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Study of Microstrip filters

Deepak Kumar, P.K.Singhal, Rahul Srivastava, Devendra Soni

***Abstract*— This paper presents a direct design of Infinite Impulse Response filters (IIR filters) which minimizes group delay without changing the magnitude response of filters. In this paper Butterworth and Chebyshev1 lowpass filters are designed by using allpass filters. The design specifications are passband and stopband frequencies and passband ripple and stopband attenuation.**

***Index Terms*— All pass filter, Butterworth filter, Chebyshev1 filter, Equalized Group delay.**

I. INTRODUCTION

Filter is a frequency selective circuit that allows a certain band of frequency to pass while attenuating the others frequencies. Filters are classified as analog and digital. The Digital Filtering is one of the most powerful tools of DSP. The digital filters consist of software and hardware. The input and output signals in the digital filter is digital or discrete time variant. The procedure for designing digital filters involves the determination of a set of filter coefficients to meet a set of design specifications. Digital filters come in two flavors: FIR and IIR. As the terminology suggest, these classifications refer to the filters impulse response. By varying the weight of the coefficients and number of filter taps, virtually any frequency response characteristics can be realized with an FIR filter. FIR filters have a very useful property: they can exhibit linear phase shift for all frequencies. IIR filters have infinite impulse response. IIR filters have much better frequency response than FIR filters of the same order. In IIR filters their phase characteristics is not linear, which can cause a problem to the systems which need phase linearity. In this paper, a direct design of digital Butterworth and Chebyshev1 filter based on allpass filter is proposed. The design parameters of proposed method are shown in fig.(1)

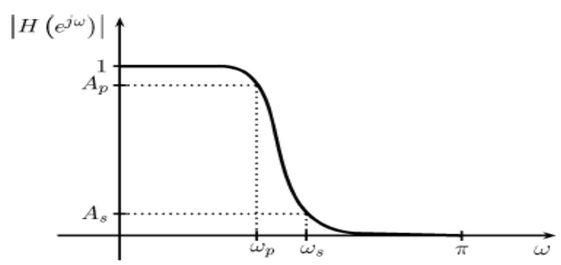

Fig.1- Design parameters

Where,
w_p- Passband frequency,
w_s- Stopband frequency,
A_p- Passband ripples,
A_s- Stopband attenuation.

II. BUTTERWORTH FILTER

The amplitude-squared transfer function for Butterworth filters that have an insertion loss L_{Ar} = 3.01 dB at the cutoff frequency $_c$ = 1 is given by

$$|S_{21}(j\Omega)|^2 = 1/(1+\Omega^{2n}) \quad (1)$$

where n is the degree or the order of filter, which corresponds to the number of reactive elements required in the lowpass prototype filter. This type of response is also referred to as maximally flat because its amplitude-squared transfer function defined in(1) has the maximum number of ($2n$ - 1) zero derivatives at = 0. Therefore, the maximally flat approximation to the ideal lowpass filter in the passband is best at = 0, but deteriorates as approaches the cutoff frequency $_c$. Fig.2 shows a typical maximally flat response.

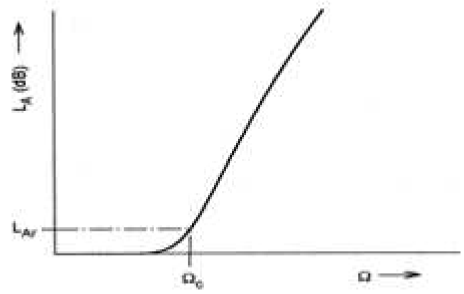

Fig. 2-Butterworth (maximally flat) lowpass response.

The Butterworth filters achieve its flatness at the expense of a relatively wide transition region from passband to stopband with average transient characteristics. This filter is completely defined mathematically by two parameters i.e. cut off frequency and number of poles. Compared to chebyshev filter, the phase linearity of buttorworth filter is better. In other words, the group delay (derivative of phase with respect to frequency) is more constant with respect to frequency. This means that the waveform distortion of the butterworth filter is lower. This Butterworth filters have the following characteristics.

The magnitude response is nearly constant (equal to 1) at lower frequencies. That means pass band is maximally flat.

- The response is monotonically decreasing from the specified cut off frequencies.

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Deepak Kumar, Arya College Of Engineering & I.T,jaipur
P.K.Singhal, MITS,Gwaliour
Rahul Srivastava, ACEIT,Jaipur
Devendra Soni, ACEIT,Jaipur Rajasthan Technical University, Kota, Rajasthan India

- The maximum gain occurs at $\Omega = 0$ and it is $|H(0)| = 1$.
- Half power frequency, or 3db down frequency, that corresponds to the specified cut off frequencies.

