Digital Signal Processing of Time Domain Field Simulation Results Using the System Identification Method

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Abstract

A new technique for significantly reducing the computation time of time domain field simulations using modern digital signal processing techniques is presented. The procedure leads to a simple model of the investigated structure in form of a digital filter. As an example the method is applied to a TLM simulation of a resonant cavity.

Introduction

Due to the simple theoretical basis and the great flexibility time domain methods as e.g. the TLM method (Transmission-Line Matrix) [1] and the FDTD method (Finite Difference Time-Domain) [2] get more importance. However, the amount of CPU time and memory required is excessive.

The present paper shows that it is possible to reduce the number of time steps by a factor of 20 up to 64 for eigenvalue problems using the TLM simulation associated with the System Identification (SI) method [3, 4]. The SI-method is able to determine a model of the microwave structure from a short observation intervall of the input and output data of the time domain simulation. Nearly arbitrarily long output signals can be calculated in a very short time by using this model which is a simple digital filter.

Theory

The TLM and FDTD meshes can be interpreted as high order digital filters with accessible input and output signals $(x(t), y(t))$.

The delay time t_0 of the original system is represented by a delay line of the order $(t_0 - 1)$ in the model

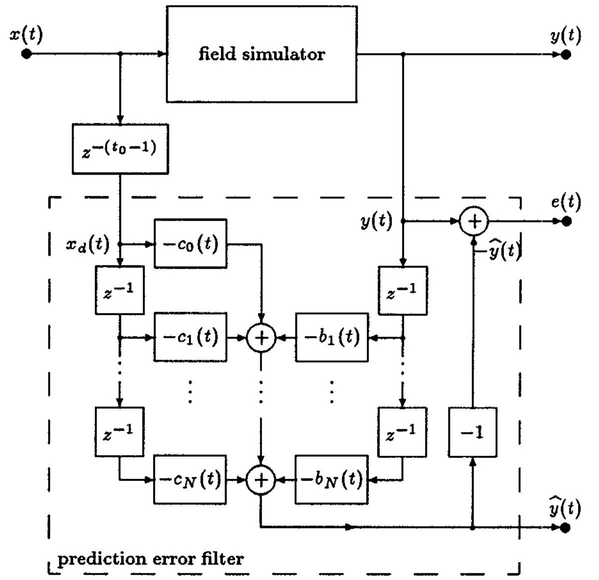

Figure 1: System identification with prediction error filter and a delay line

(Fig. 1). The remaining part is approximated by a digital filter of the order N which is determined with the SI-method. The input signals of this approach are $x_d(t) = x(t - t_0 + 1)$ and $y(t)$. Usually the identification problem is solved by predicting the output of the system from previous samples of the input and output signal of the original system (Fig. 1):

$$\hat{y}(t) = \sum_{i=0}^{N} -c_i(t) \cdot x_d(t-i) + \sum_{i=1}^{N} -b_i(t) \cdot y(t-i). \quad (1)$$

In the above equation $\hat{y}(t)$ is the prediction of the original signal $y(t)$. The prediction error is defined by:

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$$e(t) = y(t) - \hat{y}(t). \qquad (2)$$

The basic problem of fitting the model to the input data is to find a set of predictor coefficients ($c_i(t), b_i(t)$ with $i = 0, \dots, N$) that will minimize the sum of squared prediction errors (least-squares (LS) prediction):

$$\sum_{n=0}^{L-1} e(t-n) \cdot e(t-n) = \mathbf{e(t)} \cdot \mathbf{e(t)} \rightarrow min. \qquad (3)$$

$\mathbf{e(t)}$ is the vector of past observations of $e(t)$ weighted with a rectangular window of the lenght L:

$$\mathbf{e(t)} = (e(t), e(t-1), \cdots, e(t-L+1))^T. \qquad (4)$$

The adaptive prediction error filter (Fig. 1) is calculated with the first splitted generalized LeRoux-Gueguen Ladder algorithm for vectorial signals [4] which is based on the covariance and cross covariance matrix of $x_d(t)$ and $y(t)$. The algorithm belongs to the class of pure order recursive ladder algorithm (PORLA) and leads to the exact LS-ladder form [5, 6] of the prediction error filter that minimizes the sum of squared prediction errors (eq. (3)). The generalized vectorial Levinson recursion [5] is used to compute the transversal form of the prediction error filter shown in figure 1 from the ladder form. The coefficients $c_i(t)$ and $b_i(t)$ are time-dependent and converge relatively fast to the true parameters.

