IEEE International Symposium on Electromagnetic Compatibility, Denver, Colorado, Aug. 24-28, 1998, pp. 882-887
¾ ¾

Identification of Nonlinear Interference Sources with the Use of the Discrete Technique

Sergey Loyka
Belorussian State University of
Informatics & Radioelectronics,
Brovki Str. 6,
Minsk 220027, BELARUS

Vladimir Mordachev
Belorussian State University of
Informatics & Radioelectronics,
Brovki Str. 6,
Minsk 220027, BELARUS

Abstract: This paper deals with a dichotomous method for the computer-aided search of nonlinear interference sources in complex electromagnetic environment. Compared to the one-signal method, the dichotomous method allows one to carry out a much more faster search (by factor of tens or more). An example of the search process and an estimation of the number of the required analysis cycles as well as description of the algorithm are given. The relation between identification and optimization problems is outlined.

INTRODUCTION

Computer-aided modeling of a radio electronic system is a very useful tool for electromagnetic compatibility/interference (EMC/EMI) analysis, in that it allows for the simulation of system behavior for a wide variety of initial conditions, excitations and system configurations in a rapid and inexpensive way [1]. A system can often reveal nonlinear behavior and nonlinear phenomena (intermodulation, cross-modulation, gain compression/expansion etc.) has profound effect on EMC/EMI in some cases [2]. Taking into account nonlinear interference at the system/subsystem design phase makes it possible to reduce the cost of its removal considerably. An identification of nonlinear interference sources is a very important task from the viewpoint of their removal. A computer-aided simulation tool can be used for such an identification in a very efficient way [3].

This article deals with a method of automatic identification of nonlinear interference sources, which is used in order to solve EMC/EMI problems in complex electromagnetic environment (for instance, in mobile communications environment, where there is a lot of emitters and receptors of EMI) . The specific character of this task is that a very accurate simulation of signals and interference levels is not required. However, in this case the analysis of complex systems must be carried out. Because of this, the simulation should be carried out at the system level.

A nonlinear modeling technique (so called ‘discrete technique’) for numerical EMC/EMI simulation at the system level has been proposed in [4,5]. This technique allows one to carry out rapid numerical EMC/EMI analysis of a complex system or subsystem (i.e. receiver, transmitter etc.) or a set of systems/subsystems in a wide frequency range taking into account nonlinear effects (including spurious responses of a receiver) and maintaining accurate spectra representation. Such an analysis is, for instance, a very important part of EMC/EMI modeling of a mobile communication system [6-8].

THE DISCRETE TECHNIQUE

The basis of the discrete technique [4,5] is a representation of the equivalent block diagram of a system as linear filters (LF) and memoryless nonlinear elements (MNE) connected in series (or in parallel). Thus a stage which employs a nonlinear element, for example, an amplifier, can be represented as a typical radio stage (see Figure 1), which employs the linear filter at the input, the memoryless nonlinear element and the linear filter at the output [2].

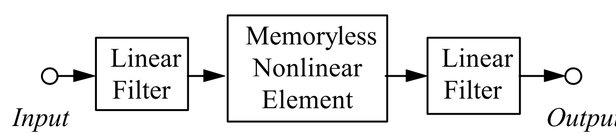

Figure 1. Representation of a typical radio frequency stage

This representation reflects characteristic peculiarities inherent to the construction of typical amplifying and converting stages. The utilization of the model with memoryless nonlinearity is not a significant limitation on the method for two reasons. First, non-zero memory effects can partially be factorized at the level of input or output filters, that is, this representation is equivalent with respect to the simulation of the "input-to-output" link. Second, the prediction of a signal spectrum at the system input taking into consideration EMC problems is, as a rule, not very accurate - the error can be as large as several dB or even tens of dB. It is an essential limitation on the simulation accuracy (the accuracy a signal at the system output can be predicted with). Thus great accuracy of system simulation is not necessarily required when the input signal is known with small accuracy. Therefore our viewpoint is that the utilization of the Volterra series for the analysis of nonlinear effects with respect to EMC problems [2] causes an essential increase of complexity without any essential increase in the analysis accuracy taken as a whole.