The magnitude squared response of low pass Butterworth filter is given by

$$|H(\Omega)| = 1/1+(\Omega/\Omega_c)^{2N} \quad (2)$$

Here $|H(\Omega)|$=Magnitude of analog low pass filter.

Ω_c=Cut-off frequency (-3db frequency)

Ω_p=Pass band edge frequency.

C=Parameter related to ripples in pass band.

N=Order of the filter.

The order of filter means the number of stages used in the design of filter. As the order of filter N increases, the response of filter is more close to the ideal response as shown in Fig.3.

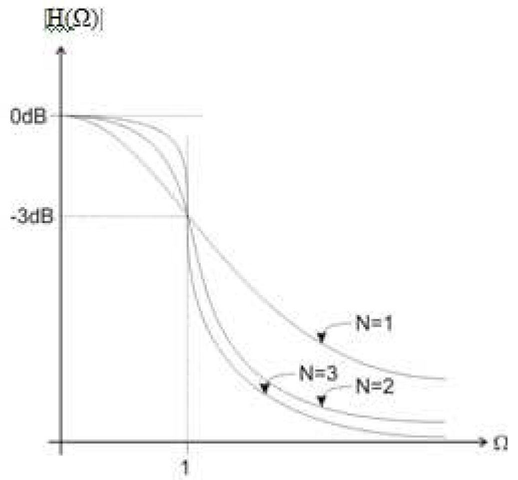

Fig.3- Effect of N on frequency response characteristics.

III. CHEBYSHEV TYPE1 FILTER

Chebyshev1 filters have a narrower transition region between the passband and the stopband. The sharp transition between filter lies on an ellipse. ripple increase (band), the roll-off becomes sharper(good). The chebyshev filter is completely defined by three parameters- cut-off frequencies, number of the passband and the stopband of a chebyshev filter produces smaller absolute errors and faster execution speeds than a butterworth filter. The poles of chebyshev1 poles and passband ripples. The chebyshev response is a mathematical strategy for achieving a faster roll off by allowing ripple in the frequency response. The chebyshev response is an optimal trade-off between these two parameters. The magnitude squared frequency response is given by

$$|H(\Omega)|^2 = 1/1+ C^2 C_N^2(\Omega/\Omega_p) \quad (3)$$

Here $|H(\Omega)|$=Magnitude of analog low pass filter.

C=Parameter related to ripples in pass band.

$C_N(x)$=Chebyshev polynomial of order N

The chebyshev1 polynomials are determined by using the equations

$$C_{N+1}(x) = 2x\, C_N(x) - C_{N-1}(x) \quad (4)$$

with $C_0(x)=1$ and $C_1(x)=x$

The following figure shows the frequency response of a lowpass Chebyshev1 filter.

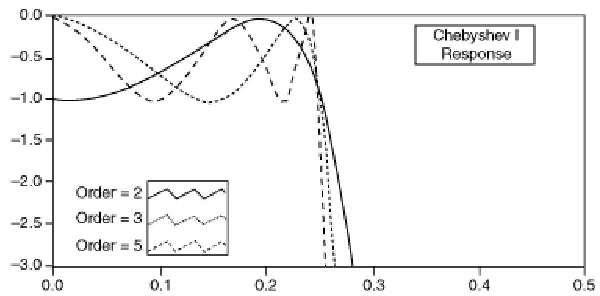

Fig.4-Effect of N on Chebyshev1 filter Characteristics

The Chebyshev response that exhibits the equal-ripple passband and maximally flat stopband is depicted in Fig.5. The amplitude-squared transfer function that describes this type of response is

$$|S_{21}(j\Omega)|^2 = \frac{1}{1+\varepsilon^2 T_n^2(\Omega)} \quad (5)$$

where the ripple constant is related to a given passband ripple L_{Ar} in dB by

$$\varepsilon = \sqrt{10^{\frac{Lar}{10}}} \quad (6)$$

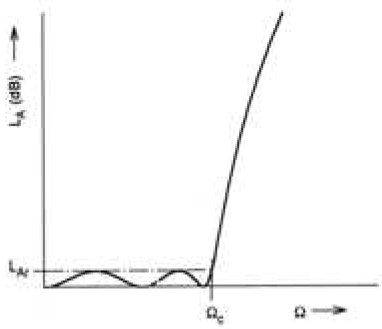

Fig.5- Chebyshev lowpass response

$T_n(\ )$ is a Chebyshev function of the first kind of order n, which is defined as

$$T_n(\Omega) = \begin{cases} \cos(n \cos^{-1}\Omega) & |\Omega| \le 1 \\ \cosh(n \cosh^{-1}\Omega) & |\Omega| \ge 1 \end{cases}$$

