filter coefficients of multipliers) through the output interface unit 3 to the filter circuit 6. The filter circuit 6 is configured by software or hardware and implements a filter having the specified characteristics in accordance with the data items set up through the output interface unit 3 by the processing unit 1.

12. Filter design procedure (low-pass filter)

[0189]   Fig. 39 is a flowchart of a filter design procedure (low-pass filter).
[0190]   This flowchart includes the following steps:

0. Input design specifications (S0)
1. Determine the maximum scaling value p = P (S1)
2. Configure passband section G_{pass} (S2)
3. Configure stopband section G_{stop} (S3)
4. G = G_{pass}·G_{stop} (S4)

[0191] The steps will be described next in detail.

0. Input design specifications (S0)

[0192] The processing unit 1 specifies beforehand design specifications (conditions (range, value, ratio, etc) of the passband and stopband, the upper limit N_{pass} of the number of multipliers, etc.) by storing them in the storage unit 5 (filter configuration file 51) through operations from the input unit 2 or by reading the values stored in the storage unit 5 (filter configuration file 51) beforehand.

[0193] Fig. 40 is a graph showing the cutoff characteristics of a cascade connection model G(f) of a product-sum module.

[0194] In the figure, f_3 and f_{80} indicate the following:

f_3: Frequency that crosses the -3 dB line (point of frequency crossing the -3 dB line first on the right of f = 0)
f_{80}: Frequency that crosses the -80 dB line (point of frequency crossing the -80 dB line first on the left of f = f_s/2)

[0195] The ratio R of the passband frequency to the stopband frequency (0 < R ≤ 1) is defined as R = f_3/f_{80}.

[0196] As the value of R approaches 1, the steepness increases. Steep cutoff characteristics is expressed by the ratio R (0 < R ≤ 1) between f_3 and f_{80}, as in the equation given below, for example.

$$R = f_3/f_{80} \approx 1$$

[0197] The processing unit 1 can specify the design specifications as follows:

Passband: -3 dB or higher at [0, f_3]
Stopband: -80 dB or lower at [f_{80}, 1]
R ≥ R_0 (The obtained filter ratio R is larger than or equal to a predetermined ratio R_0.)

where

$$0 < f_3 < f_{80} < 1$$

$$0 < R_0 \leq 1$$

Step S1. (Determine the maximum value of p)

[0198] In step S1, the processing unit 1 reads specified values such as f_3 from the storage unit 5 (filter configuration file 51) and selects the maximum scaling value p = P in accordance with the given f_3. The processing unit 1 may allow the P value to be predetermined by the input unit 2 or storage unit 5 (filter output file 53).

[0199] The processing unit 1 regards $f_3^{(p)}$ as the -3 dB point of L_p, calculates $f_3^{(0)}$ in advance, and specifies the minimum value of p that satisfies the following as P.

[Expression 32]

$$f_3^{(p)} = \frac{f_3^{(0)}}{1+p} < f_3$$

[0200] The processing unit 1 stores the obtained value of P in the storage unit 5 (filter output file 53).

Step S2. Configure the passband main part

[0201] The processing unit 1 reads the values specified in step S1, such as N_{pass} and R_0, from the storage unit 5 (filter configuration file 51, filter output file 53) and selects a set of values that satisfies the design specifications ($R \geq R_0$, for example) and minimizes the number of multipliers on a round-robin basis of
$(N_p^{(1)}, N_p^{(2)}, \alpha_p, \beta_p) \subset \{0, 1, 2, ...\}$
such that the maximum number of multipliers $N_{pass} \in \{1, 2, ...\}$ specified by

[Expression 33]

$$G_{pass} = L_{P,N_P^{(1)}}^{\alpha_P} \left[1 - H_{P,N_P^{(2)}}\right]^{\beta_P}$$

is not exceeded.

[0202] For example, the processing unit 1 reads the characteristics of the low-pass filter and high-pass filter determined by $N_p^{(1)}$, $N_p^{(2)}$, α_p, β_p, from the storage unit 5 (basic filter characteristics file 52), calculates the characteristics of G_{pass} by the equation given above, and selects the set that satisfies the design specifications ($R \geq R_0$, for example) and minimizes the number of multipliers on a round-robin basis. The processing unit 1 may select a set that also conforms to the design specifications of the passband and stopband.

[0203] However, the following rules of combination are defined.

$(\alpha_p, \beta_p) \neq (0, 0)$
$N_p^{(1)} = 0 \Rightarrow \alpha_p = 0$
$N_p^{(2)} = 0 \Rightarrow \beta_p = 0$

[0204] The numbers of multipliers are obtained as follows.

[Expression 34]

$$\text{The number of multipliers of } \quad L_{P,N_P^{(1)}}^{\alpha_P} = 5 N_P^{(1)} \alpha_P$$

$$\text{The number of multipliers of } \quad \left[1 - H_{P,N_P}^{\alpha_P}\right]^{\beta_P} = 5 N_P^{(2)} \beta_P$$

[0205] Those multipliers include a multiplier whose coefficient is 1. If the multiplier with a coefficient of 1 is excluded, a conditional expression for limiting the number of multipliers up to N_{pass} is given as follows.

$$4 (N_p^{(1)} \alpha_p + N_p^{(2)} \beta_p) \leq N_{pass}$$

[0206] The processing unit 1 stores the obtained $N_p^{(1)}$, $N_p^{(2)}$, α_p, β_p in the storage unit 5 (filter output file 53).

Step S3. Configure stopband section

[0207] The processing unit 1 selects a set of values that satisfies the design specifications ($R \geq R_0$, for example) and minimizes the number of multipliers by calculating

$$G = G_{pass} G_{stop}$$

on a round-robin basis of
$(N_k, N_1, \alpha_{pk}, \beta_{q1}) \subset \{0, 1, 2, ...\}$
when

[Expression 35]

$$G_{stop} = \left[\prod_{k=1}^{K} L_{p_k, N_k}^{\alpha_{p_k}} \right] \cdot \prod_{l=1}^{L} \left[1 - H_{q_l, N_l} \right]^{\beta_{ql}}$$

here,
$p_k, q_l \subset \{0,1,2,...\}, K, L \leq P - 1,$

[0208] For example, the processing unit 1 reads the characteristics of the low-pass filter and high-pass filter determined by N_k, N_1, α_{pk}, β_{ql} from the storage unit 5 (basic filter characteristics file 52), calculates the characteristics of G_{stop} by the equation given above, calculates
$G = Gp_{ass}G_{stop}$
on a round-robin basis, and selects a set that satisfies the design specifications ($R \geq R_0$, for example) and minimizes the number of multipliers. The processing unit 1 may select a set that also conforms to the design specifications of the passband and stopband.
[0209] The processing unit 1 stores the obtained N_k, N_l, α_{pk}, β_{ql} in the storage unit 5 (filter output file 53) .

Step S4. Configure and output G

[0210] The processing unit 1 reads from the storage unit 5 (filter output file 53) the specifications data and parameters for configuring the filter, such as $N_p^{(1)}$, $N_p^{(2)}$, α_p, β_p, N_k, N_l, α_{pk}, β_{ql}, reads also from the storage unit 5 (basic filter characteristics file 52) the characteristics of the basic low-pass filter, the corresponding filters formed by scaling it, the basic high-pass filter, and the corresponding filters formed by scaling it, forms G_{pass} and G_{stop} as expressed by the equations given above, forms the filter G by calculating $G = G_{pass}G_{stop}$, and displays the characteristics of the filter G on the display unit 4 or stores the characteristics in the storage unit 5 (filter output file 53) or both displays and stores them.
[0211] The processing unit 1 reads from the storage unit 5 (filter output file 53) the specifications data and parameters for configuring the filter, such as $N_p^{(1)}$, $N_p^{(2)}$, α_p, β_p, N_k, N_1, α_{pk}, β_{ql}, and outputs the values through the output interface unit 3 to the filter circuit 6. Then, the processing unit 1 may form G_{pass} and G_{stop} by the filter circuit 6 in accordance with the read values and equations given above to form the filter G as $G = G_{pass}G_{stop}$. The filter circuit 6 can form the filter G with a computer simulation or by software. The filter circuit 6 may also form the filter G by hardware, in accordance with the values output from the output interface unit 3.
[0212] In the description given above, the characteristics of filters formed by scaling the basic low-pass filter and the basic high-pass filter are stored in advance in the storage unit 5 (basic filter characteristics file 52). However, the characteristics of just the basic low-pass filter and the basic high-pass filter may be stored in the basic filter characteristics file 52, and the processing unit 1 may perform scaling based on the stored characteristics as needed to obtain and use the characteristics of a designated scaled filter.

13. Example of low-pass filter design

(1) Basic characteristics improvement

[0213] Fig. 45 is an illustration of an example (low-pass filter) in which the characteristics are improved by G(f).
[0214] The figure shows the characteristics of the filter G obtained by forming the filter G(f) given by the equation below, by connecting the low-pass filters $L_0(f)$ and $L_1(f)$ in cascade.

$$G(f) = L_0(f)L_1(f)$$

[0215] The characteristics can be made steeper by connecting element filters in cascade.

(2) Example design through flowchart

[0216] Fig. 41 is an illustration of the specifications of a low-pass filter.

[0217] An example configuration of the low-pass filter will now be given in accordance with the configuration means described above. The desired characteristics specifications are as shown in the figure. The passband and stopband are ranges of normalized frequency.

[0218] The configuration of the filter designed in accordance with the flowchart in this example is expressed as follows.

$$G = L^3_{7.9}L^1_{3.1}L^1_{4.2}L^1_{5.3}(1 - H^1_{1.1})(1 - H^1_{2.1})$$

[0219] In this example, P = 7 is selected; α_7 = 0, 1, 2, 3, 4 and N_7 = 0, 1, 2, 3, and so on are tested; and G_{pass} = $L^3_{7.9}$ is specified.

[0220] Fig. 42 is an illustration of G_{pass} and G_{stop} of the filter G.

[0221] The characteristics diagram of G_{pass} is shown in Fig. 42(a). The characteristics of G_{pass} include a passband section of [0, 0.096], a stopband section of [0.013, 1], and R = 0.738, and 150 multipliers are used.

[0222] In the stopband, the second to fourth side lobes of G_{pass} must be blocked. The third and fourth side lobes are generally blocked by (1 - $H^1_{1.1}$) and (1 - $H^1_{2.1}$) in Fig. 26(b). Here, P $\leq$ 5 is taken, and, as shown in Fig. 26(b), a part near the center of the second side lobe is blocked by $L^1_{3.1}$ and $L^1_{4.2}$, and a part near the right side is blocked by $L^1_{3.1}$ and 1 - $H^1_{2.1}$. To stop a part near the left side, N_5 = 0, 1, 2, ... and α_5 = 0, 1, 2, etc. are tested against p = 5, and (α_5, N_5) = (1, 3) is selected in keeping with the number of multipliers.

[0223] Fig. 43 is a characteristics diagram of filters configured by conventional methods and the disclosed method which is not a subject of the present invention. The least squares method is shown at (a) ; the Remez method is shown at (b) ; and the disclosed method is shown at (c).

[0224] Fig. 44 is a chart comparing the number of multipliers and R of the design methods. The number of multipliers here was obtained when the characteristics satisfied all the specifications for the first time. The disclosed method is superior to the least squares method in terms of both the number of multipliers and R. The Remez method is slightly better than the disclosed method in R, but the number of multipliers is as large as 1142. To check superiority to the Remez method, the number of multipliers was calculated when the R value exceeded R = 0.762 of the Remez method (R = 0.767, here). It was 365.

[0225] This result verifies the validity of the approximation model of steep low-pass filter characteristics, made by a cascade connection of element filters formed of the product-sum modules.

Industrial Applicability

[0226] As the impulse response function described above, an m-th-degree piecewise polynomial can be used, for example. An interpolation function with waveform adjustment parameters or the like can also be used.

[0227] The present invention can be applied to a variety of technologies such as acoustic technologies, video technologies, image technologies, transmission technologies, communication technologies, analog-to-digital conversion and digital-to-analog conversion technologies, compression and decompression technologies, encryption and decryption (decompression) technologies, and filter technologies.

[0228] The present invention can be used in acoustic, image, communication, and other signal processing filters in wide areas.

[0229] The present invention can be used in wide range of apparatuses such as acoustic equipment that includes amplifiers, image apparatuses for moving or still picture processing, communication devices such as mobile phones, controllers, computers, and PCs.

Claims

1. A filter having desired passband characteristics, in particular a desired cut-off frequency (f_1), a desired passband attenuation degree (A) and a desired frequency width $\Delta fb(M)$ of the passband, comprising

   -) a basic low-pass filter (L_0) with passband width $\Delta fb(0)$ and having an impulse response function (ψ) configured by a piecewise polynomial with a finite support and using values of the impulse response function (ψ) at connection points between the polynomials as filter coefficients c(k),
   -) a basic high-pass filter (H_0) with passband width $\Delta fb(0)$ and having filter coefficients b(k) obtained by inverting

the sign of the filter coefficients c(k) alternately, and
-) upsampled low-pass filters (L_p) and upsampled high-pass filters (H_q) combined in a cascade connection,

characterized in that

-) the upsampled low-pass filters (L_p) are formed by upsampling the basic low-pass filter (L_0) by all values of p (p=1,2,...P-1), with p being an upsampling factor in the time domain and P-1 being smaller than M, and
-) the upsampled high-pass filters (H_q) are formed by upsampling the basic high-pass filter (H_0) by all values of q (q=1,2,... Q-1), with q being an upsampling factor in the time domain and Q-1 being smaller than M,

with the maximum upsampling factor M+1 being determined by the ratio of the passband width $\Delta fb(0)$ of the basic filter to the desired passband width $\Delta fb(M)$.

2. A filter according to claim 1, **characterized in that** the upsampling factor P is the minimum value of p that satisfies the following equation:

$$f_3^{(p)} = f_3^{(0)}/(1+p) < f_3$$

with $f_3^{(p)}$ being the cut-off frequency of the upsampled low-pass filter (L_P) defined by a passband attenuation degree of -3dB, $f_3^{(0)}$ being the cut-off frequency of the basic low-pass filter (L_0) defined by a passband attenuation degree of -3dB and f_3 being the desired cut-off frequency (f_1) of the cascade connection stage defined by a desired passband attenuation degree (A) of -3dB.

3. A method for configuring an FIR filter to provide desired passband characteristics, in particular a desired passband width $\Delta fb(M)$, the filter comprising a cascade connection of

-) a basic low-pass filter (L_0) with a passband width $\Delta fb(0)$ and having an impulse response function (ψ) configured by a piecewise polynomial with a finite support and using a value of the impulse response function (ψ) at a connection point between the polynomials as filter coefficients c(k),
-) a basic high-pass filter (H_0) with a passband width $\Delta fb(0)$ and having filter coefficients b(k) obtained by inverting the sign of the filter coefficients c(k) alternately, and
-) upsampled low-pass filters (L_p) and upsampled high-pass filters (H_q), the method comprising the step of inputting a desired center frequency (f_c), a desired cut-off frequency (f_1), a desired stopband frequency (f_2), a desired passband attenuation degree (A) at the cut-off frequency (f_1), a desired stopband attenuation degree (Ad) at the stopband frequency (f_2), and a sampling frequency (f_s),

characterized in that the method further comprises the steps of

-) calculating the desired passband width $\Delta fb(M)$ as (f_1-f_C)*2 and obtaining the maximum upsampling factor M+1 by calculating the ratio of the passband width $\Delta fb(0)$ of the basic filter to the desired passband width $\Delta fb(M)$,
-) sequentially selecting a number of α_p (α_p=0,1,2,...) of the upsampled low-pass filters (L_p) formed by upsampling the basic low-pass filter (L_0) by all values of p (p=1,2,...P-1), with p being an upsampling factor in the time domain and P-1 being smaller than M,
-) and sequentially selecting a number of β_q (β_q =0,1,2,...) of the upsampled high-pass filters (H_q) formed by upsampling the basic high-pass filter (H_0) by all values of q (q=1,2,... Q-1), with q being an upsampling factor in the time domain and Q-1 being smaller than M, and
-) determining whether a cascade connection of the selected upsampled low-pass filters (L_p) and upsampled high-pass filters (H_q) conforms to the desired passband attenuation degree (A) at the cut-off frequency (f_1) and to the desired stopband attenuation degree (Ad) at the stopband frequency (f_2).