The process of signal passage through linear filters is simulated in the frequency domain using the complex transfer factor of the filter,

$$S_{out}(f_n) = S_{in}(f_n) \cdot K(f_n).,\qquad (1)$$

where $S_{out}(f_n)$- is the signal spectrum at the filter output, $S_{in}(f_n)$ - the signal spectrum at the filter input, $K(f_n)$ - is the complex transfer factor of the filter, f_n - are sample frequencies. It is necessary to have a sampled spectrum in order to do a calculation of this type. A spectrum sampling technique is given in [9]. The essential improvements in this technique, which adopt it for modern computers and allow one to increase accuracy, are given in [3]. The adaptive sampling technique can be used for this purpose too [10].

The process of signal passage through a nonlinear memoryless element is simulated in the time domain,

$$u_{out}(t_k) = \sum_{i=1}^{I} a_i u_{in}^i(t_k). \ ,\qquad (2)$$

where $u_{out}(t_k)$ - is the instantaneous value of the signal at the MNE output, $u_{in}(t_k)$ - is the same for the MNE input, t_k - are sample points in time, a_i - are coefficients of the high-order polynomial which describes the transfer characteristic of the nonlinear element; I - is order of the polynomial. The necessity of polynomial approximation of the nonlinear element transfer characteristic will be substantiated below.

The transition from the time domain to the frequency domain and vise versa is made with the use of the direct and inverse fast Fourier Transform (FFT). The direct FFT can be carried out by one of known methods [11] using the following ratio

$$S_n = \frac{1}{N}\sum_{k=0}^{N-1} u_k \cdot W^{nk} \ , \ W = e^{-j(2p/N)}\qquad (3)$$

where $S_n = S(f_n) = S(n\Delta f)$, $u_k = u(t_k) = u(k\Delta t)$; Df - frequency sample interval, Dt - time sample interval, N - number of samples. The inverse FFT is

$$u_k = \sum_{n=0}^{N-1} S_n \cdot W^{-nk}\qquad (4)$$

It is worth mentioning that the normalization given in (3) and (4) must be used during a nonlinear analysis. The normalization of other types which is often used in the literature will produce incorrect results.

The direct and inverse FFT vary only in the normalization and the exponent sign, which makes it possible to use the same algorithm in order to carry out the direct as well as the inverse FFT. It is necessary to make the corresponding data normalization and to arrange the data in the appropriate order before the FFT is carried out.

Let us note a number of peculiarities connected with the use of the FFT for nonlinear analysis.

1. The maximum frequency in the spectrum F_{max} , frequency sample interval Df , time sample interval Dt and the number of samples N are connected by the following ratios

$$\Delta t = \frac{1}{2 \cdot F_{\max}}, \ N = \frac{T}{\Delta t} = \frac{1}{\Delta t \cdot \Delta f}\qquad (5)$$

   where $T=1/Df$ - signal repetition period. The necessary number of samples in the frequency domain is actually equal to $N/2$, since samples with numbers arranged symmetrically with respect to $N/2$, are complex conjugate ones: $S_{N-n}=S_n*$ In the time domain, all N samples are independent.

2. Nonlinear transformation of the input signal causes its spectrum to expand I times (I - power of the polynomial which describes the amplitude characteristic of the nonlinear element); therefore, taking into account the cyclic character of the FFT in the frequency domain [11], the maximum allowable frequency in the input signal spectrum will be

$$F_{in,\max} = \frac{2 \cdot F_{\max}}{I+1} = \frac{1}{(I+1)\cdot \Delta t},\qquad (6)$$

   Thus the undistorted spectrum is obtained at the nonlinear element output within the interval $[0,F_{in,max}]$. Hence it is clear why the polynomial approximation (2) is to be used for the nonlinear element characteristic: otherwise the spectrum would expand infinitely, which would produce incorrect results. When the inverse FFT is calculated at the nonlinear element output the spectrum S_n has to be calculated only within the interval $[0,F_{in,max}]$, which allows one to reduce the calculation time. The ratios (5)-(6) make it possible to determine the number of samples (and hence the amount of computer memory) which is required in order to analyze a system if the maximum frequency at the input, frequency sample interval and the order of nonlinearity are specified.

