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Design of Microwave Filters Using DSP Technique

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***Abstract*— In this paper, a novel method for the design of microwave filters using digital signal processing technique with arbitrary frequency response is presented. It is mainly based on the conversion of the microwave specifications to the digital specification, where the digital filter design technique mainly IIRPNORM algorithm is applied. With the use of digital signal processing techniques, the method calculates the poles and zeros for the analog frequency response that satisfies the given specifications. By the poles and zeros, the microwave filter can be realized using conventional techniques. As an example to demonstrate the proposed technique, a filter with user-defined specifications over two independent passbands has been implemented and this would be tested in microstrip technology.**

***Index Terms*— Digital signal processing technique, *a*rbitrary frequency response, digital filter design, microwave filter.**

I. Introduction

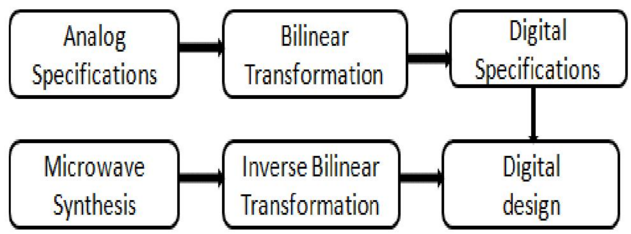
MICROWAVE filters with their various responses can be fabricated in different forms; examples are band pass or low-pass filters with non-uniform transmission lines and band pass or band stop filters with coupled lines. Among conventional procedures for designing microwave filters, either the image-parameter method or the insertion-loss method provides the configuration of lumped-element circuits.

Once the values of lumped elements are obtained, Richard's transformation is generally used to convert lumped filter elements to stubs of the same electrical length. In addition, Kuroda's identities can be used to separate the stubs with transmission-line sections of the same electrical length as that of the stubs. In recent years, filter design has been studied in the z-domain, by replacing the phasor $e^{-j\theta}$ with the delay operator, and several effective interesting methods have been exploited. Actually, when stubs and transmission-line sections are manufactured with the same electrical length and are cascaded orderly, parameters describing the characteristics of the network can also be represented as functions of the delay operator. Therefore, abundant assortment of well-developed discrete-domain techniques can be applied to the studies of such networks. Upto now, the systematic design of microwave filters has been mainly limited to the use of the well-known classical functions, i.e. Butterworth, et al [1],[2]. However, more sophisticated and general functions like Zolotarev have been also explored [3] and specifically, for the case of multipassband filters, some limited analytical techniques have been proposed [4], although usually direct optimization methods have been applied for these cases, with the involved difficulties [5], [6].

Now we demonstrate that microwave filters with especially demanding magnitude responses can be synthesized by using the well established and readily available digital filter design techniques [7]. The proposed methodology rests on the translation of the target specifications from the analog to the digital domain, where the microwave designer can take immediate advantage of these sophisticated and continuously developing digital techniques, without requiring a deep understanding of the complex mathematics involved. The details of the novel filter design technique will be explained.

II. Microwave Filter Design

This design technique would be achieved by the following steps as shown in Fig.1.