4. A method according to claim 3, **characterized in that** the selection of upsampled low-pass filters (L_p) and upsampled high-pass filters (H_q) is made depending on whether the attenuation degree at the stopband frequency (f_2) is lower than or equal to the desired stopband attenuation degree (Ad).

5. A method according to claim 3 or 4, **characterized in that** the selection of upsampled low-pass filters (L_p) and upsampled high-pass filters (H_q) is made depending on whether the attenuation degree at the cut-off frequency (f_1)

is higher than or equal to a predetermined permissible level below the desired passband attenuation degree (A).

6. A method according to one of the claims 3 to 5, **characterized in that** an optimum combination of p, q, α_p, and β_q is selected by selecting α_p and β_q in descending order from stored maximum values and selecting p and q in descending order from the maximum upsampling factor M.

7. A method according to one of the claims 3 to 6, **characterized in that** α_p and β_q are increased, if this causes a filter gain at the desired stopband frequency (f_2) of the filter to decrease.

Patentansprüche

1. Filter mit Soll-Bandpasskennlinien, insbesondere einer Soll-Grenzfrequenz (f_1), einem Soll-Grunddämpfungsgrad (A) und einer Soll-Frequenzbreite $\Delta fb(M)$ des Bandpasses, umfassend

-) ein Basistiefpassfilter (L_0) mit einer Bandpassbreite $\Delta fb(0)$, das eine Impulsantwortfunktion (ψ) aufweist, die durch stückweise Polynome mit endlichen Stützstellen konfiguriert ist, und Werte der Impulsantwortfunktion (ψ) an Verbindungspunkten zwischen den Polynomen als Filterkoeffizienten c(k) verwendet,
-) ein Basishochpassfilter (H_0) mit einer Bandpassbreite $\Delta fb(0)$ und mit Filterkoeffizienten b(k), die durch abwechselndes Umkehren des Vorzeichens der Filterkoeffizienten c(k) erlangt werden, und
-) Tiefpassfilter (L_p) mit erhöhter Abtastrate und Hochpassfilter (H_q) mit erhöhter Abtastrate, die in einer Kaskadenschaltung kombiniert sind,

dadurch gekennzeichnet, dass

-) die Tiefpassfilter (L_p) mit erhöhter Abtastrate durch Abtastratenerhöhung des Basistiefpassfilters (L_0) durch alle Werte von p (p = 1, 2, ..., P-1) gebildet werden, wobei p ein Abtastratenerhöhungsfaktor im Zeitbereich ist und P-1 kleiner als M ist, und
-) die Hochpassfilter (H_q) mit erhöhter Abtastrate durch Abtastratenerhöhung des Basishochpassfilters (H_0) durch alle Werte von q (q = 1, 2, ..., Q-1) gebildet werden, wobei q ein Abtastratenerhöhungsfaktor im Zeitbereich ist und Q-1 kleiner als M ist,

wobei der maximale Abtastratenerhöhungsfaktor M+1 durch das Verhältnis der Bandpassbreite $\Delta fb(0)$ des Basisfilters zur Soll-Bandpassbreite $\Delta fb(M)$ bestimmt wird.

2. Filter nach Anspruch 1, **dadurch gekennzeichnet, dass** der Abtastratenerhöhungsfaktor P der Mindestwert von p ist, der die folgende Gleichung erfüllt:

$$f_3{}^{(p)} = f_3{}^{(0)}/(1+p) < f_3$$

wobei $f_3{}^{(p)}$ die Grenzfrequenz des Tiefpassfilters (L_p) mit erhöhter Abtastrate ist, die durch einen Grunddämpfungsgrad von -3 dB definiert ist, $f_3{}^{(0)}$ die Grenzfrequenz des Basistiefpassfilters (L_0) ist, die durch einen Grunddämpfungsgrad von -3 dB definiert ist, und f_3 die Soll-Grenzfrequenz (f_1) der Kaskadenschaltstufe ist, die durch einen Soll-Grunddämpfungsgrad (A) von -3 dB definiert ist.

3. Verfahren zum Konfigurieren eines FIR-Filters, um Soll-Bandpasskennlinien, insbesondere eine Soll-Bandpassbreite $\Delta fb(M)$, bereitzustellen, wobei das Filter eine Kaskadenschaltung aus Folgendem umfasst:

-) einem Basistiefpassfilter (L_0) mit einer Bandpassbreite $\Delta fb(0)$, das eine Impulsantwortfunktion (ψ) aufweist, die durch stückweise Polynome mit endlichen Stützstellen konfiguriert ist, und einen Wert der Impulsantwortfunktion (ψ) an einem Verbindungspunkt zwischen den Polynomen als Filterkoeffizienten c(k) verwendet,
-) einem Basishochpassfilter (H_0) mit einer Bandpassbreite $\Delta fb(0)$ und mit Filterkoeffizienten b(k), die durch abwechselndes Umkehren des Vorzeichens der Filterkoeffizienten c(k) erlangt werden, und
-) Tiefpassfilter (L_p) mit erhöhter Abtastrate und Hochpassfilter (H_q) mit erhöhter Abtastrate, wobei das Verfahren den Schritt zum Eingeben einer Soll-Mittenfrequenz (f_c), einer Soll-Grenzfrequenz (f_1), einer Soll-Sperrfrequenz (f_2), eines Soll-Grunddämpfungsgrads (A) bei der Grenzfrequenz (f_1), eines Soll-Sperrdämpfungsgrads (Ad)

bei der Sperrfrequenz (f_2) und einer Abtastfrequenz (f_s) umfasst,

dadurch gekennzeichnet, dass das Verfahren ferner die folgenden Schritte umfasst:

-) Berechnen der Soll-Bandpassbreite $\Delta fb(M)$ als $(f_1 - f_C) * 2$ und Erlangen des maximalen Abtastratenerhöhungs- faktors M+1 durch Berechnen des Verhältnisses der Bandpassbreite $\Delta fb(0)$ des Basisfilters zur Soll-Bandpass- breite $\Delta fb(M)$,
-) nacheinander Auswählen einer Anzahl α_p (α_p = 0, 1, 2, ...) der Tiefpassfilter (L_p) mit erhöhter Abtastrate, die durch Abtastratenerhöhung des Basistiefpassfilters (L_0) durch alle Werte von p (p = 1, 2, ..., P-1) gebildet werden, wobei p ein Abtastratenerhöhungsfaktor im Zeitbereich ist und P-1 kleiner als M ist,
-) und nacheinander Auswählen einer Anzahl β_q (β_q = 0, 1, 2, ...) der Hochpassfilter (H_q) mit erhöhter Abtastrate, die durch Abtastratenerhöhung des Basishochpassfilters (H_0) durch alle Werte von q (q = 1, 2, ..., Q-1) gebildet werden, wobei q ein Abtastratenerhöhungsfaktor im Zeitbereich ist und Q-1 kleiner als M ist, und
-) Bestimmen, ob eine Kaskadenschaltung der ausgewählten Tiefpassfilter (L_p) mit erhöhter Abtastrate und Hochpassfilter (H_q) mit erhöhter Abtastrate dem Soll-Grunddämpfungsgrad (A) bei der Grenzfrequenz (f_1) und dem Soll-Sperrdämpfungsgrad (Ad) bei der Sperrfrequenz (f_2) entspricht.

4. Verfahren nach Anspruch 3, **dadurch gekennzeichnet, dass** die Auswahl an Tiefpassfiltern (L_p) mit erhöhter Abtastrate und Hochpassfiltern (H_q) mit erhöhter Abtastrate in Abhängigkeit davon erfolgt, ob der Dämpfungsgrad bei der Sperrfrequenz (f_2) kleiner gleich dem Soll-Spendämpfungsgrad (Ad) ist.

5. Verfahren nach Anspruch 3 oder 4, **dadurch gekennzeichnet, dass** die Auswahl an Tiefpassfiltern (L_p) mit erhöhter Abtastrate und Hochpassfiltern (H_q) mit erhöhter Abtastrate in Abhängigkeit davon erfolgt, ob der Dämpfungsgrad bei der Grenzfrequenz (f_1) größer gleich einem vorgegebenen zulässigen Niveau unter dem Soll-Grunddämpfungs- grad (A) ist.

6. Verfahren nach einem der Ansprüche 3 bis 5, **dadurch gekennzeichnet, dass** eine optimale Kombination aus p, q, α_p und β_q durch Auswählen von α_p und β_q in absteigender Reihenfolge von gespeicherten Höchstwerten und Auswählen von p und q in absteigender Reihenfolge von dem maximalen Abtastratenerhöhungsfaktor M ausgewählt wird.

7. Verfahren nach einem der Ansprüche 3 bis 6, **dadurch gekennzeichnet, dass** α_p und β_q erhöht werden, wenn dies dazu führt, dass eine Filterverstärkung bei der Soll-Sperrfrequenz (f_2) des Filters abnimmt.

Revendications

1. Filtre ayant des caractéristiques de bande passante souhaitées, en particulier une fréquence de coupure (f_1) sou- haitée, un degré d'atténuation de bande passante (A) souhaité et une largeur de fréquence souhaitée $\Delta fb(M)$ de la bande passante, comprenant :

-) un filtre passe-bas de base (L_0) avec une largeur de bande passante $\Delta fb(0)$ et ayant une fonction de réponse impulsionnelle (ψ) configurée par une polynomiale par morceaux avec un support fini et utilisant des valeurs de fonction de réponse impulsionnelle (ψ) en des points de connexion entre les polynomiales en tant que coefficients de filtre c(k),
-) un filtre passe-haut de base (H_0) avec une largeur de bande passante $\Delta fb(0)$ et ayant des coefficients de filtre b(k) obtenus par inversion du signe des coefficients de filtre c(k) de manière alternée, et
-) des filtres passe-bas suréchantillonnés (L_p) et des filtres passe-haut suréchantillonnés (H_q)

combinés dans une connexion en cascade,
caractérisé en ce que

-) les filtres passe-bas suréchantillonnés (L_p) sont formés par suréchantillonnage du filtre passe-bas de base (L_0) de toutes les valeurs de p (p = 1, 2, ... P-1), p étant un facteur de suréchantillonnage dans le domaine temporel et P-1 étant plus petit que M, et
-) les filtres passe-haut suréchantillonnés (H_q) sont formés par suréchantillonnage du filtre passe-haut de base (H_0) de toutes les valeurs de q (q = 1, 2, ... Q-1), q étant un facteur de suréchantillonnage dans le domaine temporel et Q-1 étant plus petit que M,

le facteur de suréchantillonnage maximal M+1 étant déterminé par le rapport entre la largeur de bande passante Δfb(0) du filtre de base et la largeur de bande passante souhaitée Δfb(M).

2. Filtre selon la revendication 1, **caractérisé en ce que** le facteur de suréchantillonnage P est la valeur minimale de p qui satisfait l'équation suivante :

$$f_3{}^{(p)} = f_3{}^{(0)}/(1+p) < f_3$$

$f_3{}^{(p)}$ étant la fréquence de coupure du filtre passe-bas suréchantillonné (L_p) définie par un degré d'atténuation de bande passante de -3dB, $f_3{}^{(0)}$ étant la fréquence de coupure du filtre passe-bas de base (L_0) définie par un degré d'atténuation de bande passante de -3dB et f_3 étant la fréquence de coupure souhaitée (f_1) de l'étage de connexion en cascade définie par un degré d'atténuation de bande passante souhaité (A) de -3dB.

3. Procédé pour configurer un filtre FIR pour fournir des caractéristiques de bande passante souhaitées, en particulier une largeur de bande passante souhaitée Δfb(M), le filtre comprenant une connexion en cascade de

-) un filtre passe-bas de base (L_0) avec une largeur de bande passante Δfb(0) et ayant une fonction de réponse impulsionnelle (ψ) configurée par une polynomiale par morceaux avec un support fini et utilisant une valeur de fonction de réponse impulsionnelle (ψ) en un point de connexion entre les polynomiales en tant que coefficients de filtre c(k),
-) un filtre passe-haut de base (H_0) avec une largeur de bande passante Δfb(0) et ayant des coefficients de filtre b(k) obtenus par inversion du signe des coefficients de filtre c(k) de manière alternée, et
-) des filtres passe-bas suréchantillonnés (L_p) et des filtres passe-haut suréchantillonnés (H_q),

le procédé comprenant l'étape consistant à entrer une fréquence centrale souhaitée (f_c), une fréquence de coupure souhaitée (f_1), une fréquence de bande d'affaiblissement souhaité (f_2), un degré d'atténuation de bande passante souhaité (A) à la fréquence de coupure (f_1), un degré d'atténuation de bande d'affaiblissement souhaité (Ad) à la fréquence de bande d'affaiblissement (f_2), et une fréquence d'échantillonnage (f_s),
caractérisé en ce que le procédé comprend en outre les étapes consistant à

-) calculer la largeur de bande passante souhaitée Δfb(M) comme (f_1-f_C)*2 et obtenir le facteur de suréchantillonnage maximal M+1 par calcul du rapport entre la largeur de bande passante Δfb (0) du filtre de base et la largeur de bande passante souhaitée Δfb (M),
-) sélectionner séquentiellement un nombre α_p (α_p = 0, 1, 2, ...) des filtres passe-bas suréchantillonnés (L_p) formés par suréchantillonnage du filtre passe-bas de base (L_0) de toutes les valeurs de p (p = 1, 2, ... P-1), p étant un facteur de suréchantillonnage dans le domaine temporel et P-1 étant plus petit que M,
-) et sélectionner séquentiellement un nombre β_q (β_q = 0, 1, 2, ...) des filtres passe-haut suréchantillonnés (H_q) formés par suréchantillonnage du filtre passe-haut de base (H_0) de toutes les valeurs de q (q = 1, 2, ... Q-1), q étant un facteur de suréchantillonnage dans le domaine temporel et Q-1 étant plus petit que M, et
-) déterminer si une connexion en cascade des filtres passe-bas suréchantillonnés (L_p) et des filtres passe-haut suréchantillonnés (H_q) sélectionnés est conforme au degré d'atténuation de bande passante souhaité (A) à la fréquence de coupure (f_1) et au degré d'atténuation de bande d'affaiblissement souhaité (Ad) à la fréquence de bande d'affaiblissement (f_2).

4. Procédé selon la revendication 3, **caractérisé en ce que** la sélection de filtres passe-bas suréchantillonnés (L_p) et de filtres passe-haut suréchantillonnés (H_q) est effectuée en fonction du fait que le degré d'atténuation à la fréquence de bande d'affaiblissement (f_2) est inférieur ou égal au degré d'atténuation de bande d'affaiblissement souhaité (Ad).

5. Procédé selon la revendication 3 ou 4, **caractérisé en ce que** la sélection de filtres passe-bas suréchantillonnés (L_p) et de filtres passe-haut suréchantillonnés (H_q) est effectuée en fonction du fait que le degré d'atténuation à la fréquence de coupure (f_1) est supérieur ou égal à un niveau admissible prédéterminé au-dessous du degré d'atténuation de bande passante souhaité (A).

6. Procédé selon l'une des revendications 3 à 5, **caractérisé en ce qu'**une combinaison optimale de p, q, α_p, et β_q est sélectionnée par sélection de α_p et β_q dans l'ordre décroissant à partir de valeurs maximales stockées et sélection de p et q dans l'ordre décroissant à partir du facteur de suréchantillonnage maximal M.

