18-2

MODELING OF NONLINEAR DISTORTION IN GaAs MESFETS

R. J. Gilmore (Student Member IEEE) and F. J. Rosenbaum (Fellow IEEE)

Central Microwave Co., 12180 Prichard Farm Rd., St. Louis, Missouri

Summary

A modified harmonic balance approach is applied to a GaAs FET model to simulate two-tone intermodulation distortion and multiple signal suppression in MESFET amplifiers and mixers. Experimental results of a single stage FET amplifier are presented.

Introduction

Digital communication system applications of FET power amplifiers place severe demands on the intermodulation distortion performance of the FETs. Amplifier design for specified IMD has been approached largely from an empirical point of view. In order to meet the challenging specifications, it is helpful to have a model capable of predicting the non-linear distortion behavior of the FETs. This paper presents an efficient approach to such modeling.

Several approaches exist for modeling distortion in a nonlinear circuit. The most obvious steady-state dynamic nonlinear circuit analysis method is numerical time-domain integration (1). One major drawback of this method is that time-constants associated with the circuit may differ by orders of magnitude, requiring integration over many periods of the microwave cycle. In the case of mixers, the IF is usually much less than the RF or LO frequency, necessitating integration over thousands of cycles to extract the IF response.

A second method of nonlinear circuit analysis is the harmonic balance frequency-domain simulation (2). Only the response of the nonlinear subnetwork is calculated in the time-domain, as the response of the linear embedding network is most easily found in the frequency domain. However, such approaches typically deal only with harmonically related inputs and outputs. The circuit inputs are assumed to have a single period T_0, or some multiple thereof. Consequently, the calculated circuit response is limited to frequencies of n/T_0, n=1,2,..k.

A more restrictive approach to modeling nonlinear distortion is to assume a power series expansion relating the device voltage and current, or to assign nonlinear characteristics to the elements of a linear device model (3). Such approaches enable analytic expressions for gain compression, AM to PM conversion, and intermodulation distortion to be obtained for simple nonlinearities. However, such approaches can be used only in the range for which the nonlinear extension to the linear model is valid. In FETs, for example, the effects of self-biasing, Schottky barrier breakdown, and heating, cannot easily be handled by such an approach.

This paper presents a truly nonlinear analysis of distortion in GaAs MESFETS, using an efficient modified harmonic-balance approach where two, non-harmonically related frequencies provide the excitation. The approach retains the efficiency of the harmonic balance method, allowing linear subnetworks of any complexity to be handled easily. It is applicable to large-signal nonlinear analysis of mixers and amplifiers in which two closely spaced frequency components are present.

The Modified Harmonic Balance Approach

A general circuit will contain one or more nonlinear elements embedded in linear circuitry. By deembedding as shown in Figure 1, two sets of equations relating voltage to current can be derived, one set for the linear network, the other for the nonlinear network. The overall effect of

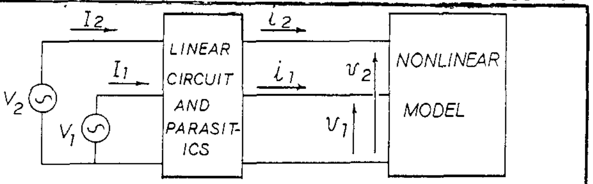

Figure 1
Circuit Partitioned into Linear-Nonlinear Networks

harmonic balance is to solve these two sets simultaneously. Because the nonlinear model is a time domain one, and the linear network is most efficiently represented in the frequency domain (by, for instance, its Z-parameters), frequency-to time- to frequency-conversions form the heart of the harmonic balance method.

In the standard harmonic balance approach, the input voltages and their derivatives are sampled uniformly at the Nyquist rate to obtain time-samples. These are input to the nonlinear model, which calculates the terminal currents at the same instants of time. A discrete Fourier transform (DFT) may then be used to extract frequency components of the current from these time samples, provided a sufficient number of samples are initially taken. (The frequency components

0149-645X/84/0000-0187 $01.00 © 1984 IEEE

are needed for subsequent input to the linear embedding network, which is represented in the frequency domain). Because all components of the current are the fundamental and higher harmonics of the same frequency ω_0, the DFT coefficients represent frequency components which are integer multiples of the fundamental frequency ω_0. A nonlinear optimization routine iterates until there is agreement between the circuit variables at the linear-nonlinear interface.

The difficulty with two nonharmonically-related inputs (as in two-tone amplifier measurements or in mixers), arises in the selection of the time-samples for the DFT. Because the difference frequency between the two input signals is $\Delta = \omega_2 - \omega_1$, the DFT fundamental frequency must now be Δ, in order to include the intermodulation products, and both input signals, as components. Since Δ is usually considerably less than either ω_1 or ω_2, a very large number N of time-domain samples must be taken at the Nyquist rate to include every component as a Fourier term. This is necessary because for an N-point sequence in time, the DFT yields only N Fourier coefficients, with the Fourier series expanded in harmonics of Δ.