~~Fig. 1. Proposed Design Technique~~

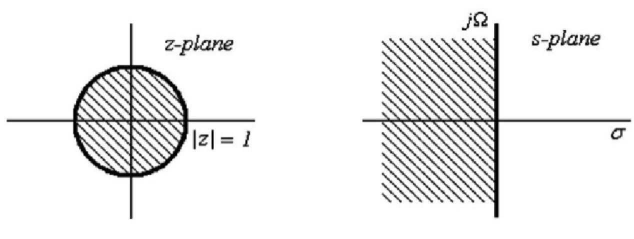


Fig. 2. Mapping between z-plane and s-plane

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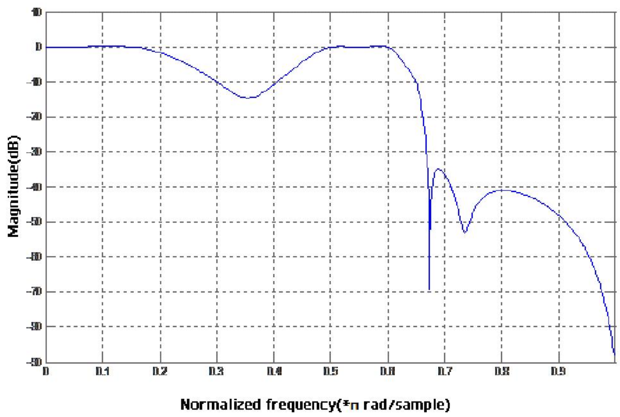

Fig.3. Digital Frequency Response using IIRPNORM Algorithm

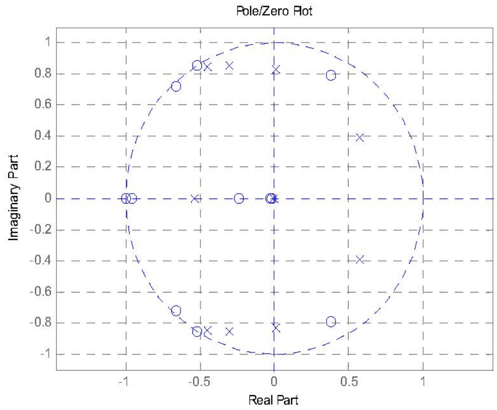

Fig.5. Pole-Zero Plot of Frequency Response

A. *From Analog Domain to Digital Domain*

The desired microwave filter is characterized by a set of analog specifications. From them, the digital design specifications are obtained by means of a bilinear transformation [7].

$$s = a.\frac{z-1}{z+1} \tag{1}$$

$$\text{where} \quad a = \frac{2\pi.f_p}{\tan\left(\frac{f_p}{f_s}\right)}$$

Equation (1) is an algebraic transformation between the analog Laplace-transform s-plane and the digital z-transform plane. In order to relate the analog frequency f[Hz], with the digital frequency ω [rad /sample], has to be valued at s= $j\Omega$, where $\Omega = 2\pi.f$ and $z = e^{j\omega}$ [7]. The bilinear transformation employs two parameters: 1) the sampling frequency fs, which must satisfy the Nyquist criterion ($f_s \geq 2.\ f_{max}$, being f_{max} the maximum frequency in the specifications with significant spectral content) and 2) the pre-warping frequency f_p, which can be fixed to make the digital requirements more relaxed, as it will be shown later. To demonstrate the proposed design methodology, an 11th order filter is being designed which would be implemented in micro strip technology. Once the analog specifications are chosen pass bands from dc to 1GHz and from 4 to 5.5GHz , with return losses around 15dB; and stop bands from 2 to 3GHz with rejection level around 10dB, from 7 to 9GHz with rejection level better than 40dB. With f_s=2.f_{max}=14GHz, f_p fixed at 2.4GHz to make the digital specification more relaxed at the first stop band.

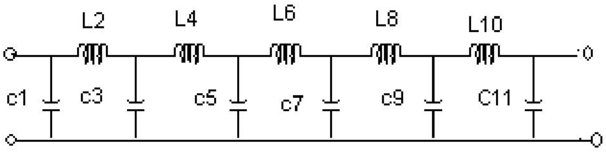

Fig..4. Ladder network of lumped-element.

B. *Digital Design*

There are many sophisticated and continuously developing techniques to design a digital filter for the digital specifications obtained in the above subsection. By IIRPNORM algorithm, available in MATLAB, which minimizes the error function defined as the p-norm of the weighted difference between the target filter frequency response and the designed one, is shown in Fig.3. The value of p is increased step-by-step so that the resulting design tends to minimize the maximum modulus of the error function. The algorithm provides some parameters that allow us to control the design process with high flexibility. The most relevant ones are the filter order, the weighting vector and the range of values for p. As a result, the poles and zeros of the digital filter are obtained is shown in Fig.5.