3. The maximum possible range of amplitudes in the signal spectrum is determined by errors in the signal amplitude quantization in the time domain, that is by the accuracy of computer data presentation (for a floating-point number with "double" format this value is 280 dB). When simulating multistage systems, the quantization noise caused by the amplitude quantization is accumulated. This effect can be nullified by periodic "clearing" of the spectrum (that is, zeroing of the components whose level is lower than a certain threshold).

4. The utilization of geometrically spaced sample frequencies makes it possible to reduce the number of samples, or to reduce the frequency sample interval, or to increase the

order of simulated nonlinearity. However, it will slightly increase the simulation time.

Further improvement in the computational efficiency of the radio systems simulation can be achieved by means of a two-stage simulation scheme [4]. At the first stage, the radio system simulation correct to carrier frequencies (low frequency resolution) is carried out. All interference signals revealed at the first stage are sequentially analyzed at high frequency resolution (correct to modulating spectra) and with transformation to low frequencies.

A polynomial synthesis technique has been discussed in [12]. A detector can also be simulated by means of this technique [13]. Using the technique, a radio receiver can be simulated in a wide frequency range with very high frequency resolution (up to 10^6 - 10^7 sample frequencies) on a modern PC in dozens of minutes (a conventional circuit-level simulation would require several years for such an analysis).

IDENTIFICATION OF NONLINEAR INTERFERENCE SOURCES

Next we will consider the simulation of radio receivers (all obtained results can be easily applied to systems of other kinds too). A situation under analysis is shown in Figure 2. Interference signals S_1-S_N (separate spectral components of signals can also be used as S) affect the victim receiver Rx and cause nonlinear interference at its output.

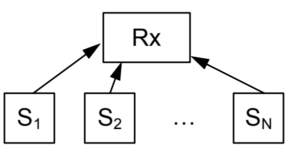

Figure 2. Situation under analysis. Interference signals S_1-S_N affect the victim receiver Rx and cause nonlinear interference at its output.

In the general case the problem of nonlinear interference sources identification is much more complex than the interference sources identification during linear analysis. The general approach to nonlinear interference sources identification may be formulated on the basis of the fact that a nonlinear interference disappears when at least one signal which takes part in its formation is excluded (is "turned off"). For example, a second-order intermodulation product is proportional to the product of amplitudes of signals which take part in its formation: $IMP_2 \sim U_1 \cdot U_2$. If $U_1=0$, then $IMP_2=0$ (the same for U_2). A similar principle is also true for the case of IMP of higher orders which may be formed by more than 2 signals and for the whole class of other nonlinear interference types (desensitization, cross modulation, local oscillator noise conversion, etc.).

This principle may be used as a basis for a number of identification methods which consist in repeated recalculation of the signal at the receiver output while one or several sources are excluded from the analysis. The simplest identification method is

(1) to carry out the calculation of the output signal when all the signals S_1-S_N are active ("turned on"),

(2) to exclude ("to turn off") the signal S_1 ,

(3) make the analysis (i.e. computation of the total signal at the receiver output) for the other signals (S_2 - S_N),

(4) check whether the interference disappeared. The interference amplitude A_{int} is an indicator of the disappearance:

$$A_{int} < a \times A_{int,0} \quad , \tag{7}$$

where $A_{int,0}$ - is interference level at the step 1 (when the signal S_1 was turned on), a - is a reduction in the interference level, which indicates its disappearance (a » *0.5 ... 0.1*). If the interference did not disappear then S_1 is not its source; otherwise it is its source.

(5) Then the procedure is repeated for the signals S_2 - S_N.

This method may be called the one-signal method. Its use is expedient when the signals number N is not large ($N<10$), since the nonlinear receiver analysis itself requires for a lot of time (this value may vary from several seconds up to several hours depending on the receiver complexity and a computer type.). The required number of analysis cycles is

$$n_A=N \tag{8}$$

This method cannot be used if there is a large number of signals. In this case it is necessary to use the dichotomous search method.