The application of the SI-method to a TLM simulation is described in the following points:

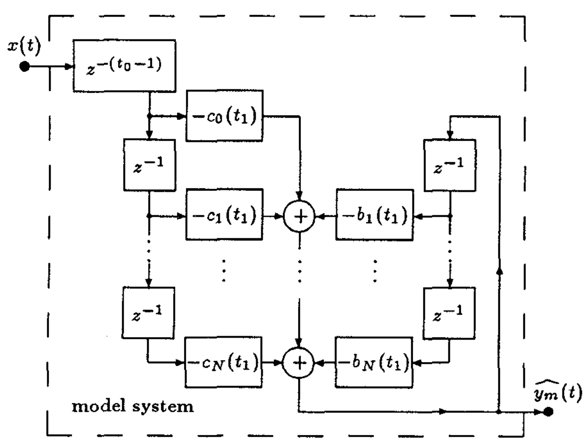

Figure 2: Model system of the microwave structure

1. First the time domain method is started to calculate $y(t)$ as a reaction of $x(t)$.
2. The delay time t_0 is defined when the first value of $y(t)$ is not equal to zero.
3. Now the SI-method is started which operates in synchronism with the field simulator.
4. The field simulator and the SI-method are stopped at the time $t = t_1$ if the variation of the coefficients $(c_i(t), b_i(t))$ is negligible and the error $e(t)$ is small enough.
5. Using the prediction error filter with the coefficients $(c_i(t_1), b_i(t_1))$ at the time step t_1 the model shown in figure 2 is determined. $\widehat{y_m}(t)$ can be calculated for nearly arbitrary long time by exiting the model with $x(t)$. The knowledge of the original output signal $y(t)$ is not necessary here.
6. $\widehat{Y_m}(\omega)$ or the frequency response $\widehat{H}(\omega)$ can be computed from $\widehat{y_m}(t)$.

Results

To demonstrate the function and the efficiency the SI-method is applied to a TLM simulation of a resonant cavity [7]. The field of the investigated structure was simulated using a mesh of 288 nodes over 16384 time steps. To observe $y(t)$ the mesh was exited with Dirac's delta function $x(t) = \delta(t)$ (eigenvalue analysis).

The time domain simulation results are processed by the method described above. The delay time is $t_0 = 7$.

If the filter order is chosen to $N = 40$ the error $e(t)$ is relatively small compared to $y(t)$ which indicates a good prediction (Fig. 3). After about 200 time steps the coefficients of the filter are nearly constant (Fig. 4). A comparison of the original spectrum $Y(\omega)$ (Fig. 5) with the estimated one $\widehat{Y_m}(\omega)$ (Fig. 6) shows that the frequency of the three significant eigenvalues of the original spectrum are predicted quite good (see also table 1). A filter order much smaller than $N = 40$ yields to inaccurate results.

With a filter order of $N = 70$ the error $e(t)$ can be reduced significantly. As shown in figure 7 and table 1 the agreement of the frequency response $\widehat{Y_m}(\omega)$ to the original one is excellent. The resonant frequencies can be calculated exactly and the amplitude with a small error. To determine the filter coefficients 500...1000

N	1.eigenvalue	2.eigenvalue	3.eigenvalue
40	0 Δf	1 Δf	13 Δf
70	0 Δf	0 Δf	0 Δf

Table 1: Comparison of the eigenvalues: Deviation of the resonant frequency in $\triangle f = 2^{-13} f_{max} = 61.035 MHz$