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C. *Digital to Analog conversion*

The bilinear transformation used in Section II-A has been also typically employed to transform classical analog filters into recursive digital filters. Using this transformation in the opposite way, it will transform the digital filter into the pursued analog filter. This gives rise to the inverse bilinear transformation, as defined in the following equation:

$$z = \frac{a+s}{a-s} \tag{2}$$

By using the above formula the analog poles and zeros are obtained from the digital poles and zeros.

The inverse bilinear transformation has the following advantageous features:

1) *Mapping Properties:* Inspecting (2), three important regions can be distinguished as depicted in Fig. 2: the shaded (white) region in the z-plane is mapped to the shaded (white) region in the s-plane, while the unit circle in the z-plane is mapped to the imaginary axis in the s-plane. As a consequence, it can be shown that a causal and stable digital

filter is transformed into a causal and stable analog filter as intended.

2) *Warping Effect:* As it can be easily inferred, the analytical relationship that holds between the digital and the analog frequencies is not linear. This produces the so-called warping effect. However, since the whole design method includes also the bilinear transformation in Section II-A, the total effect cancels out completely, providing a distortion-free procedure.

D. *Synthesis of the Microwave Filter*

From the result in Section II-C, it is converted into a transfer function S_{21} (s) of desired response of analog filter. The ladder network of lumped elements that implements the frequency response is shown in Fig.4. The values of the elements L, C are easily obtained by using the classical continuous fraction expansion method [8]. The capacitance values in pF are: C1=1.428, C2=2.389, C5=0.691, C7=0.709, C9=2.25, C11=1.29. The inductance values in nH are: L2=1.079, L4=2.429, L6=6.419, L8=2.381 and L10=1.16. The microwave filter will be implemented as a stepped-impedance filter in micro strip technology after deriving the associated ladder network of lumped elements. However, it is important to stress that other technology (even non-planar) and/or implementation schemes could be used [2]. It can be implement as a stepped impedance filter in two wire transmission line.

The frequency response of the lumped-element network has been simulated using Zeland IE3D as shown in Fig.6. As it can be seen, it matches with the desired analog response. After that the microwave filter will be finally implemented as a stepped-impedance filter in micro strip technology using a CUCLAD-250 LX substrate as transmission line alternating high impedance and low impedance sections [9].

III. CONCLUSION

We have simulated a novel design technique to design microwave filters with arbitrary frequency response based on digital signal processing technique. The novel technique allows the microwave designer to take immediate advantage of the efficient and sophisticated digital filter design methods. This technique is successfully implemented in two wire transmission line and stepped impedance filter technology. It can also be implemented in other technologies and gives a better solution to this competitive world.

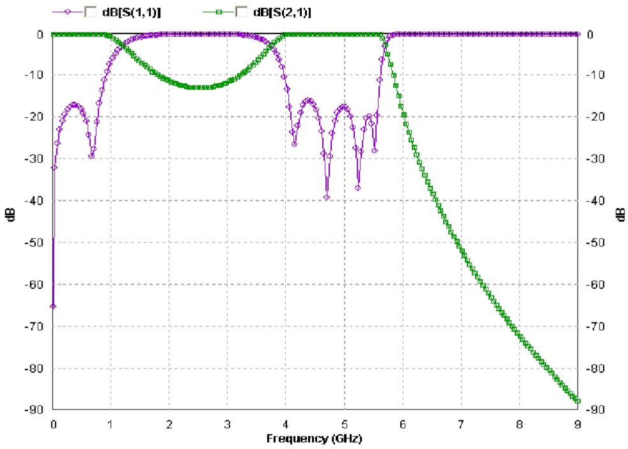

Fig.6. Simulated Parameter of Lumped-Element Ladder Network

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