DICHOTOMOUS SEARCH METHOD

The essence of this method is as follows: a group of signals rather than each separate signal is turned off. If the exclusion of the group of signals does not cause the interference to disappear then this group of signals does not contain interference sources and can be discarded from the further consideration. If the interference does disappear then this group contains an interference source. In this case the group is to be divided into parts and these parts are to be analyzed with the use of the method described above. When the dichotomous method is used the group under analysis is divided into 2 equal parts at each step. This process is repeated until each group contains one signal whose exclusion makes it possible to determine whether or not this signal is an interference source. This method is schematically represented in Figure 3. In the case under consideration there are 8 signals (S_1 - S_8); the signals S_2 and S_5 are the interference sources. Each group of signals is divided into two parts at each step of the analysis. The parts whose exclusion does not cause the interference to disappear are discarded from the further search steps.

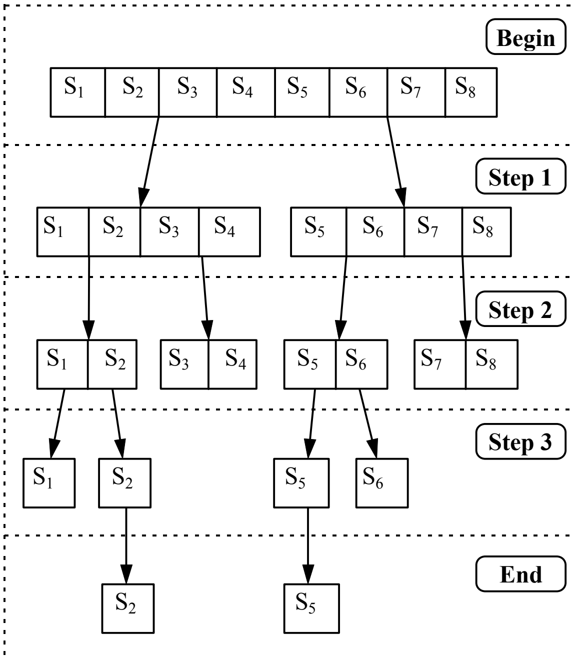

Figure 3. An example of the search of nonlinear interference sources by the dichotomous method. The total number of signals N=8. S_2 and S_5 are the sources of the nonlinear interference.

The number of analysis cycles required to identify an interference which has *k* sources is

$$n_A » 2k\, log_2(N) \tag{9}$$

The comparison of (8) with (9) shows that the dichotomous search method provides considerable advantage over the one-signal method when there is a large number of signals. Here is an example. For $k=2$ and $N=10^3$, the one-signal method requires for 1000 analysis cycles and the dichotomous method - for about 40 analysis cycles (for higher *N* values this difference is even more pronounced). If one analysis cycle takes 1 minute to carry out then the analysis with the use of the one- signal method will last for about 16 hours and the analysis with the use of the dichotomous method will last for about 40 minutes (this difference is similar to the difference between discrete Fourier transform and fast Fourier transform).

The search time can be significantly reduced if the signals are previously sorted in accordance with their amplitude and the group which contains the smaller signals is excluded from the analysis in the first turn, since large signals are the most probable nonlinear interference sources. It is expedient to take into consideration the intermodulation dynamic range of the receiver. It is also expedient to determine whether the signals fall into the RF preselector bandwidth (the signals which do not fall into the RF preselector bandwidth are excluded in the first turn).

