AUTOMATED THIRD-ORDER DISTORTION MEASUREMENTS

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Abstract

A technique is described for the determination of third-order distortion produced in "linear" amplifiers by analyzing their complex transfer characteristic. A minicomputer is used for both control of the test equipment and analysis of the data.

Introduction

The complex transfer characteristic, consisting of a "linear" amplifier is sufficient information to determine the spurious responses that are related to third-order distortion. A simple technique for determining intermodulation and cross-modulation distortion in amplifiers, utilizing an automated transmission magnitude and phase indicator, is discussed.

Theory

An active device used for amplification is not ideally linear. Under large signal conditions the transfer characteristic would have to be represented by an infinite series.

$$y = \sum_{n=1}^{\infty} k_n(x) f^n(x) \qquad (1)$$

In general the transforms of each coefficient, K_n, will be complex quantities whose values can be determined by an analysis of the output waveforms or by using a Volterra series approach. Small signal operating conditions, i.e., operation in a small region about the bias point, can be represented by Eq. (1) when all the coefficients of the expansion are zero except K_1. The small signal output, y_{ss}, is then given by

$$y_{ss} = k_1(x) f(x) \qquad (2)$$

Eq. (2) is used as a linear approximation for small signal circuit analysis and design. An important region of operation is the one in which small signal operation is assumed even though the signal levels have increased enough to produce a slight amount of distortion. An amplifier of this type is usually considered "linear." To analyze this mode of operation, three terms of the infinite series in Eq. (1) are required.

$$y = k_1 f(x) + k_2 f(x)^2 + k_3 f(x)^3 \qquad (3)$$

This is the first-order approximation needed to extract information about the intermodulation distortion and is assumed to be valid for small deviations from linear operation. If the input signal consists of m sinusoidal signals, i.e.,

$$f(x) = \sum_{n=1}^{m} R_e \left[A_n e^{(2\pi f_n t + \emptyset_n)} \right] \qquad (4)$$

the output will consist of a series of sinusoidal terms which contribute to an output at the m fundamental frequencies, their second and third harmonics and a variety of mixing frequencies between the fundamentals and harmonics.

At any one of the frequencies, f_n, the ratio of the fundamental output amplitude at that frequency to the small signal fundamental output amplitude, is the gain saturation, g_c.

$$g_c(f_n) = 1 + \frac{3}{4} \frac{K_3(f_n)}{K_1(f_n)} \left(A_n^2 + 2 \sum_{p=1, p \neq n}^{m} A_p^2 \right) \qquad (5)$$

Since K_3 and K_1 are complex quantities, true gain compression will occur only if the relative phase of K_3 to K_1 is 180°. Note that the gain saturation is a complex quantity, i.e., it has both magnitude and phase. The gain saturation for a single signal is

$$g_c\big|_{1\ \text{tone}} = 1 + \frac{3}{4} \frac{K_3}{K_1} A_1^2 \qquad (6)$$

The gain saturation for two equal level signals that are close enough in frequency so that the K_3 and K_1 values are the same is

$$g_c\big|_{2\ \text{tone}} = 1 + \frac{9}{4} \frac{K_3}{K_1} A_1^2 \qquad (7)$$

$$g_c\big|_{2\ \text{tone}} = 3 g_c\big|_{1\ \text{tone}} - 2 \qquad (8)$$

The third term of Eq. (3) indicates there are outputs at 2m(m-1) frequencies with output frequencies at $2f_n \pm f_p$, where f_n and f_p are any two of the m input frequencies. These outputs are commonly referred to as intermodulation distortion. The intermodulation distortion ratio, imr, is defined as the ratio of the amplitude of the intermodulation distortion at $2f_n - f_p$, to the amplitude of its respective fundamental output at f_n.

$$\text{imr}(2f_n - f_p) = \frac{3}{4} \frac{K_3(2f_n - f_p)}{K_1(f_n)} A_n A_p \qquad (9)$$

Intermodulation distortion for two equal level signals is related to gain saturation of each by

$$\text{imr}(2f_n - f_p) = \frac{1}{3} \frac{K_3(2f_n - f_p)}{K_3(f_n)} \left[g_c(f_n)\big|_{2\ \text{tone}} - 1 \right] \qquad (10)$$