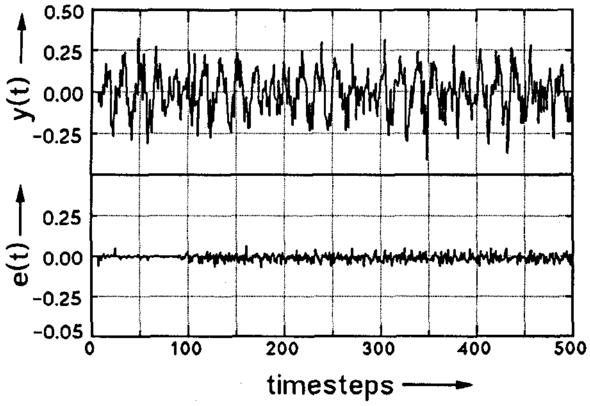

Figure 3: Original output signal $y(t)$ and prediction error $e(t)$

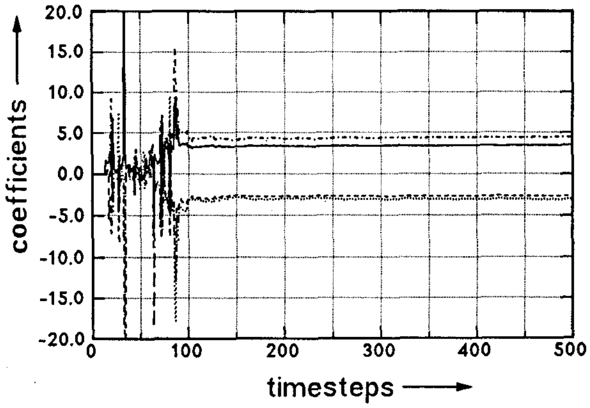

Figure 4: Several filter coefficients $b_i(t)$ with $i = 4, 5, 6, 7$

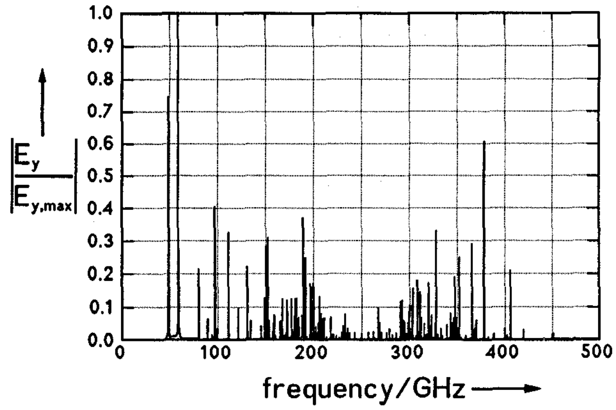

Figure 5: Original output spectrum $Y(\omega)$. The amplitude is normalized to the maximum of the electric field $E_{y,max}$.

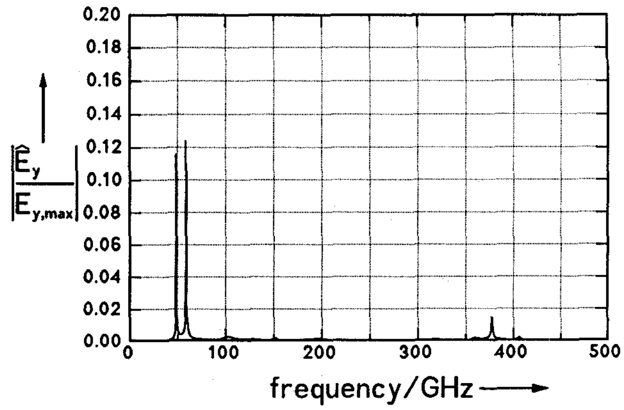

Figure 6: Estimated output spectrum $\widehat{Y_m}(\omega)$ with $N = 40$. The amplitude is normalized to the maximum $E_{y,max}$ of the original spectrum.

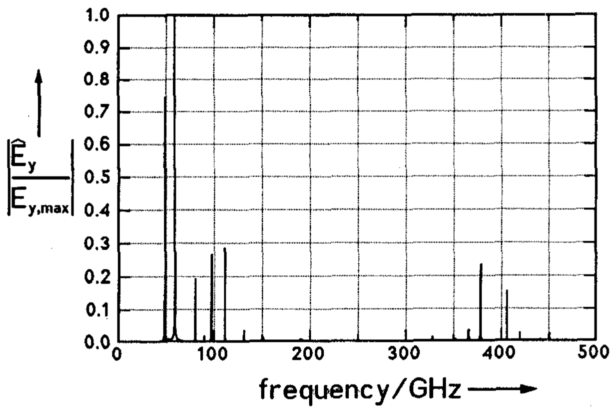

Figure 7: Estimated output spectrum $\widehat{Y_m}(\omega)$ with $N = 70$. The amplitude is normalized to the maximum $E_{y,max}$ of the original spectrum.

time steps are necessary. An increase of the based information lead to a more accurate estimation of the spectrum.

Conclusions

In the present paper the digital signal processing of time-domain simulation results by using the SI-method was introduced first. The computation time of time domain simulations of eigenvalue problems can be reduced by a factor of 20 (for $N = 70$) to 64 (for $N = 40$).

In the literature [8, 9, 10] only the Prony-Pisarenko method is known to reduce the number of required time steps of time-domain simulations. In [8] the overall CPU time is reduced by a factor of 2 to 3.

It is expected that the SI-method can easily be adapted to the application of transient analysis with e.g. the FDTD method [2]. Furthermore a model of the investigated structure is determined which might be useful for time-domain circuit analysis. Thus the SI-method is a new powerful tool for modern time-domain simulations and seems to be superiour compared to the Prony-Pisarenko method.

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