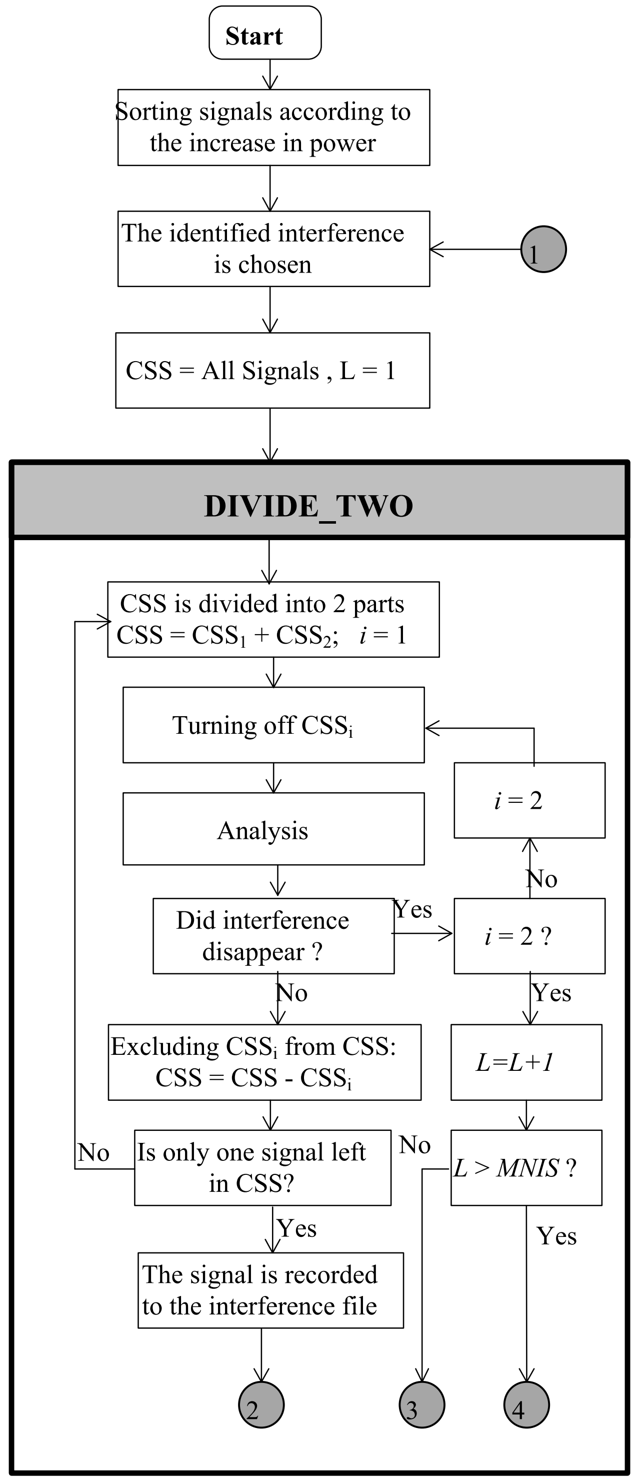

Figure 4. The Dichotomous Search Algorithm. CSS - current sets of signals, *MNIS* - the maximum number of interference signals, *L* - its current value, *i* - internal variable.