7. Procédé selon l'une des revendications 3 à 6, **caractérisé en ce que** α_p et β_q sont augmentés, si cela entraîne une diminution d'un gain de filtre à la fréquence de bande d'affaiblissement souhaitée (f_2) du filtre.
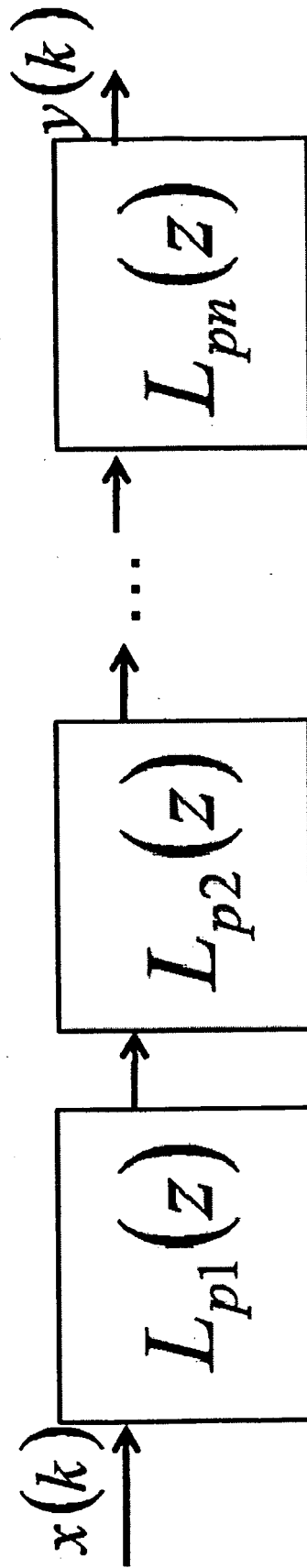
FIG. 1

ARITHMETIC CIRCUIT OF

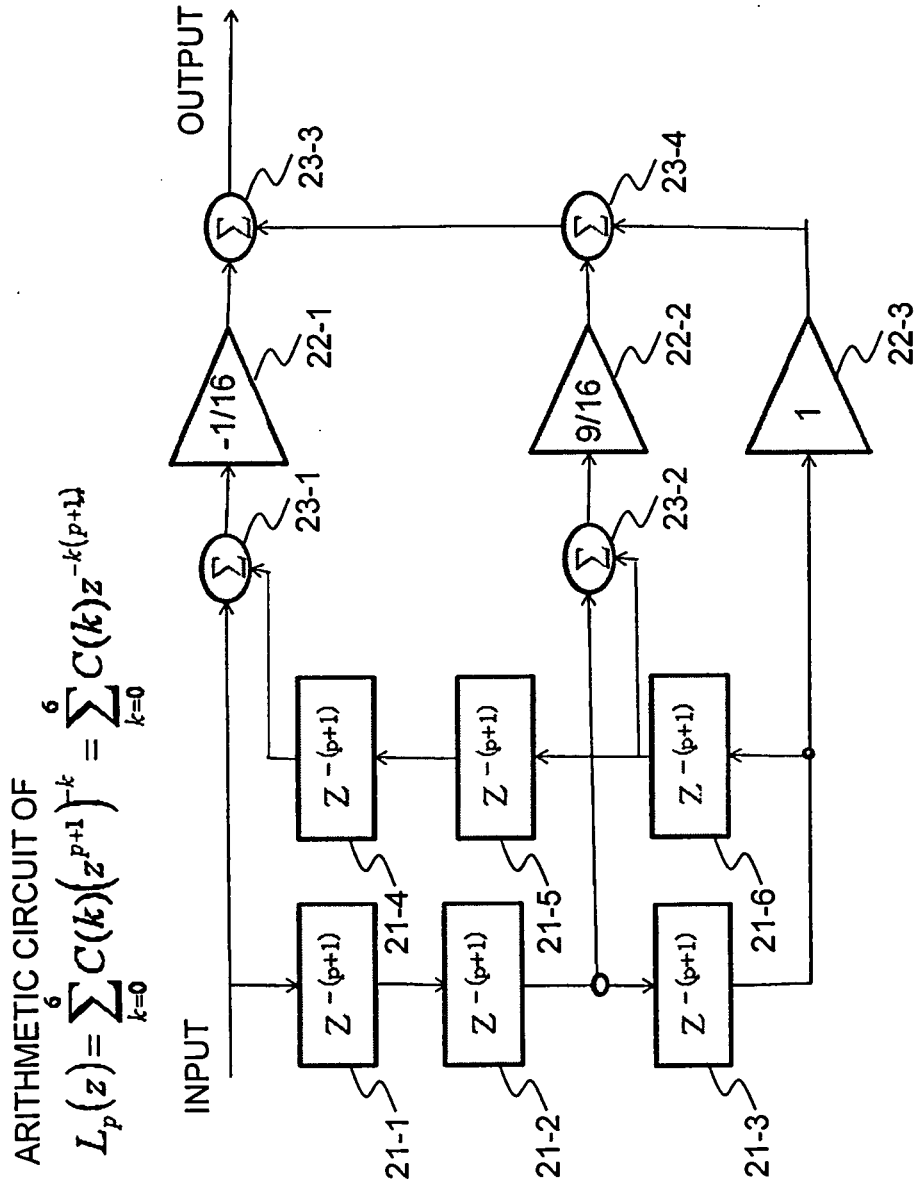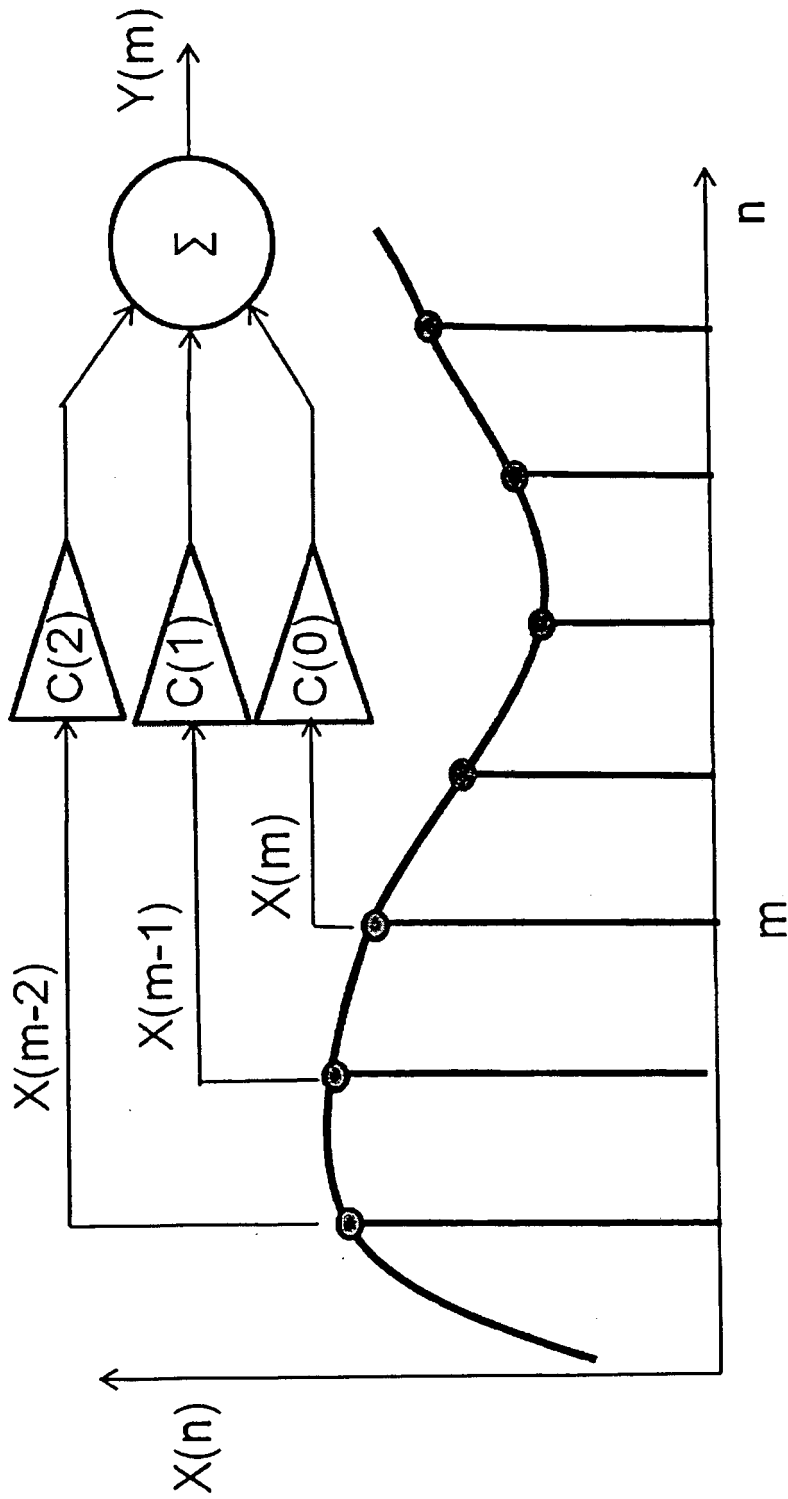$$L_p(z) = \sum_{k=0}^{6} C(k)\left(z^{p+1}\right)^{-k} = \sum_{k=0}^{6} C(k)z^{-k(p+1)}$$

FIG. 2

FIG. 3

$$OUTPUT \quad y(m) = \sum_{k=0}^{p} C_k * x_{m-k}$$
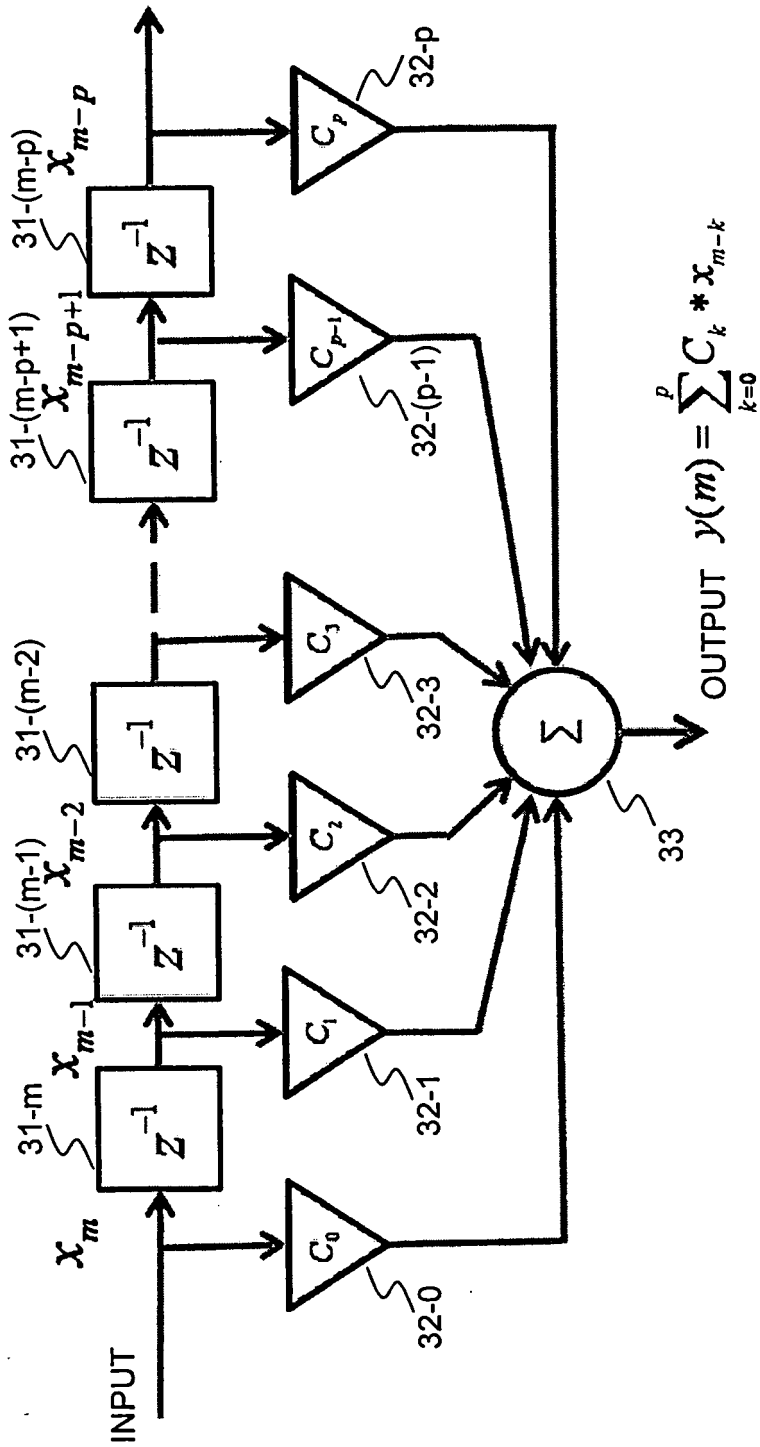
FIG. 4
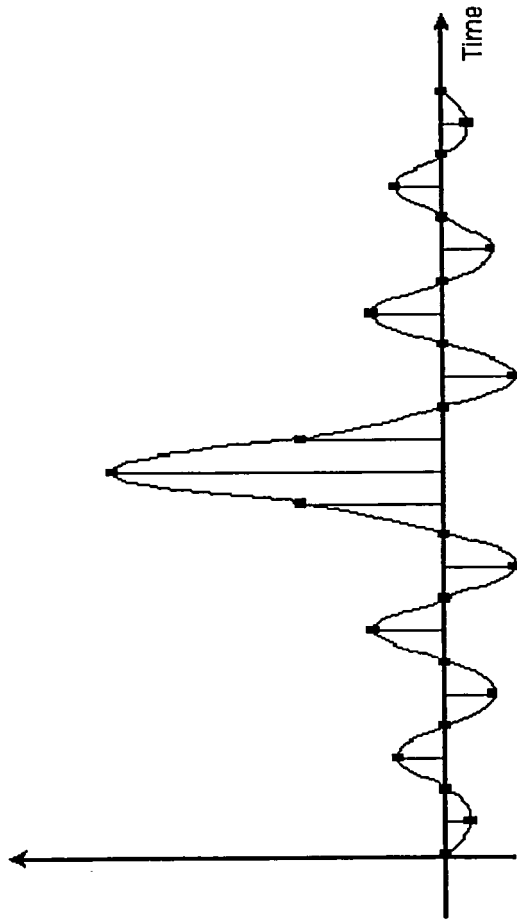
FIG. 5
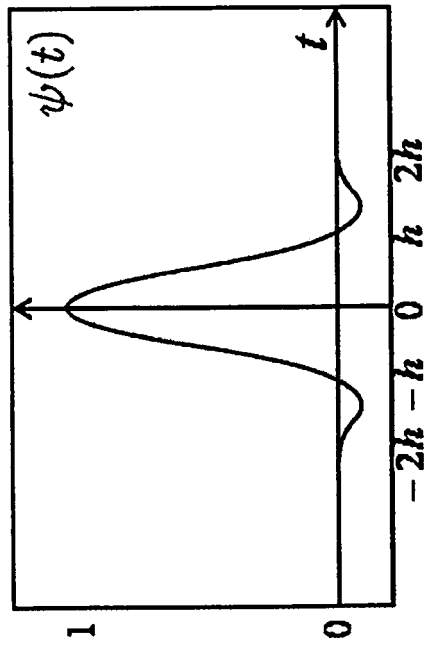
FIG. 6
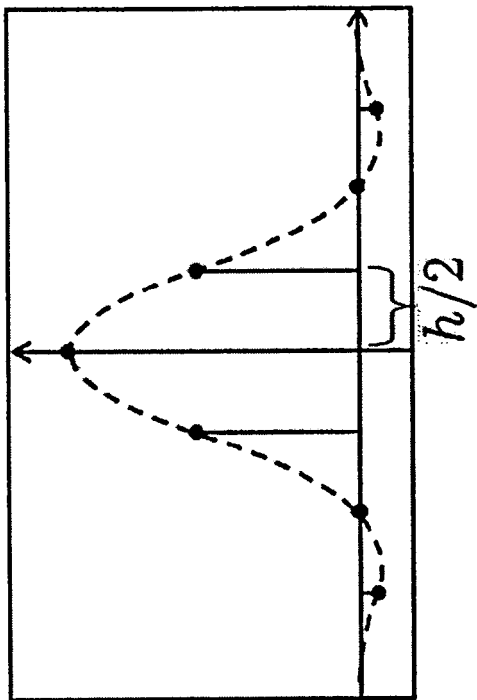
FIG. 7
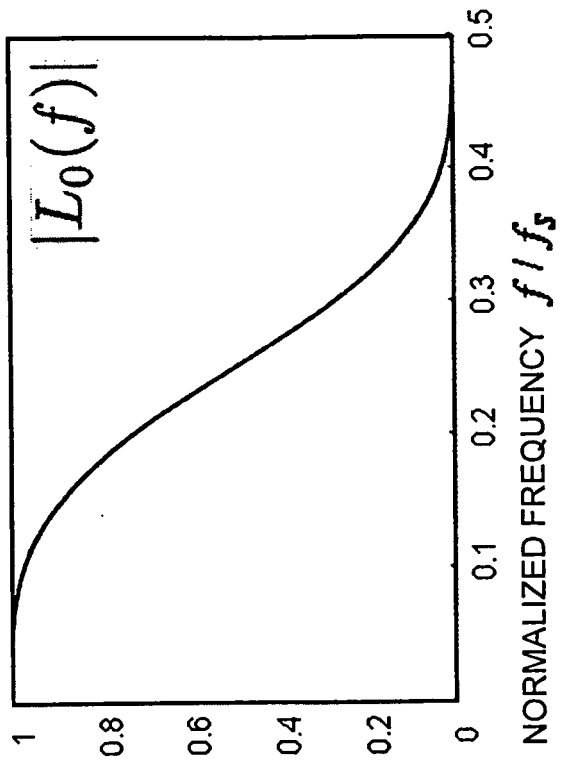
FIG. 8