It may also be related to the single signal gain saturation. If each of the two signals has a power level equal to a single signal that produces a gain saturation of $g_c\big|_{1\ \text{tone}}$, then

$$\mathrm{imr}(2f_n - f_p) = \frac{K_3(2f_n - f_p)}{K_3(f_n)} \left[g_c(f_n)\big|_{1\ \mathrm{tone}} - 1 \right] \quad (11)$$

In the case of true gain compression, i.e., when the relative phase of K_1 and K_3 is 180^o, the two equal tone intermodulation distortion ratio is -19.3 dB when each of the two signals are at the same output level which produces a gain compression of 1 dB for a single tone. This assumes that the signals are close enough in frequency so that $|K_3(2f_n - f_p)| = |K_3(f_n)|$. Note, however, that the intermodulation distortion ratio could also be -19.3 dB at some specified output level without any apparent gain compression or expansion. This occurs for the case of pure phase distortion. Under these conditions, the magnitude of the single tone gain saturation is unity but its phase has changed ±6.2 from its small signal value.

Other forms of third-order distortion such as triple-beat distortion, cross-modulation distortion (which is a special case of triple-beat), modulation distortion (which is a special case of intermodulation distortion), and AM to PM conversion can also be related to the gain compression in a similar manner.

Automated Testing

Measurement of the transfer characteristic may be accomplished by utilizing any instrumentation capable of measuring transfer amplitude and phase such as a vector voltmeter or network analyzer. The system used for measurements is shown in Figure 1. The minicomputer used was an HP2100S operating as part of a multitasking real-time executive (RTE) disk operating system. This system provided all hardware interfacing, i.e., A/D, D/A, multiplexor, relay control, in addition to a real-time clock. The flow chart is shown in Figure 2.

Since the measurement of small changes in amplitude and phase are required for the precise determination of distortion, several measures have been taken to guarantee the accuracy of the readings. After the initial setting of frequency and signal level, the DVM samples the error voltage of the phase-locked loop. If the vector instrumentation is not locked, the RTE sets up a delay and again checks for phase-lock. A non-phase-locked condition, after the maximum delay has been achieved, results in a change in frequency of .1% or any other predetermined value and the delay cycle is repeated. If phase-locking is not achieved when the frequency change limit has been reached, a command to adjust the loop control manually is issued. This procedure has increased the automatic locking range of the vector voltmeter from approximately one octave to over three octaves.

When a locked condition is achieved, the DVM scans amplitude, phase and locked voltage outputs repeatedly. It has been found that twenty cycles is sufficient for accurate characterization of the amplifier under test. During any of these cycles, if the phase-lock voltage indicates a locking error, the amplitude and phase voltages are disregarded as data points. The resulting data points are then processed by calculating the standard deviation and any points outside these limits are also disregarded. The remaining data is then averaged and stored.

The signal level is changed and the entire cycle is repeated.

Experimental Results

Representative experimental results are shown in Figures 3 through 5. The data points obtained by the automated test set are shown as dots. All drawn lines are computer generated. For comparison purposes data points manually obtained with a spectrum analyzer are shown encircled.

Figure 3 is an example of an amplifier with practically no phase distortion at its 1 dB compression point. Note that the intercept point is approximately 9 dB above the single tone 1 dB compression point. This is very close to the theoretical value of 9.65 dB for pure gain compression.

Figure 4 indicates the results for an amplifier which exhibits relatively large amounts of phase distortion before any noticeable gain compression occurs. The phase distortion is greater than 15^o at the 1 dB gain compression point. Notice the ambiguity that may occur in the intercept point if the intermodulation distortion data used is above an IMR of 35 dB. In this case, the IMR is much greater than 19.3 dB at the 1 dB compression point.

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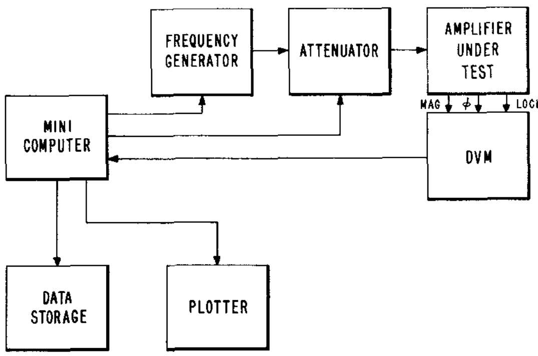