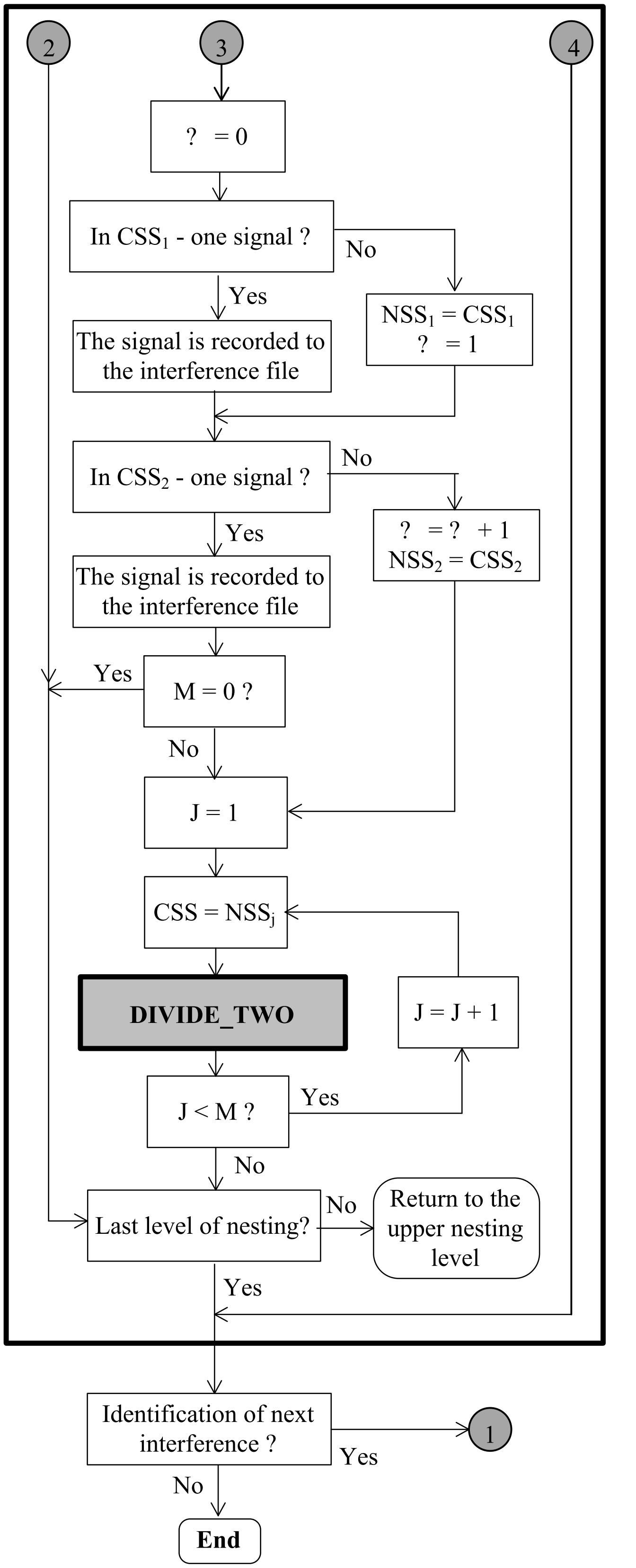

Figure 4. The Dichotomous Search Algorithm (continued). NSS - new sets of signals, M and J - internal variables.

DICHOTOMOUS SEARCH ALGORITHM

The dichotomous search algorithm based on the method given in the previous section are presented in Figure 4. Let's now consider the main steps of the algorithm. After the start, all signals are sorted according to the increase in power. Then the user chooses an interference to identify and the current set of signals is equated to all the signals. After that the procedure DIVIDE_TWO is called. The main function of this procedure is to divide the current signal set CSS into two parts (CSS_1 and CSS_2) in such a way that the first part contains smaller signals and the second one - lager signals, to «turn off» the first part, to conduct the analysis and to check whether the interference disappeared.

If it did not disappear then the turned-off part is excluded from the current signal set and the process of division is continued. If interference disappeared then the second part (CSS_2) is turned off (the first part remains to be turned on) and the analysis is repeated. If the interference did not disappear then the interference source is in the first part only and the process of division is continued for this part (the second part is excluded from further consideration). If interference disappeared, then the second part contains interference sources.

In this case the procedure DIVIDE_TWO executes a series of internal settings and checks whether the number of interference signals (L) exceeds the maximum admissible value (*MNIS*) which is set by the user. If it does not then the search procedure is continued (if it does then the process of the current interference sources search is stopped and the user can choose a next interference to search its sources; the limitation of the interference sources number is necessary in order to limit the time the search process requires and the number of sequential calls to the nested procedure DIVIDE_TWO).

If there is only one signal in each part then they are interference sources, and then the exit from the procedure is made. Otherwise the parts which contain more than one signal are divided into two parts and the above-mentioned operations are repeated (two new sets of signals are introduced and the procedure DIVIDE_TWO is called again). After the exit from the procedure DIVIDE_TWO of the uppermost level the user can choose a next interference for identifying. If it is not necessary, then the algorithm is completed. The interference signals have been saved to the interference file.

THE RELATION BETWEEN IDENTIFICATION AND OPTIMIZATION PROBLEMS

It should be pointed out that the problem of the interference source identification is similar to some optimization problems [14,15]. If we define the goal function F as a function of several signals (each interference has its own goal function)

$$F = F(S_{n1}, S_{n2}, \dots S_{nk}) \,, \qquad (10)$$

where k - is the number of interference sources, in such a way that this function is equal to 1 for the interference sources and to 0 for all other combinations of signals,

$$F = \begin{cases} 1 & \textit{if } S_{n1} \ldots S_{nk} \textit{ is the full set of interference sources} \\ 0 & \textit{otherwise} \end{cases} \tag{11}$$

then the identification problem will be completely similar to the problem of maximizing F which can be solved with the use of a number of well-known techniques [14,15], among which are the Fibonacci method, the golden section method as well as the dichotomous method.