FIG. 9

FIG. 10

FIG. 11

M ZEROS
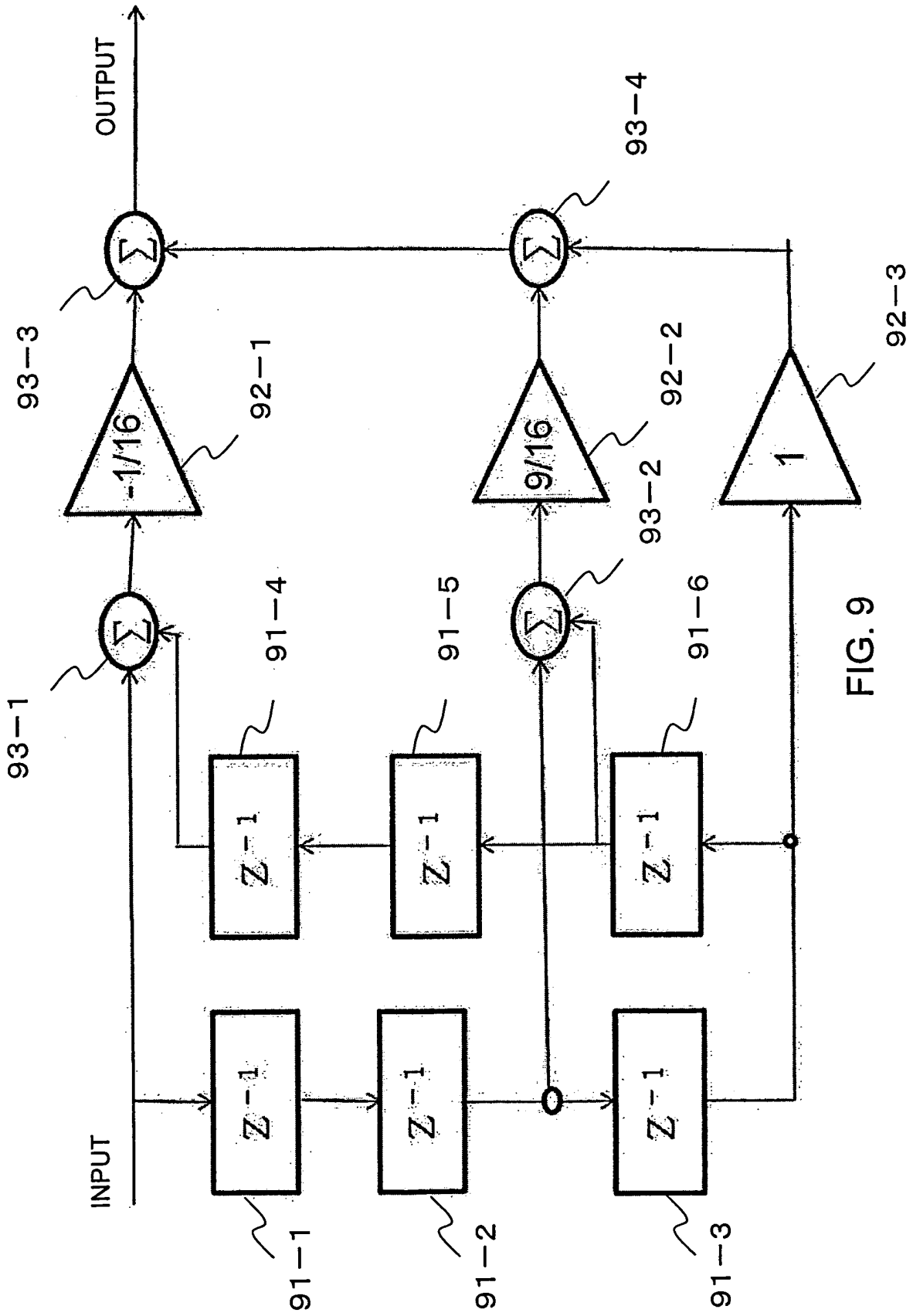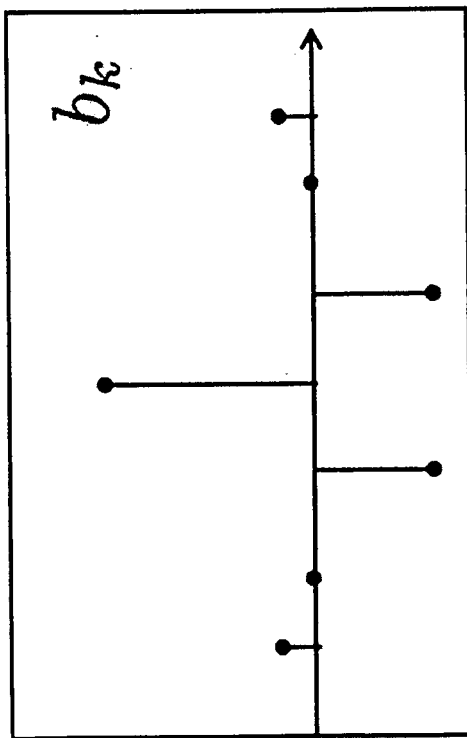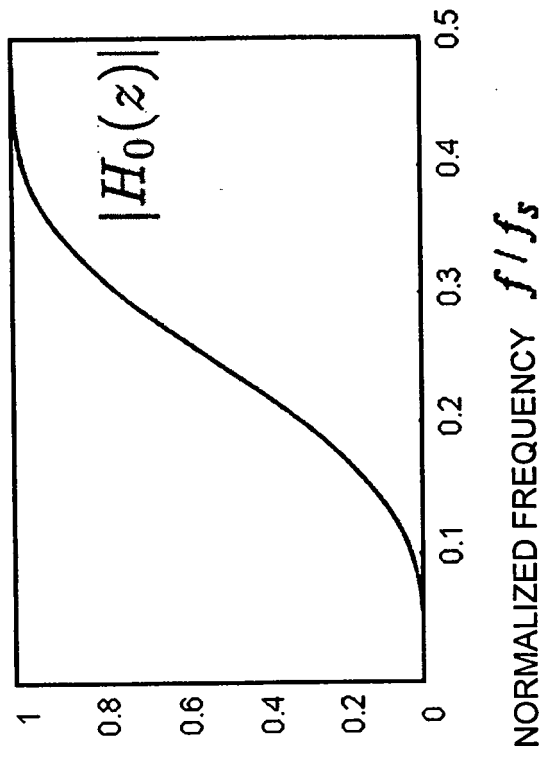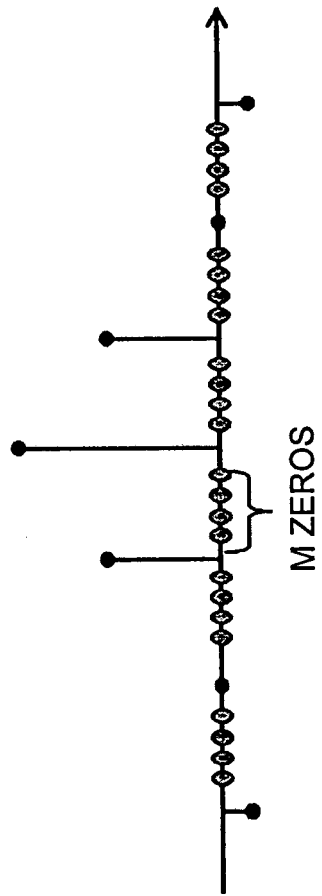
FIG. 12
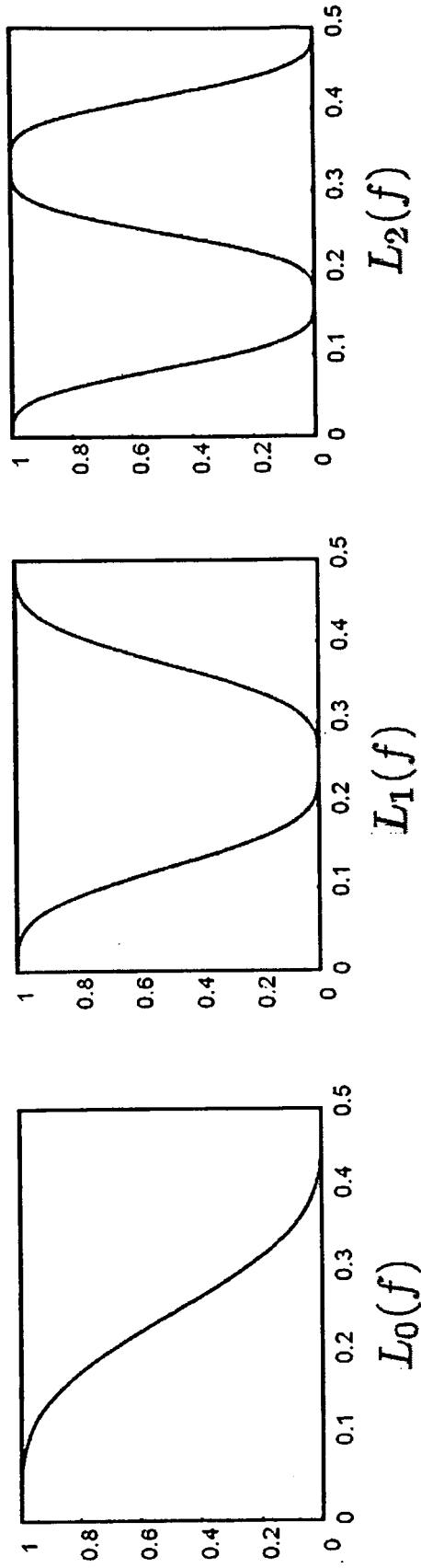
FIG. 13

FIG. 14

FIG. 15

FIG. 16

FIG. 17

START

S01

INPUT DESIRED CHARACTERISTICS DATA
CENTER FREQUENCY fc, CUTOFF FREQUENCY f1, STOPBAND FREQUENCY f2,
STOPBAND ATTENUATION DEGREE Ad, SAMPLING FREQUENCY fs

S03

CALCULATE SHIFT AMOUNT B BY WHICH CENTER FREQUENCY fc IS SHIFTED TO ORIGIN 0
$B = [fc/fs]$ SQUARE BRACKETS INDICATE THAT FRACTIONS ARE ROUNDED OFF
TO WHOLE NUMBERS.

CALCULATE MAXIMUM SCALING DEGREE
PASSBAND WIDTH $\Delta fb(M) = (f1 - fc) * 2$
$M = P = f - 3 (\Delta f)$:SCALING CURVE THAT CAUSES ATTENUATION OF -3 dB
$= [\Delta fb(0) / \Delta fb(M)] - 1$];SQUARE BRACKETS INDICATE THAT FRACTIONS
ARE ROUNDED OFF TO WHOLE NUMBERS.

S05

COMBINE FILTERS

$$B(z) = B_0(z) \prod_{p=0}^{P} L_p^{\alpha_p}(z) \cdot \prod_{q=0}^{Q} H_q^{\beta_q}(z), \quad \alpha_p, \beta_q \in N_0$$

$$B_0(z) = L_M(z) \quad or \quad H_M(z)$$
$$L_p(z) = L_0(z^{p+1}), H_q(z) = H_0(z^{q+1})$$

SELECT OPTIMUM COMBINATION OF p, q, AND α, AND β
UNTIL MAXIMUM ATTENUATION DEGREE Xd OF B(z) AT FREQUENCY LARGER THAN OR
EQUAL TO STOPBAND FREQUENCY f2 BECOMES Xd < Ad.
(MAXIMUM ATTENUATION IS JUDGED BY Xd = X(z) AT f = fs/(p + 1);
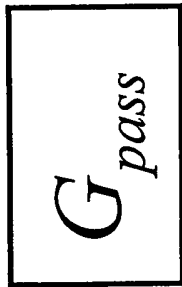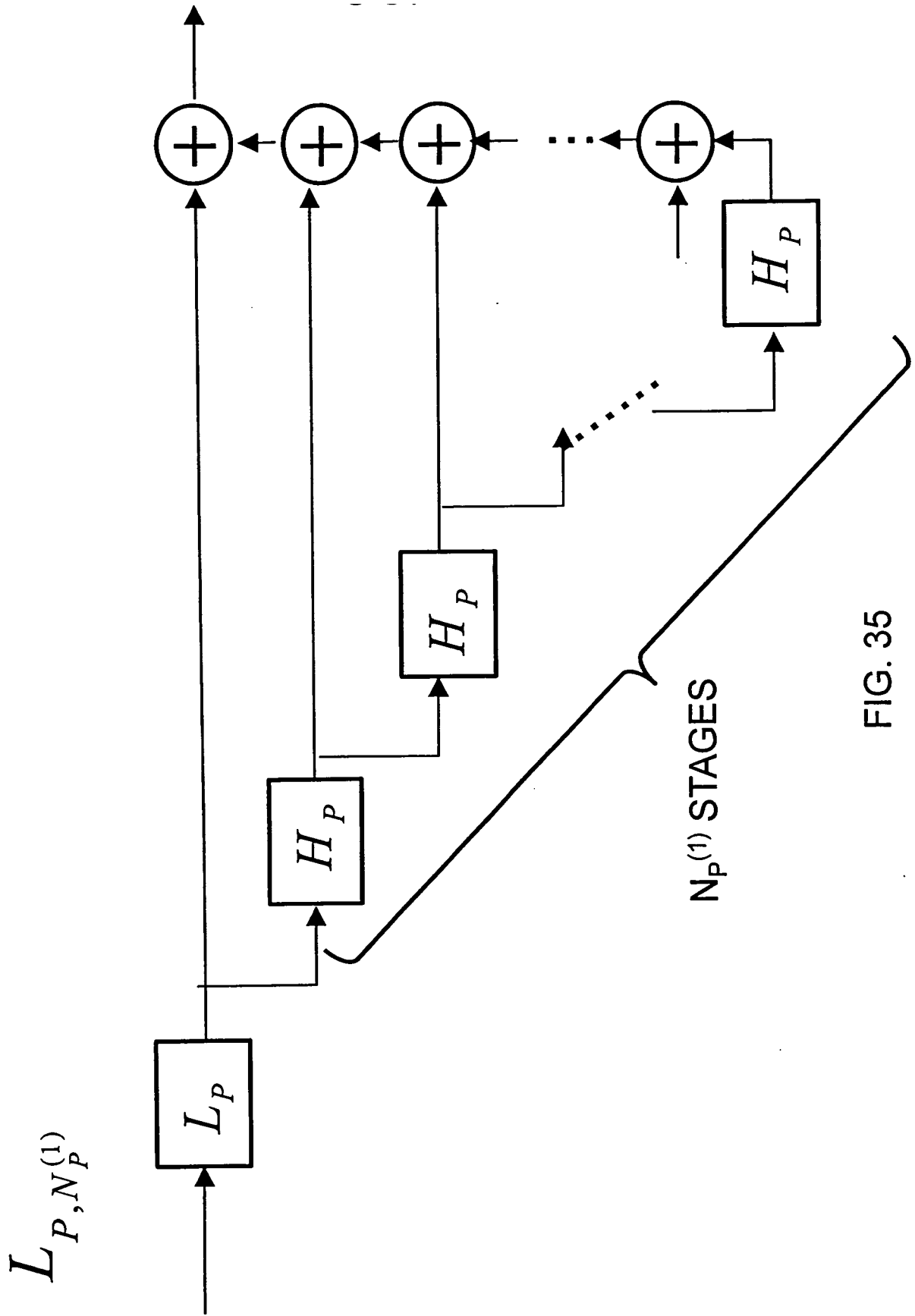WHEN Xd DECREASES AT f = f2, α AND β ARE INCREASED;
WHEN Xd DECREASES AT f = f1, ADDED Lp AND Hp ARE DELETED.)

