## FIR transformation from analog domain to discrete-time domain

Nikzad Babaii AbdolReza Nabavi Department Of Electrical Engineering Tarbiat Modares University Jalal ale Ahmad Highway Tehran ,Iran

<u>Abstract</u>: In this letter, we have introduced an approximating method for FIR transformation from continuous-time filters to discrete-time filters. We have used fourier series expansion for approximation and we have founded FIR relation as z for replacing instead of s in s-plane.

## **1.Introduction**

one method to design digital filter is the design of it in the analog domain and then take advantage of their relationship to continuous-time filters made up of networks of discrete elements.convert the design into the digital domain. Based on this idea, several transformation were created. IIR filter e.g. design approximation of derivatives, IIR filter design by impulse invariance, bilinear transformation. and matched-z transformation [1,3]. which all of them are IIR transformation .In this letter ,we are going to find a FIR transformation by approximation.

#### 2. Approximation approach

we have from mathematics that any signals can approximated by series of cosine and sine functions [2].If O is frequency in continuous-time domain and ? is frequency in discrete-time domain ,we

$$\mathbf{w} \cong \sum_{k=1}^{M} a_k \sin(k\mathbf{w})....(1)$$

have ?=O.T where T is period of sampling rate[2]. thus s=j?/T is definition of s in discrete-time domain. it is seen that s only has component in image axis. also, the ? function is a odd function and approximation it by fourier series has only sine terms .We obtain

$$k = 1, 2, \dots, M \dots (3)$$
$$a_k = \frac{1}{p} \int_0^p \mathbf{w} \sin(k\mathbf{w}) d\mathbf{w}$$

with to multiply j in the both side of the above equation

$$j\boldsymbol{w} \cong j\sum_{k=1}^{M} a_k \sin(k\boldsymbol{w})$$

with to notice to following relations

so,

$$s = j\boldsymbol{w}, \sin(k\boldsymbol{w}) = \frac{z^k - z^{-k}}{2j} \Big|_{z=e^{j\boldsymbol{w}}}$$

by replacing the above statement instead of s in analog filter, we get equivalent digital filter.

## **3.** Comparsion

we suppose an analog filter with transfer function (T=0.5)

Fig .1 and Fig .2 show magnitude and

$$H(s) = \frac{s + 0.4}{2s + 9}$$

phase of founded digital filter with different transformation .the final H(z) is as the form

## **4.**Conclusion

Other transformations only poise the universal charastrictics of te continuoustime filters (e.g. transition band ,...)and change the magnitude and phase shaps.but this method protects them. This method in comparison has greater degree than the others.

$$s \cong \frac{\sum_{k=1}^{M} a_k(\frac{z^k - z^{-k}}{2})}{T}....(5)$$

that

#### References :

- 1 I.G.Proakis ,D.G.Manolakis ,digital signal processing principles ,algorithms ,and applications,3nd Ed.,John Wiley & Sons Inc. ,1996
- 2 R.K.Livesley ,Mathematical methods for engineers ,(Ellis Horwood imited West Sussex , 1989
- 3 N.Babaii ,E.BinaeeBash ,Sh.Bayani ,Different methods for approximation signals,2th ISCEE conf. ,1999,pp.52-58

# **Figure caption**

Fig 1. The magnitude of the analog filter with several transformation

- o : analog filter
- . : approximation of derivatives
- -. : bilinear
- \* : this method

Fig 2. The phase of the analog filter with several transformation

- o : analog filter
- . : approximation of derivatives
- -. : bilinear
- \* : this method

# Table caption Table 1. The coefficients of this method $(a_k)$

Coefficient	Value
$A_0$	-0.0488
A <sub>1</sub>	0.0610
A <sub>2</sub>	-0.0812
A <sub>3</sub>	0.1219
A <sub>4</sub>	0.5063
A <sub>5</sub>	0
A <sub>6</sub>	-0.5063
A <sub>7</sub>	-0.1219
A <sub>8</sub>	0.0812
A <sub>9</sub>	-0.0610
A <sub>10</sub>	0.0488

Table .1

Fig .1

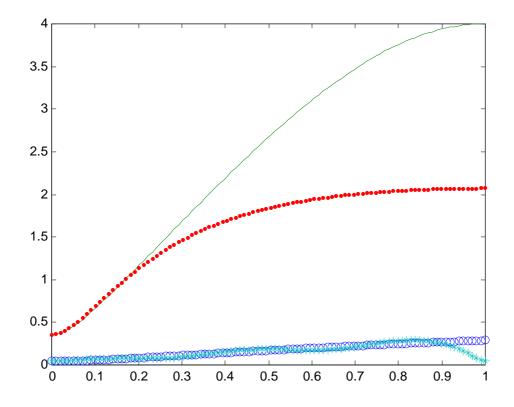


Fig.2