However, since only a few frequency components are present, most of the Fourier coefficients between Δ and ω_1 are zero. The approach used here, which reduces the number of samples required by several orders of magnitude, is to sample at a much slower rate. This will cause aliasing. Then, by frequency translating both input signals successively by Δ and repeating the sample and DFT

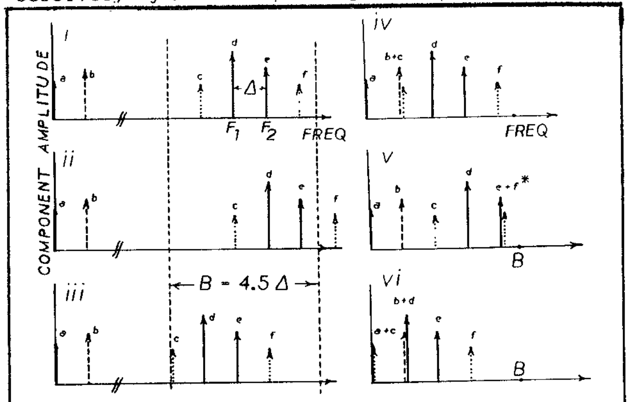

Figure 2

i) Typical FET output spectra for two-tone excitation. The solid lines are the two fundamental tones at f_1 and f_2 (d,e). For clarity, only the 3rd order products (c,f), beat frequency (b), and DC components (a), generated by the nonlinearity are shown.
ii) As in (i), but with excitations of the same magnitude at $f_1 + \Delta = f_2$, and at $f_2 + \Delta$.
iii) As in (i), but with excitations of the same magnitude at $f_1 - \Delta$, and at $f_2 - \Delta = f_1$.
iv) Result of the DFT applied to the time signal represented by the spectra in (i), but with only 9 time-samples taken, spaced $1/9\,\Delta$ in time. Note that aliasing has caused the lower intermodulation product to rollover onto the beat frequency (b+c)
v) Result of the DFT applied to (ii). Note the aliasing produces different frequency components.
vi) Result of the DFT applied to (iii).

process, the aliased signal components will fall into different frequency slots, but with very little change in magnitude. Now, when aliasing occurs in the DFT, a different amount of frequency overlap will occur for each frequency translation. Since the frequencies of the distortion products are known, the terms contributing to each component in the resultant (aliased) spectrum are also known. It is then possible to deembed all frequency components simply by taking linear combinations of the spectra produced by several frequency shifts (Figure 2).

As an example, for a cubic polynomial, which produces two third-order intermodulation products, a total of nine time samples spaced $1/9\,\Delta$ are taken. By frequency translating the two input signals at ω_1 and ω_2 twice, once by $-\Delta$, and again by Δ, and resampling, every frequency component generated by the third-order nonlinearity (including the second and third harmonics, intermodulation products, and beat frequencies) can be deembedded from the three resultant baseband spectra.

Application

Preliminary applications of the method to third order intermodulation in a single-stage GaAs FET amplifier are given in Fig. 3. (4). The powers in the fundamental and third-order product are plotted as a function of input drive level. Experimental results are shown for comparison.

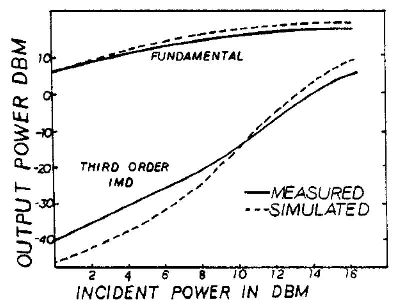

Figure 3

Comparison of measured and simulated output powers of both the fundamental and the third-order intermodulation product, from a single stage GaAs MESFET amplifier

Investigations are underway of multiple signal suppression in limiting amplifiers, where two signals of unequal amplitude are present. Using this method, gain suppression of a smaller signal in the presence of a larger signal can be investigated.

Acknowledgement The authors wish to recognize the work of Stacie Kawano for her pioneering efforts in exploring techniques enabling the distortion characteristics of FETS to be modelled.

References (1) C. W. Gear, Numerical Initial Value Problems in Ordinary Differential Equations. Prentice-Hall Inc., Englewood Cliffs, N.J. 1971
(2) M. S. Nakhla and J. Vlach,"A piecewise harmonic balance technique for determination of periodic response of nonlinear systems," Vol. CAS-23, No. 2, Feb. 1976.
(3) R. S. Tucker,"3rd order Intermodulation Distortion and Gain compression in GaAs Fets," Vol. MTT-27, No. 5, May 1979.
(4) S. Kawano,"Modelling of Third-Order Intermodulation Distortion in GaAs MESFETs," MSEE Thesis, Wash U., St. Louis, MO