S07

OBTAIN FILTER BB(z) EXPRESSED AS FOLLOWS

$$B_B(z) = B_0(z) \prod_{p=0}^{P} L_{pB}^{\alpha_p}(z) \cdot \prod_{q=0}^{Q} H_{qB}^{\beta_q}(z) = \sum_{k=1}^{N} h_B(k) z^{-k}$$

$$h_B(k) = 2 h(k) \cos(\omega_c k)$$

$$B(z) = \sum_{k=1}^{N} h(k) z^{-k}$$

END

FIG. 18

FIG. 19

FIG. 20

FIG. 21

FIG. 22

FIG. 23

$$H_{0B}(z)$$

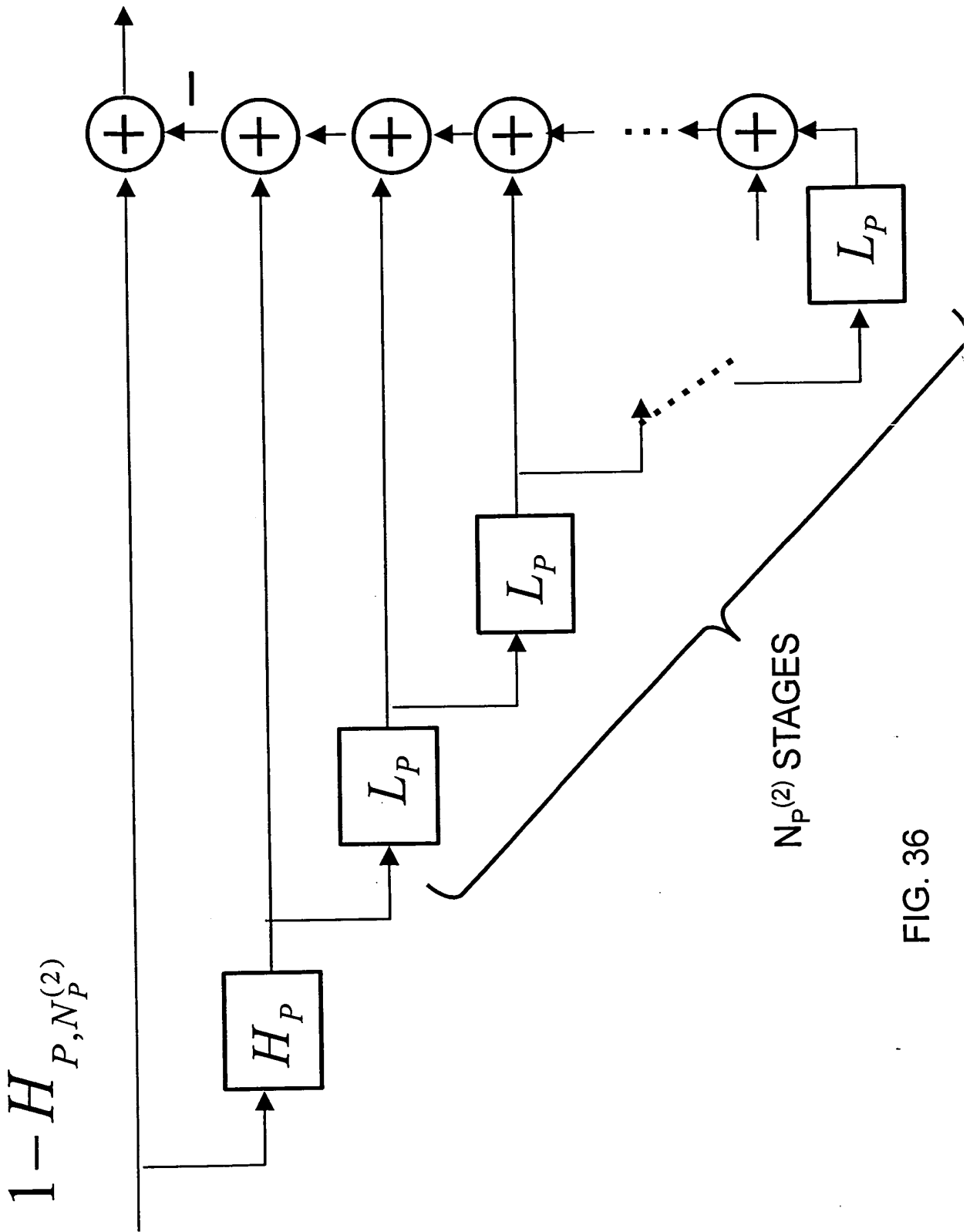FIG. 24

$H_{qB}(z)$ EACH BLOCK HAS $(q+1)$ DELAY ELEMENTS

FIG. 25

H_{qB}

FIG. 26

FIG. 27
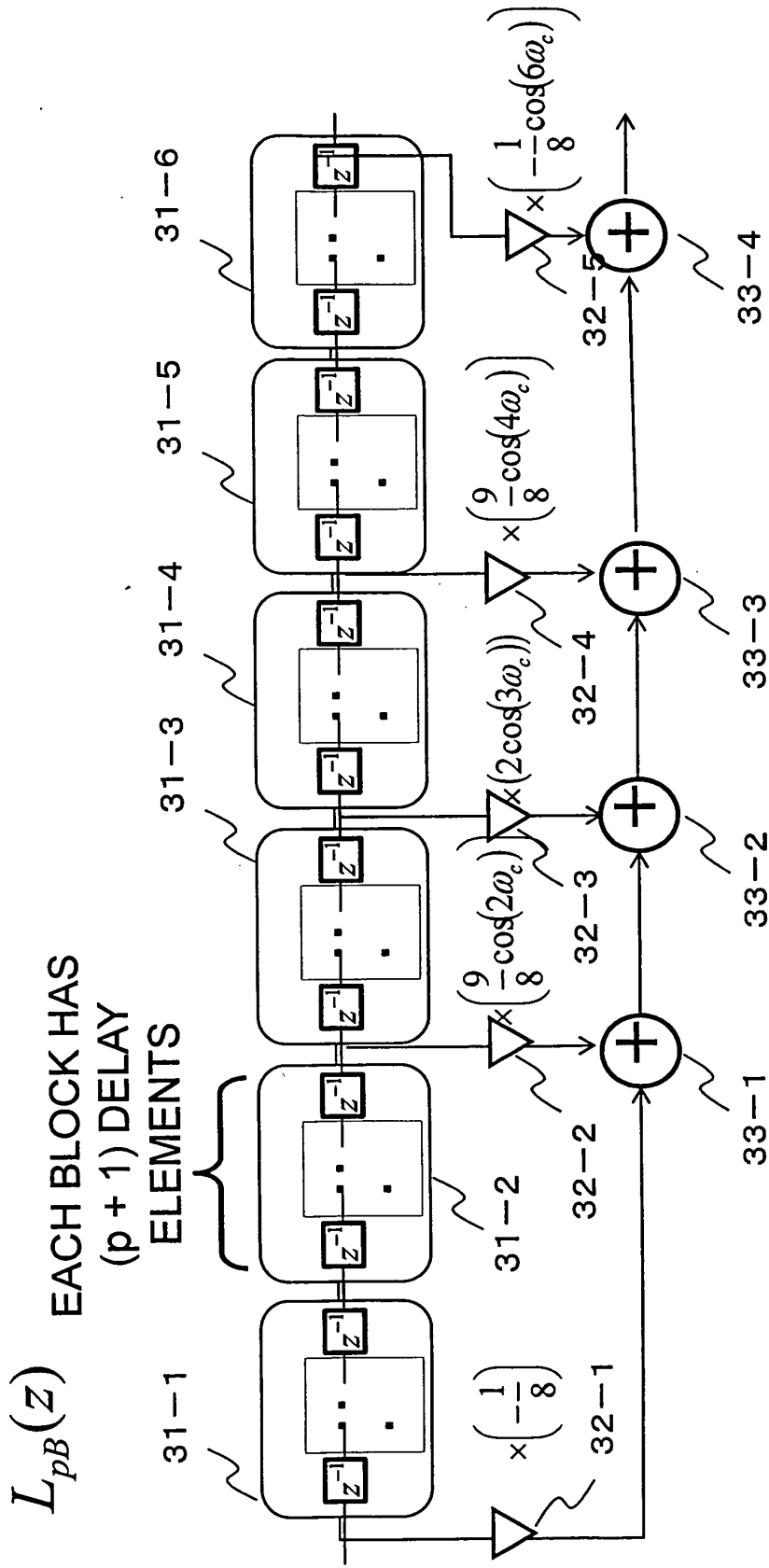
FIG. 28
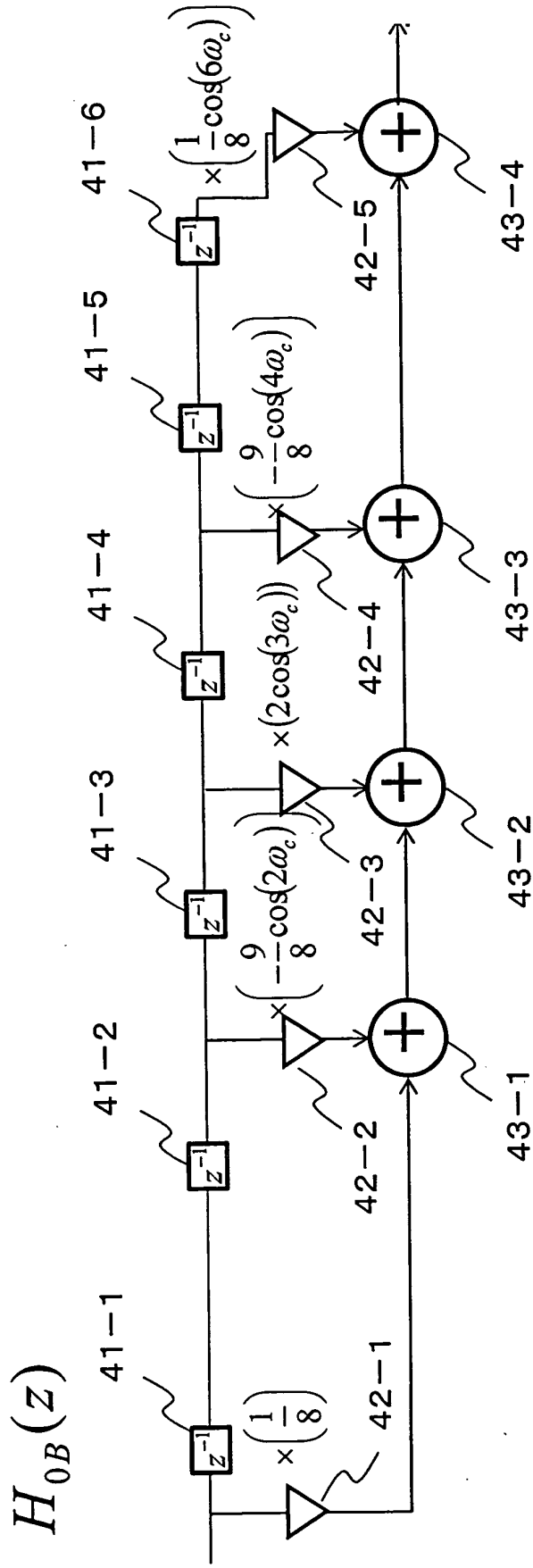
FIG. 29
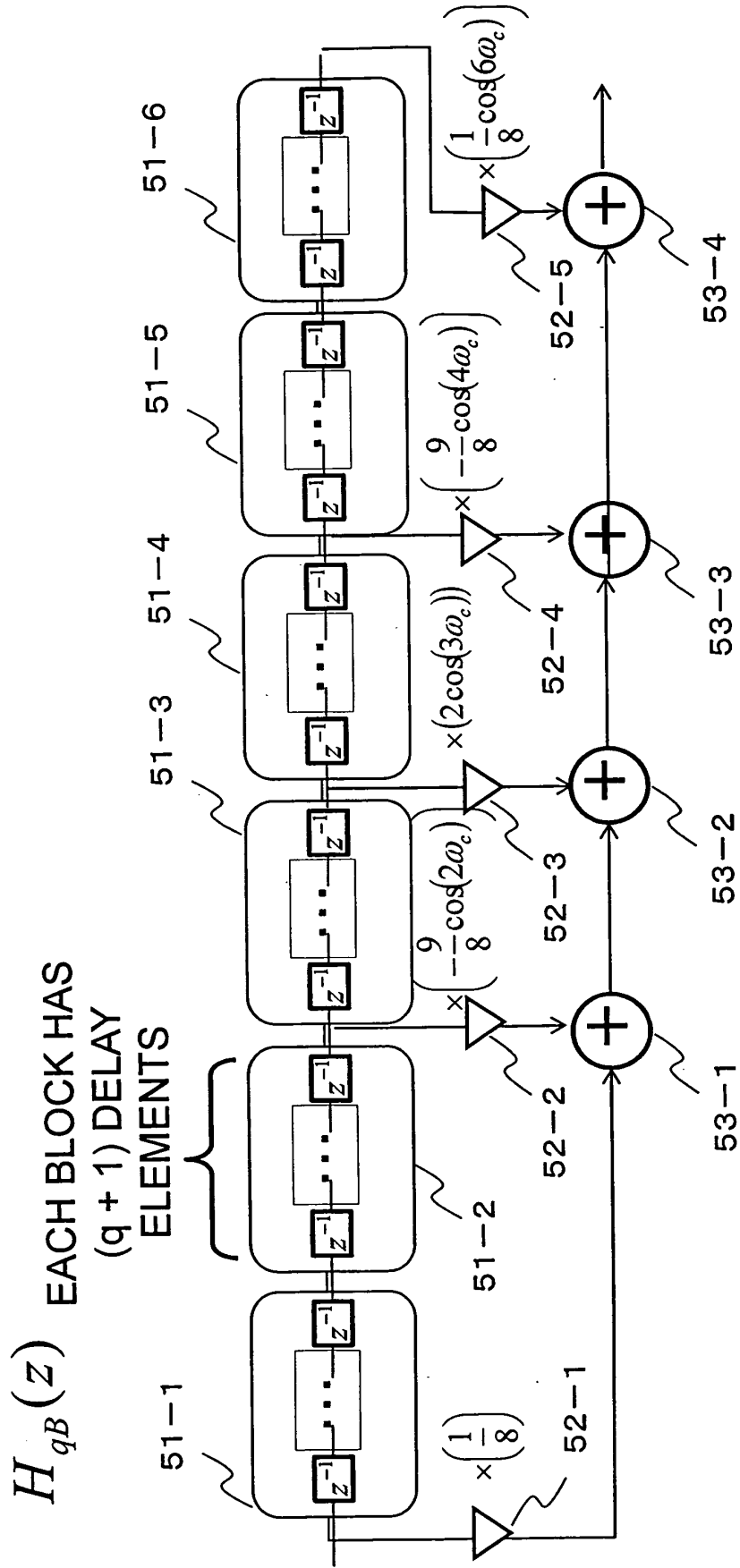
PRESENT INVENTION
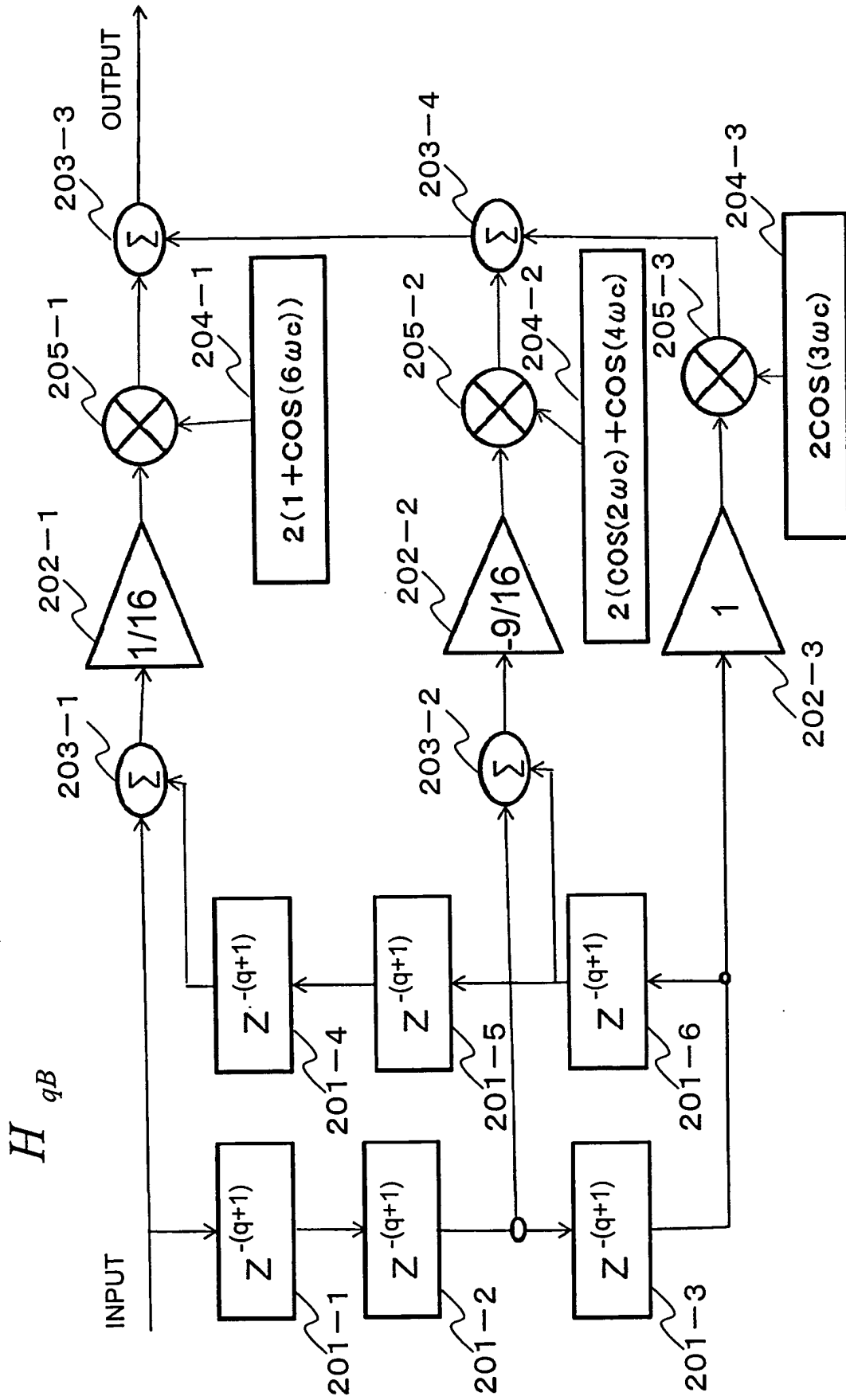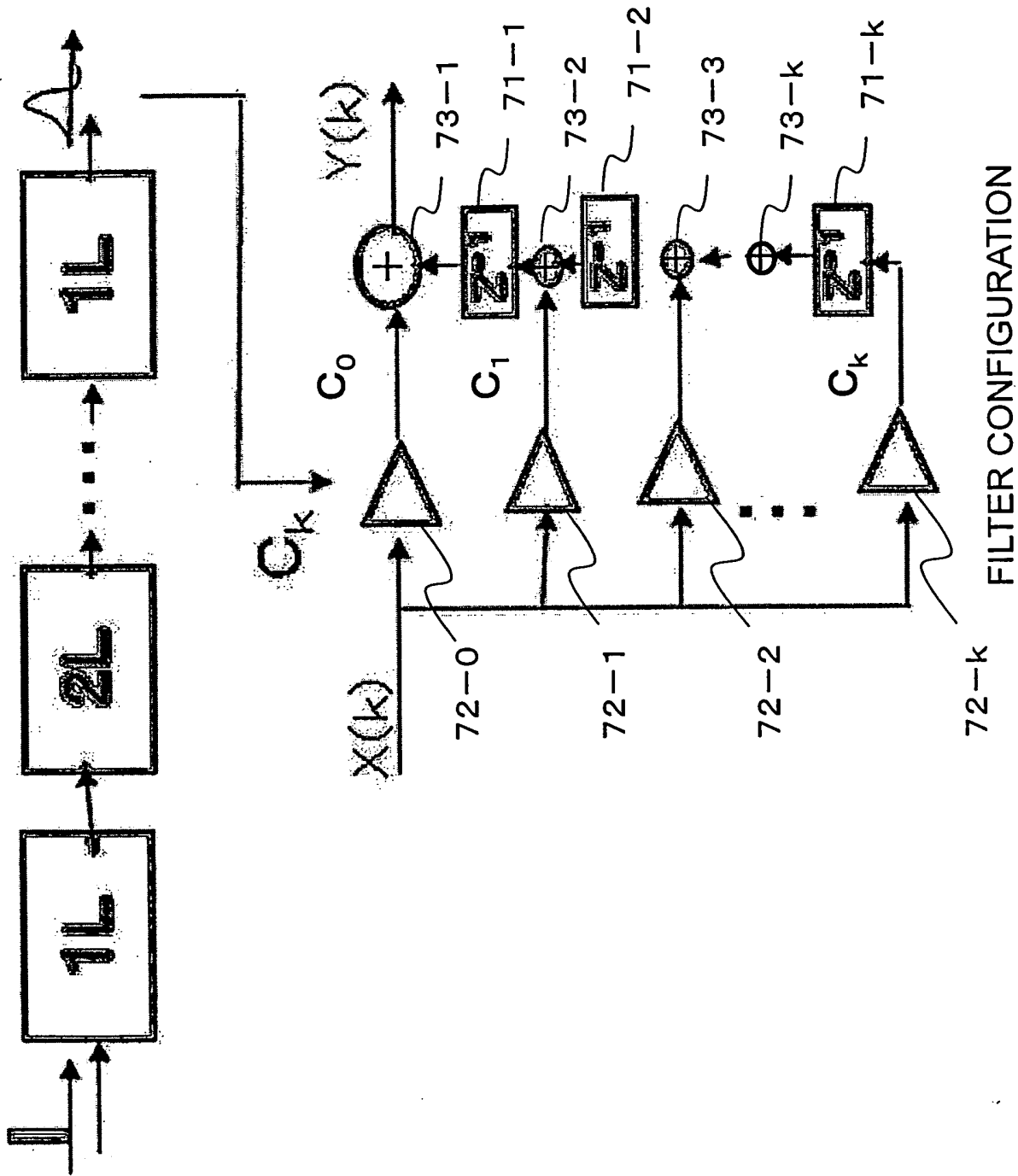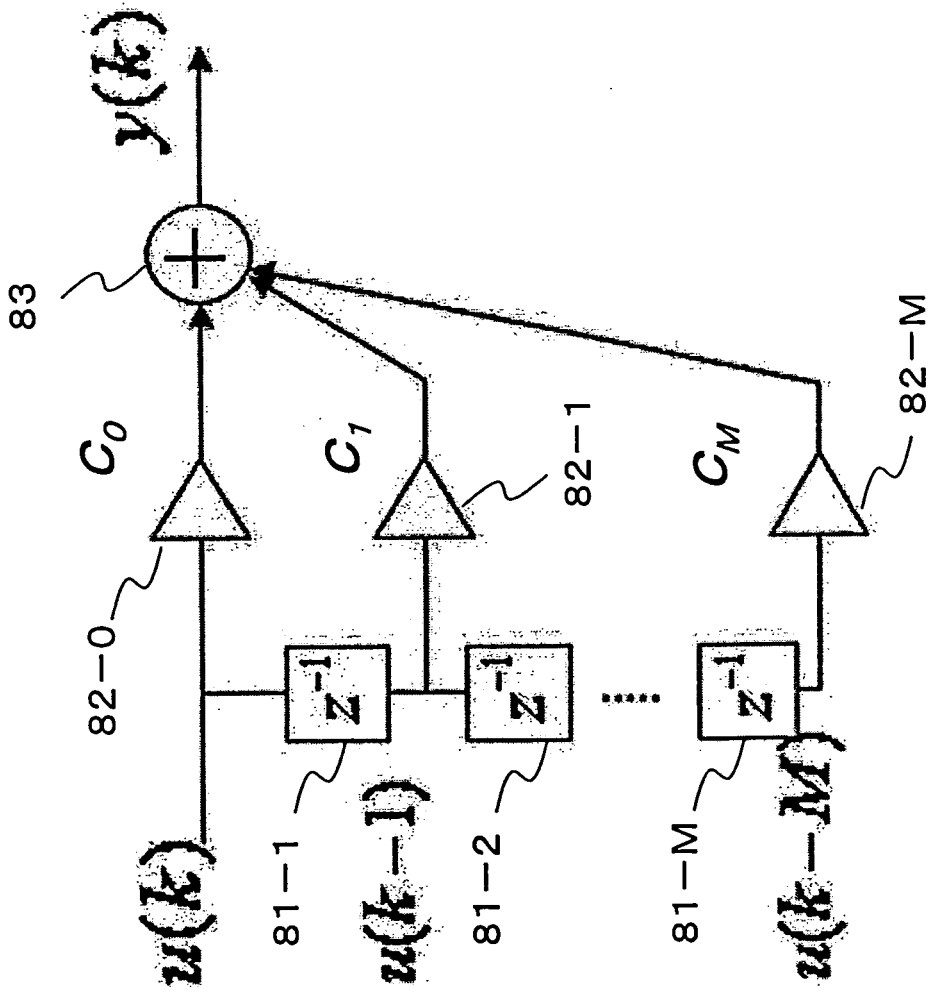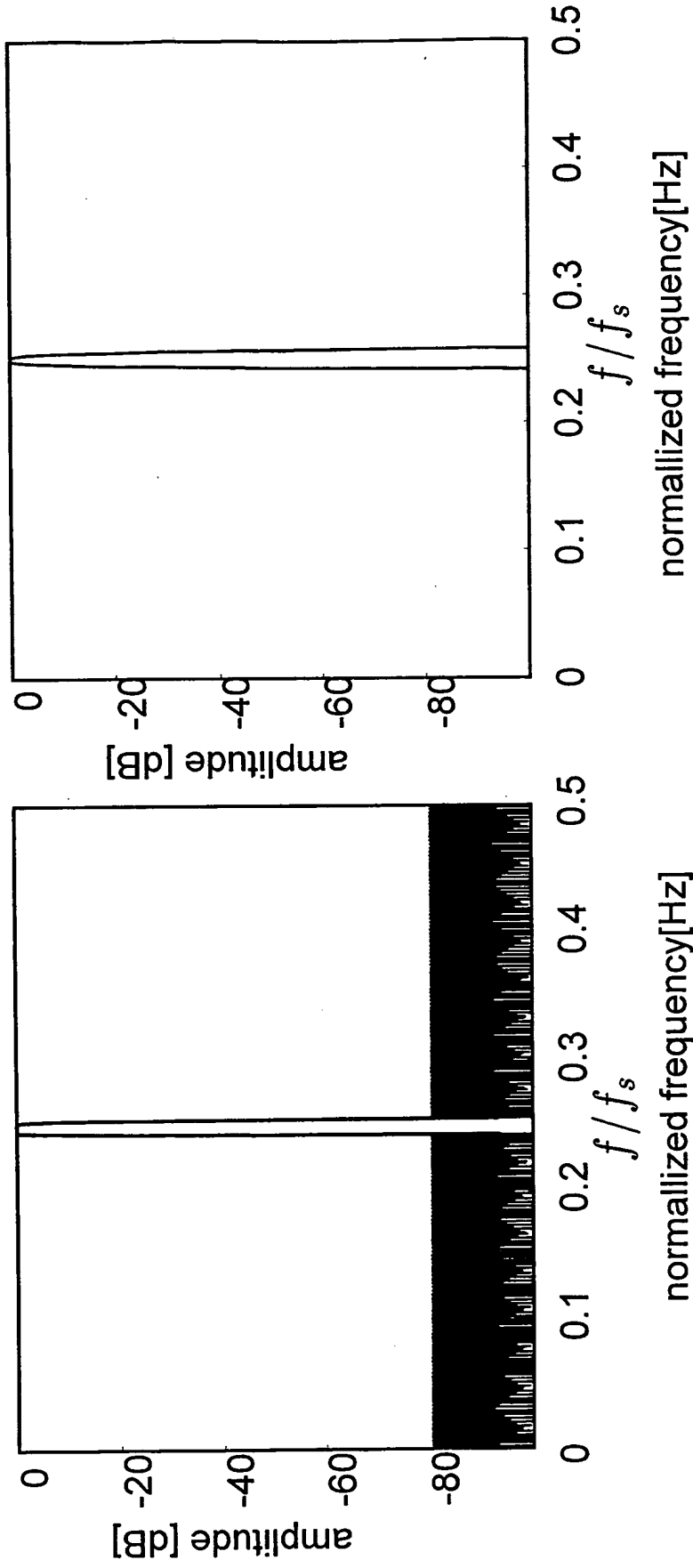
Remez METHOD

FIG. 30

	Remez METHOD	PRESENT INVENTION
PASSBAND WIDTH	7.29kHz	5.58kHz