CONCLUSIONS

The computer-aided method of nonlinear interference sources identification has been presented in this paper. This method can be applied for the identification of nonlinear interference (intermodulation, cross-modulation, desensitization etc.) sources in a complicated electromagnetic environment, when there is a lot of interference (for instance, in mobile communications, in co-site situations etc.) and when it's difficult to find out interference sources manually.

Further improvement in the computational efficiency of the identification technique can be achieved by use of methods known from optimization theory.

The technique proposed can also be used for the identification of linear interference sources. However, this is unsuitable because identification of this interference can be carried out at the stage of linear analysis, which requires less time.

REFERENCES

[1] F.M. Tesche, *Numerical Modeling for EMC*, Proc. of 12th Inter. Zurich Symp. On EMC, Zurich, Switzerland, Feb. 18-20, 1997, pp.269-274.

[2] D.D. Weiner, *Nonlinear Interference Effects in EMC*, Supplement to the Proc. of 10th Int. Zurich Symp. On EMC, Zurich, March 1993, pp.114-127.

[3] S.L.Loyka and V.I. Mordachev, *Mathematical models and algorithms of electromagnetic compatibility analysis and prediction software complex*, Technical report, vol.1, Belorussian State University of Informatics and Radioelectronics, Minsk, Belarus, 1997.

[4] V.I. Mordachev, *Express analysis of electromagnetic compatibility of radioelectronic equipment with use of discrete models of interference and fast Fourier transform*, Proc. of IX Inter. Wroclaw Symp. on EMC, Poland, Wroclaw, 1988, Part 2, pp.565-570.

[5] S.L. Loyka,. V.I Mordachev, *Computer-aided nonlinear simulation at the system level,* Proc. of 5th Inter. Confer. On EMC/EMI (INCEMIC'97), Hyderabad, India, Dec. 3-5, 1997, pp. 93-98.

[6] S.W.Chen, W. Panton and R. Gilmore, *Effects of Nonlinear Distortion on CDMA Communication Systems*, IEEE Trans. On MTT, vol. 44, No. 12, Dec. 1996, pp.2743-2750.

[7] B. Gallagher, *Estimating and Measuring C/I in a GSM Wireless Local Loop Receiver*, Microwave Journal, vol.40, No. 10, Oct. 1997, pp.70-83.

[8] J. Staudinger, *Applying the Quadrature Modeling Technique to Wireless Power Amplifiers*, Microwave Journal, vol.40, No. 11, Nov. 1997, pp.66-86.

[9] J.L. Bogdanor, R.A. Pearlman and M.D. Sieyel, *Intrasystem Electromagnetic Compatibility Analysis Program: Volume I - User's Manual Engineering Section,* Mc.Donnel Douglas Aircraft Corp., F30602-72-C-0277, Rome Air Development Center, Griffiss AFB NY, Dec. 1974.

[10] F.M. Tesche, A.W. Kalin, M. Nyffeler and B.R. Brandli, *Representation of Wide-Band Spectra Using An Adaptive, Nonuniform Sampling Scheme*, Proc. of 12th Inter. Zurich Symp. EMC, Zurich, Feb. 18-20, 1997.

[11] W.H. Press, B.P. Flanneay, S.A. Teukolsky, W.T. Vetterling, *Numerical Recipes in C*, Cambridge University Press, 1988.

[12] I.D Cheremisinov., S.L. Loyka,. V.I Mordachev, *Synthesis of the polynomial model of nonlinear elements based on intermodulation dynamic ranges*, Proc. of 3rd Inter. Confer. On Telecommunications in Modern Satellite, Cable and Broadcasting Services (TELSIKS'97), Oct. 8-10, Nis, Yugoslavia, 1997, pp. 519-522.

[13] S.L. Loyka, *Detector Simulation with the Use of the Discrete Technique*, 14th Inter. Wroclaw Symp. on EMC, Poland, Wroclaw, to be published.

[14] Yu.I. Degtjarev, *Optimization Methods*, Moscow, Sovetskoe Radio, 1980.(In Russian).

[15] M. Aoki, *Introduction to Optimization Techniques*, The Macmillan Company, New York, 1976.