Hence, the filters realized from (5) are commonly known as Chebyshev filters. Rhodes [2] has derived a general formula of the rational transfer function from (5) for the Chebyshev filter, that is

$$S_{21}(p) = \frac{\prod_{i=1}^{n}[\eta^2+\sin^2(i\pi/n)]^{1/2}}{\prod_{i=1}^{n}(p+p_i)} \quad (7)$$

with

$$P_i = j\cos\left[\sin^{-1} j\eta + \frac{(2i-1)\pi}{2n}\right] \quad (8)$$

$$\eta = \sinh(1/n \sinh^{-1} 1/\varepsilon) \quad (9)$$

Similar to the maximally flat case, all the transmission zeros of

$S_{21}(p)$ are located at infinity. Therefore, the Butterworth and Chebyshev filters dealt with so far are sometimes referred to as all-pole filters. However, the pole locations for the Cheby- shev case are different, and lie on an ellipse in the left half-plane. The major axis of the ellipse is on the *j* -axis and its size is 1 + 2; the minor axis is on the -axis and is of size . The pole distribution is shown, for $n = 5$, in Fig.6.

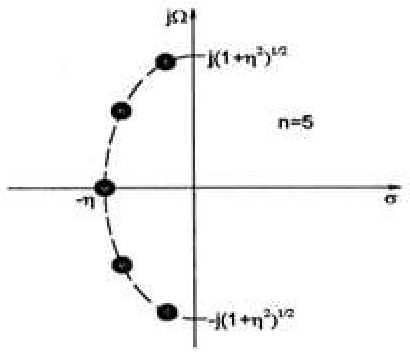

Fig.6- Pole distribution for Chebyshev response.

IV. ELLIPTIC FUNCTION RESPONSE

The response that is equal-ripple in both the passband and stopband is the elliptic function response, as illustrated in Figure 3.5. The transfer function for this type of response is

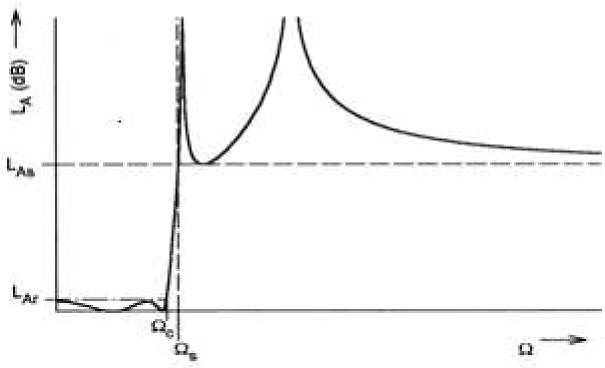

Fig.5- Elliptic function lowpass response.

V. ALLPASS FILTER

The allpass filter is an important building block for signal processing system. The magnitude response of an allpass filter is unity over its entire frequency range. In other words, all frequencies are "passed" in the same sense as in "lowpass", "highpass" and "bandpass" filters. The phase response (which determines the delay versus frequency) is variable. The allpass filters are typically appended in a cascade arrangement following a standard IIR filter H1(z) as shown in Fig.7.

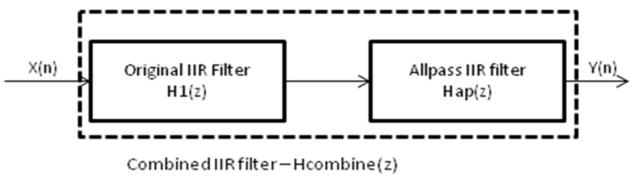

Fig 7- Cascade arrangement of original IIR filter and allpass filter.

The transfer function of IIR allpass filter with unity magnitude response for all frequencies is given by |A(ejw)|2=1 for all w. The Nth order causal real coefficient allpass transfer function is given by equation

$$A_N(z) = \pm \frac{d_N + d_{N-1}z^{-1} \text{-------------}+ d_1 z^{-N+1} + z^{-N}}{1 + d_1 z^{-1+}\text{-------------} d_{N-1} z^{-N+1} + d_N z^{-N}} \qquad (11)$$

If the denominator polynomials of $A_N(z)$ is denoted by $D_N(z)$ then equation (11) can be written as

$$A_N(z)=\pm z\text{-}ND_N(z\text{-}1)/D_N(z) \qquad (12)$$

The numerator of the allpass filter is the mirror image polynomial of the denominator.

$$D_N(z) = z\text{-}ND_N(z\text{-}1) \qquad (13)$$

Equation (13) implies that poles and zeros of a real coefficient allpass function exhibit mirror image symmetry in the z plane.

$$|S_{21}(j\Omega)|2 = \frac{1}{1+\varepsilon 2Fn2(\Omega)} \qquad (10)$$

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