NUMBER OF TAPS	DIRECT CONFIGURATION 1707 (LOGICAL VALUE) 829 (AFTER QUANTIZATION)	95 (CASCADE CONFIGURATION)
STOPBAND ERROR	LOGICAL VALUE: -80 dB OR LOWER AFTER QUANTIZATION: HIGHER THAN -80 dB ACROSS THE WHOLE RANGE	LOGICAL VALUE: -100 dB OR LOWER AFTER QUANTIZATION: -80 dB OR LOWER ALMOST ACROSS THE WHOLE RANGE

(18 QUANTIZATION BITS)

FIG. 31

FIG. 32

$$G = G_{pass} \; \boxed{G} \; G_{stop}$$

FIG. 33

$$G_{pass} = L_{P,N_P^{(1)}}^{\alpha_P} \left[1 - H_{P,N_P^{(2)}} \right]^{\beta_P}$$

$$\boxed{G_{pass}}$$

$$\boxed{L_{P,N_P^{(1)}}} \cdots \cdots \boxed{L_{P,N_P^{(1)}}} \quad \boxed{1-H_{P,N_P^{(2)}}} \cdots \cdots \boxed{1-H_{P,N_P^{(2)}}}$$

CASCADE CONNECTION
OF α_P STAGES

CASCADE CONNECTION
OF β_P STAGES

FIG. 34

$$L_{P,N_P^{(1)}}$$

FIG. 35

$N_P^{(1)}$ STAGES

$$1 - H_{P,N_P^{(2)}}$$

$N_P^{(2)}$ STAGES

FIG. 36

$$G_{stop} = \left[\prod_{k=1}^{K} L_{p_k,N_k}^{\alpha_k} \right] \cdot \prod_{l=1}^{L} \left[1 - H_{q_l,N_l} \right]^{\beta_l}$$

CASCADE CONNECTION OF K STAGES

CASCADE CONNECTION OF L STAGES

FIG. 37

FIG. 38

START

0. INPUT DESIGN SPECIFICATIONS

1. DETERMINE MAXIMUM SCALING VALUE p = P

2. CONFIGURE PASSBAND Gpass

3. CONFIGURE STOPBAND Gstop

4. G=Gpass·Gstop

END
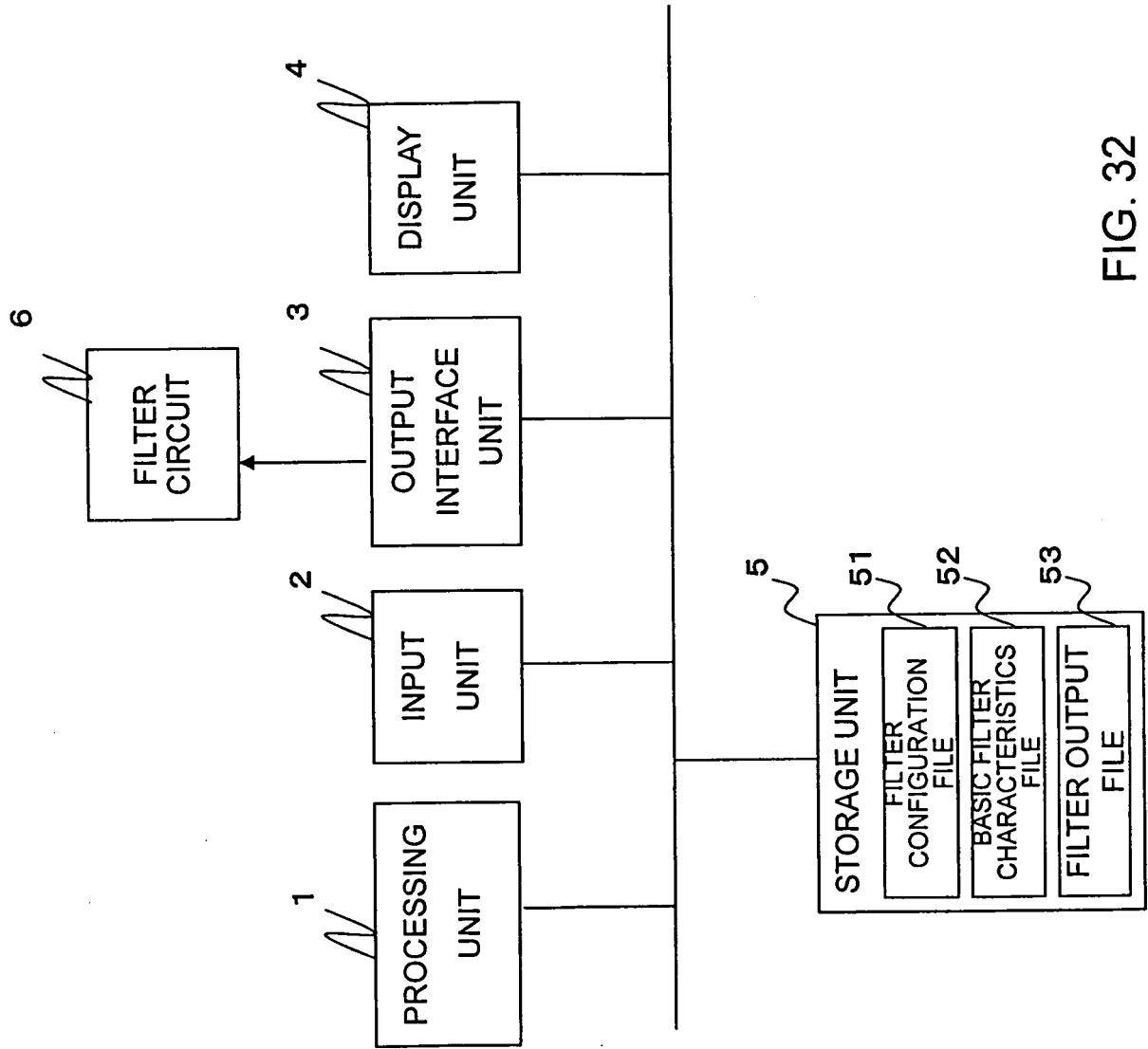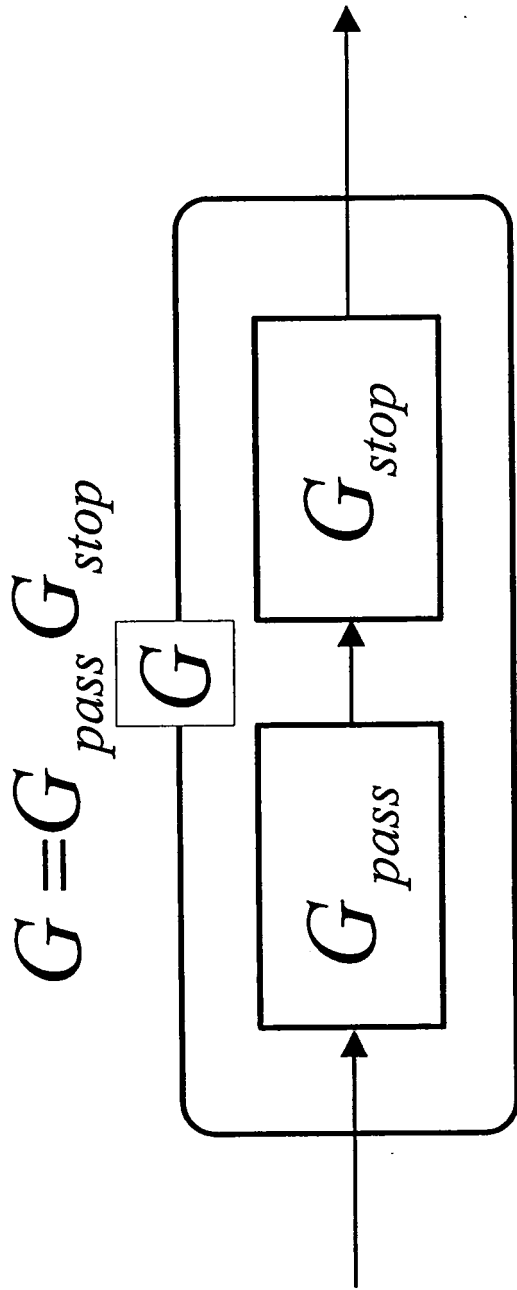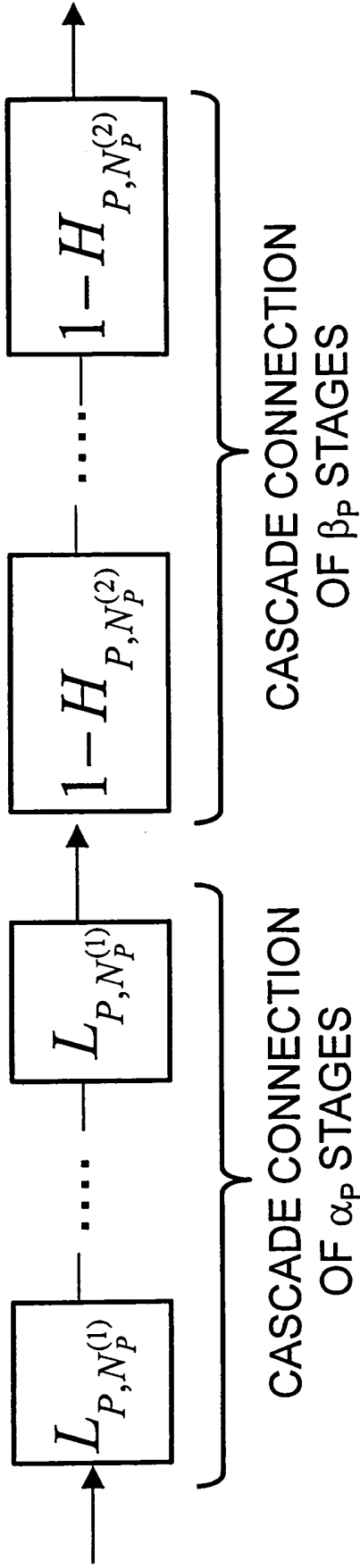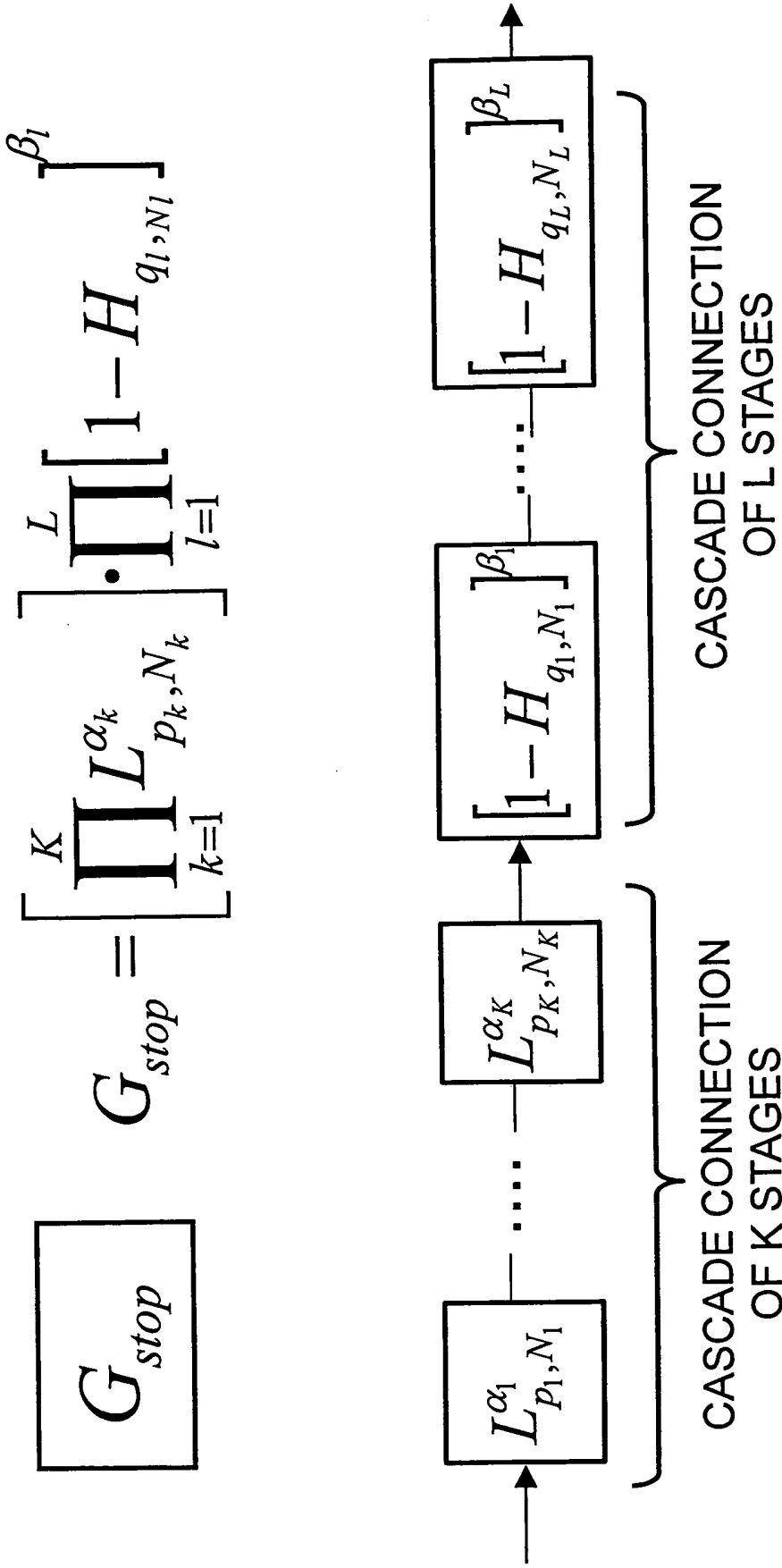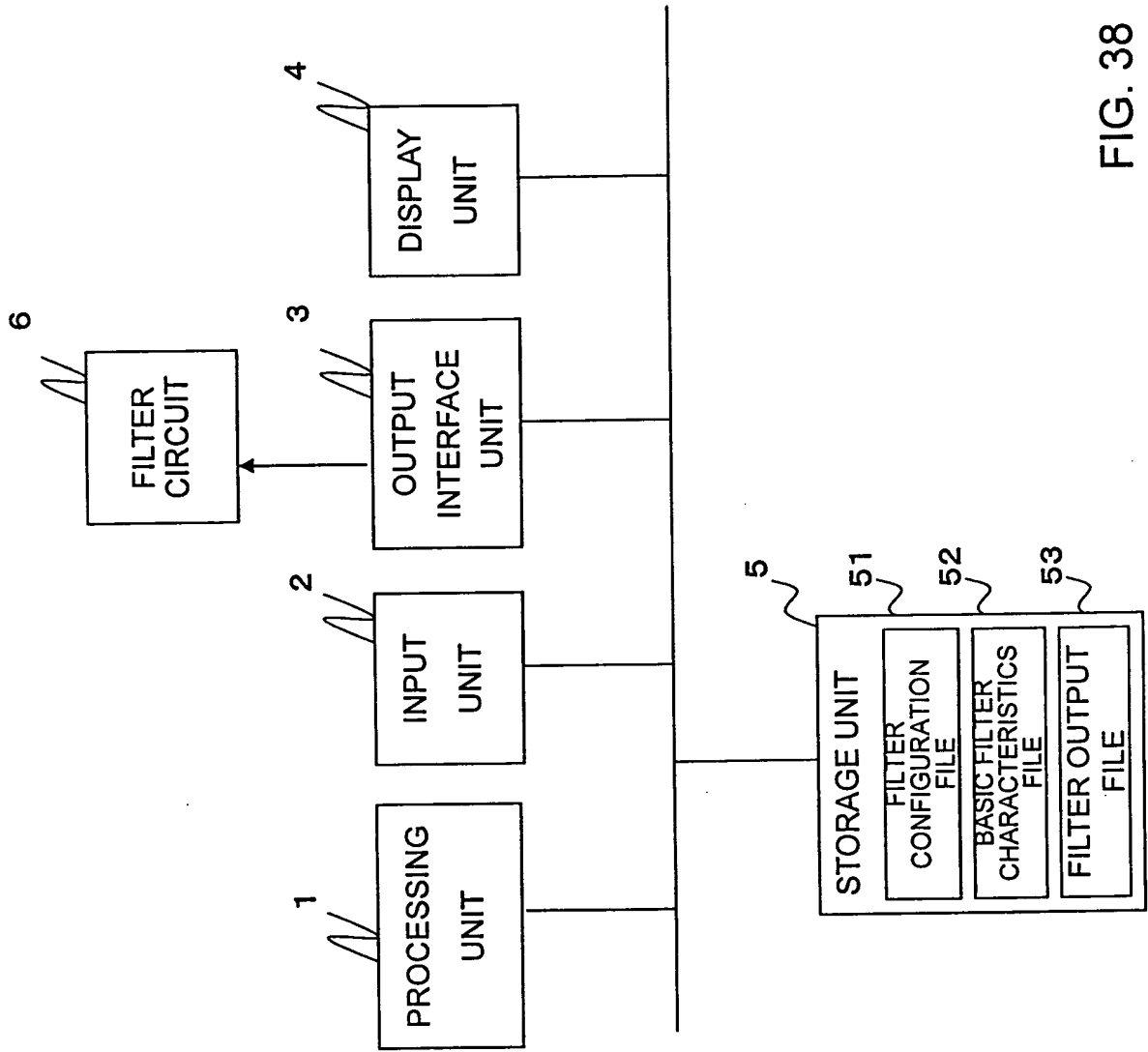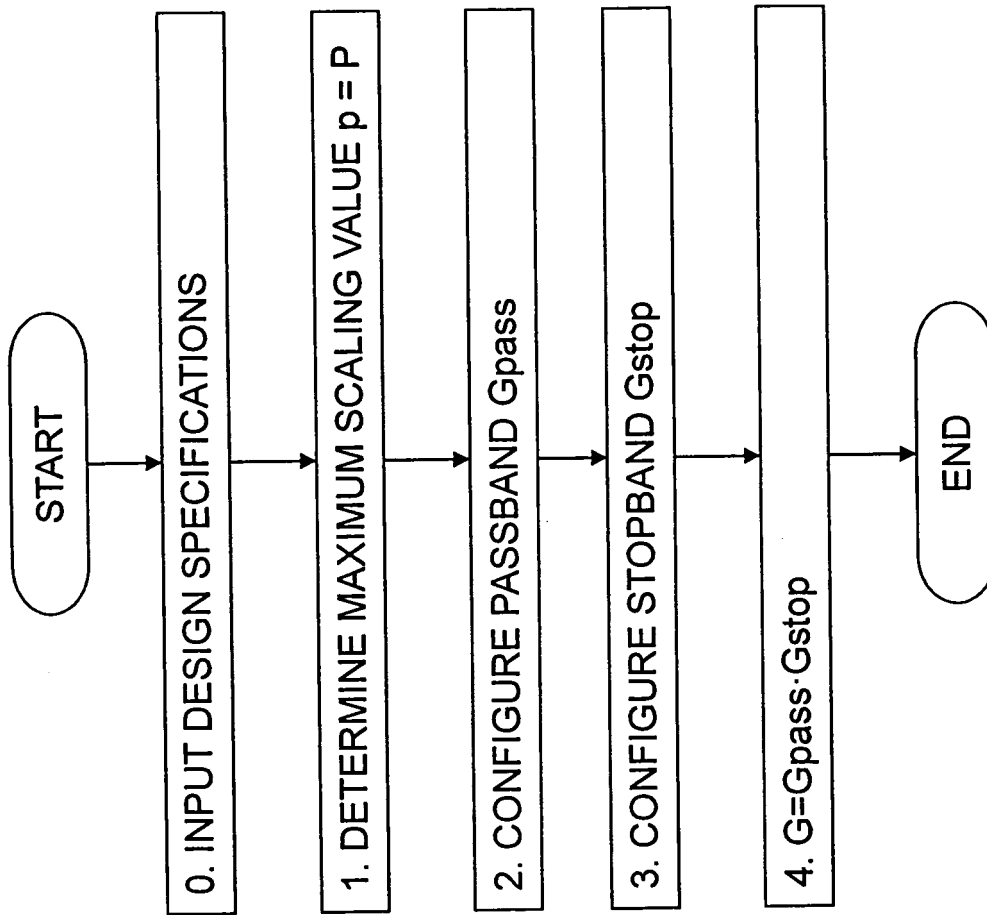
FIG. 39
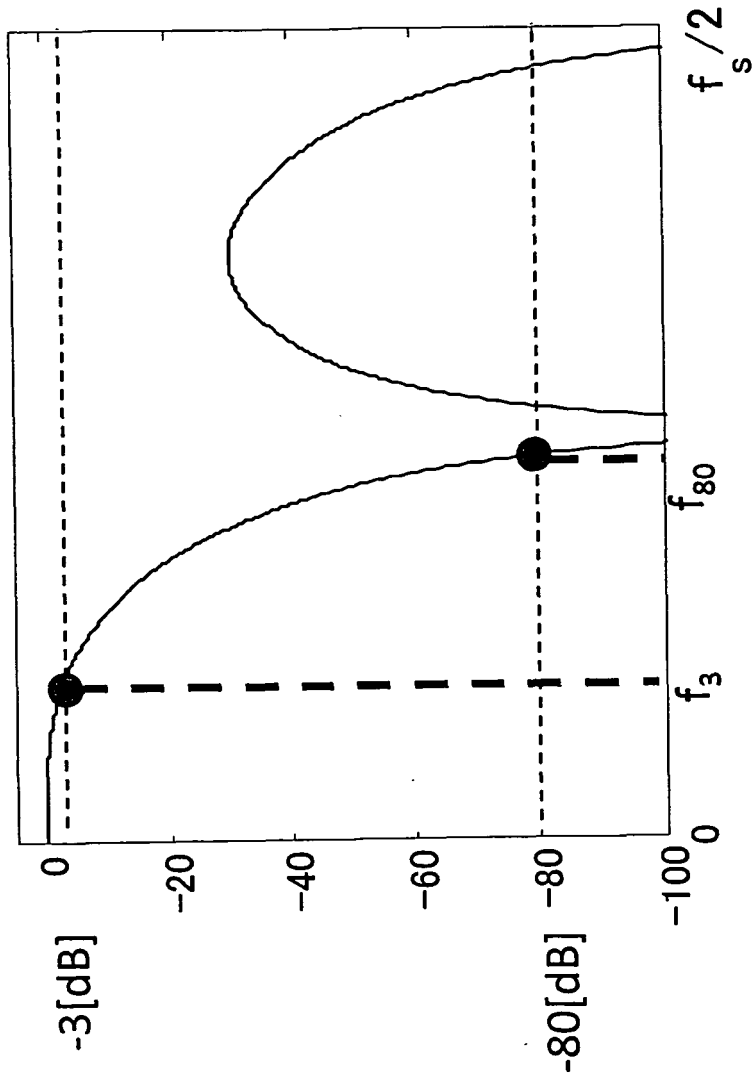
FIG. 40

SPECIFICATIONS (NORMALIZING $f_S/2$ TO 1)

PASSBAND: 0 TO 9/100 ... FLATNESS OF ±3 [dB]

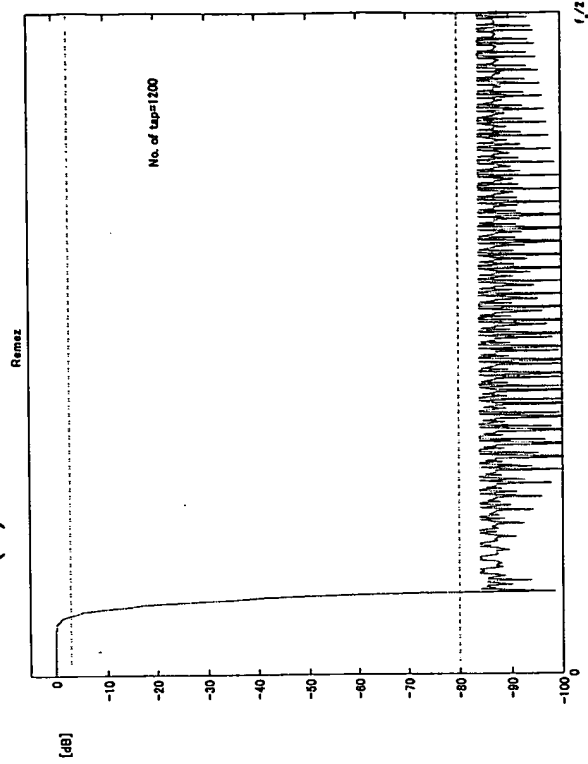STOPBAND: 14/100 TO 1 ... -80 [dB] OR BELOW

FIG. 41
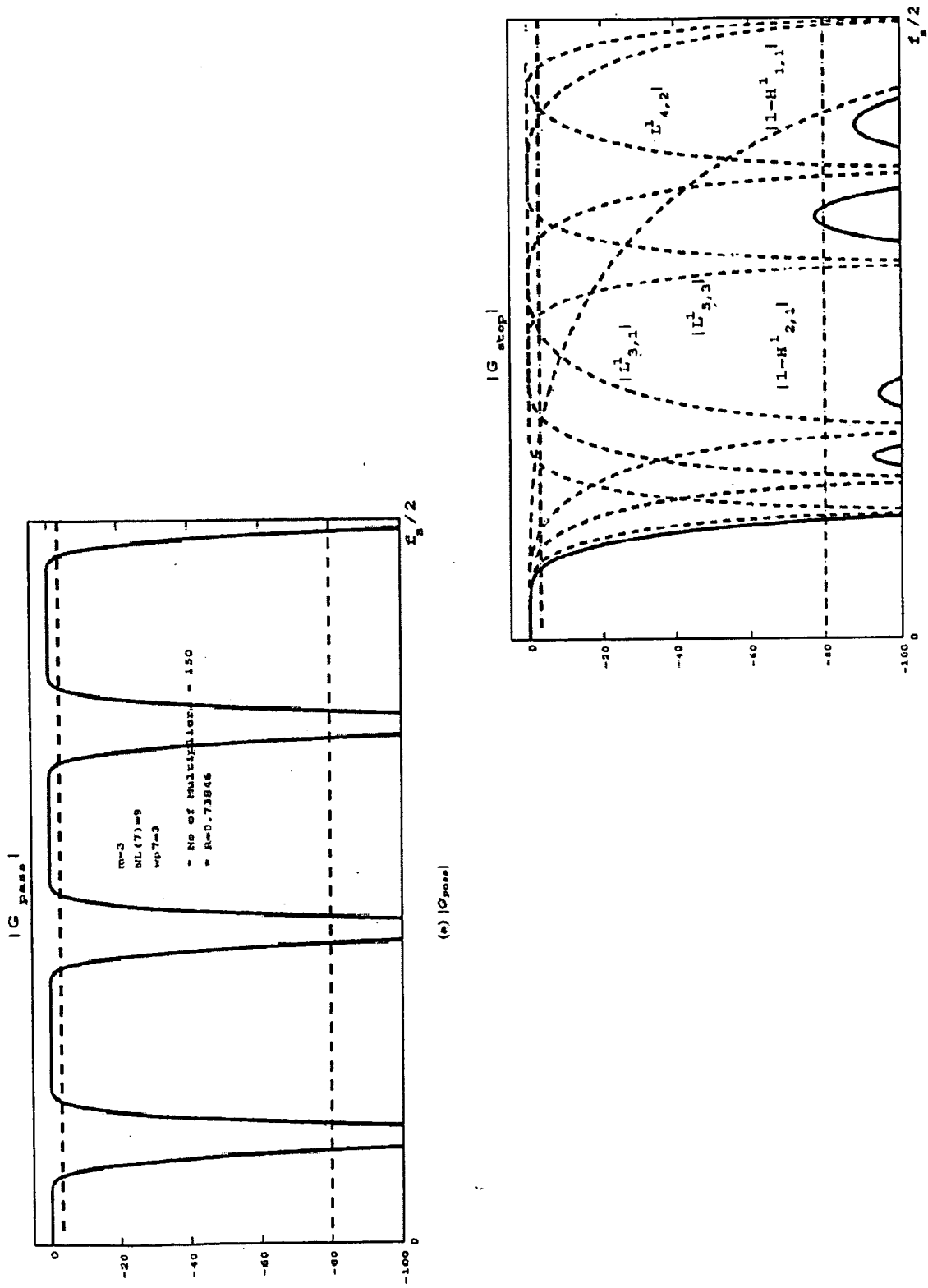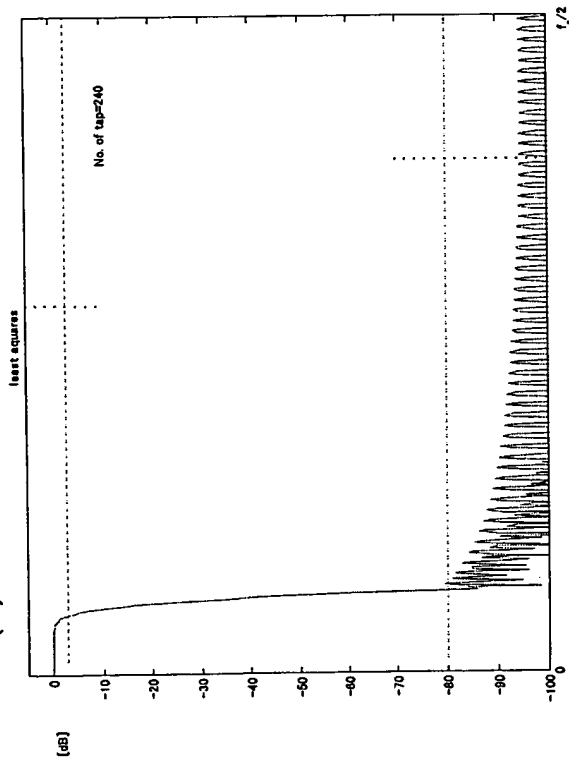
FIG. 42

(a) LEAST SQUARES METHOD

(b) Remez METHOD

(c) PRESENT INVENTION (fluency)
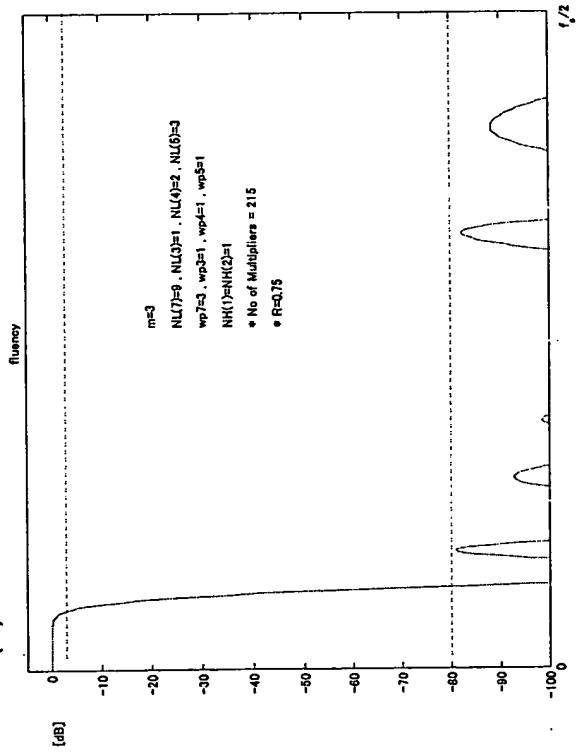
FIG. 43

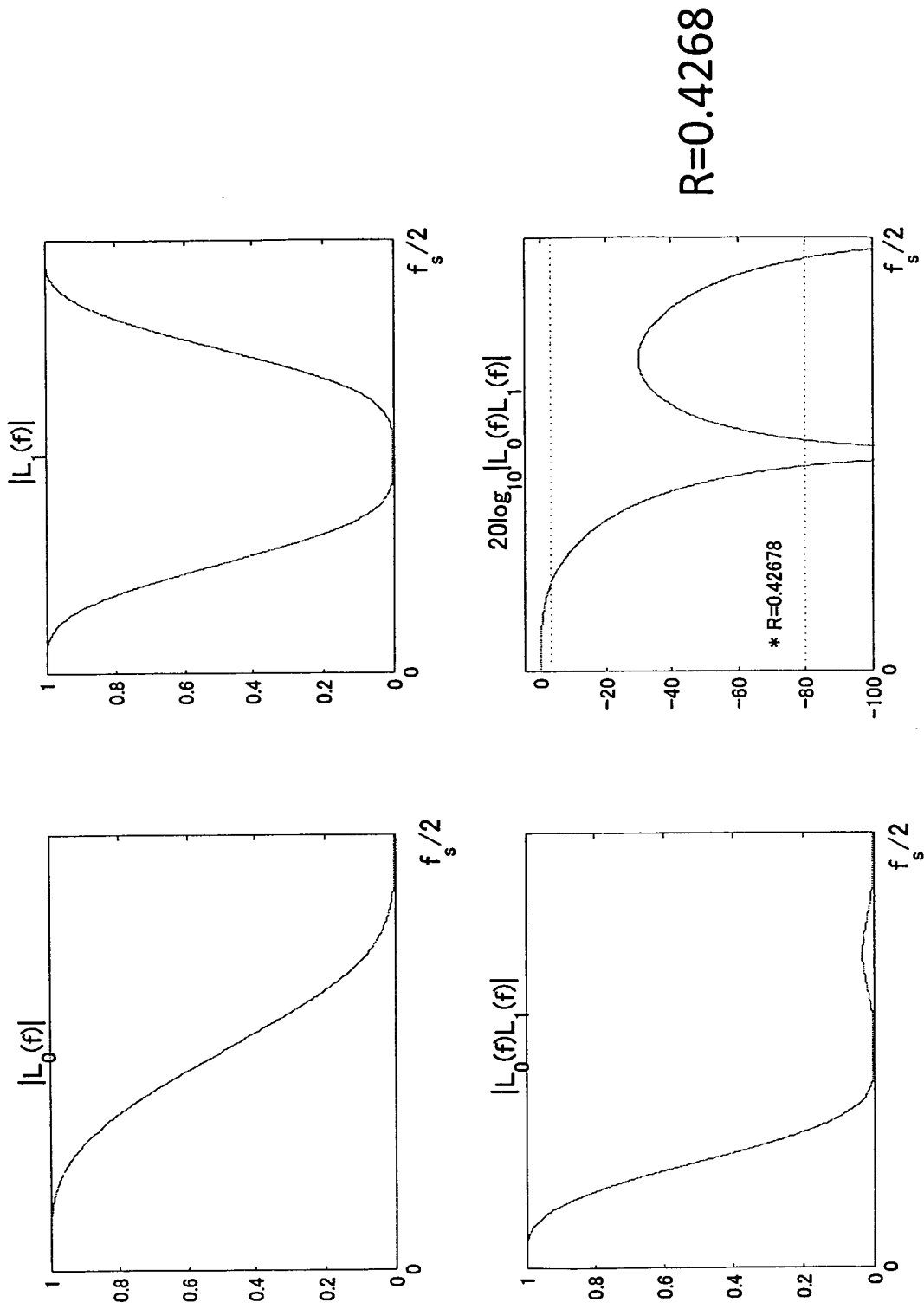|	NUMBER OF MULTIPLIERS	R
-LEAST SQUARES METHOD:	244	0.716
-Remez METHOD:	1150	0.762
-METHOD OF PRESENT INVENTION:	215	0.750
	365	0.767

FIG. 44

R=0.4268

FIG. 45

REFERENCES CITED IN THE DESCRIPTION

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