Figure 1 Computer Controlled System

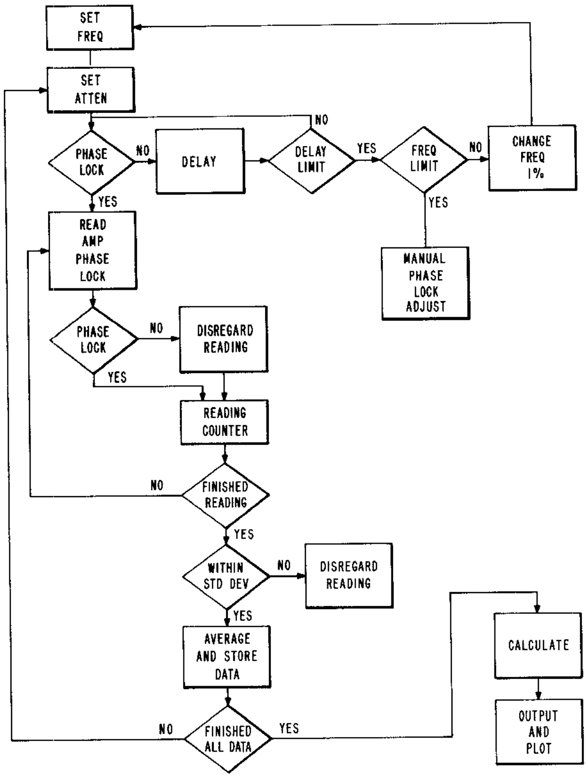

Figure 2 Flow Chart

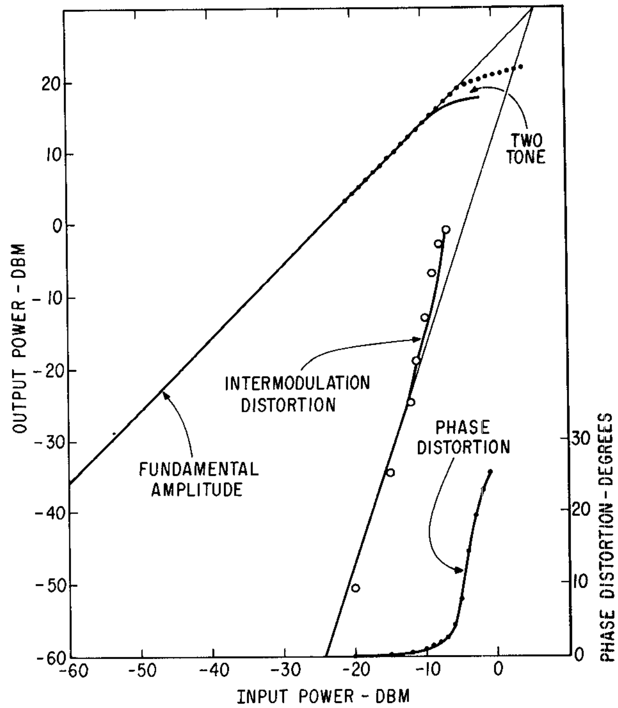

Figure 4 Phase Distortion

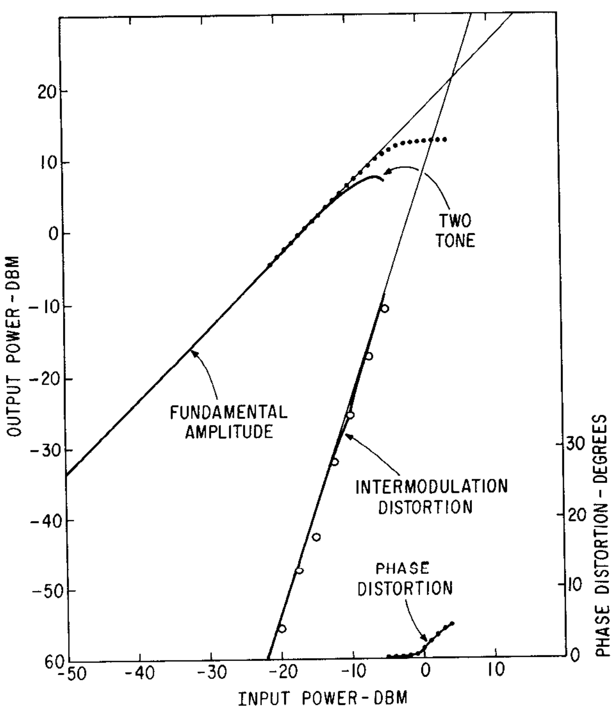

Figure 3 Gain Compression

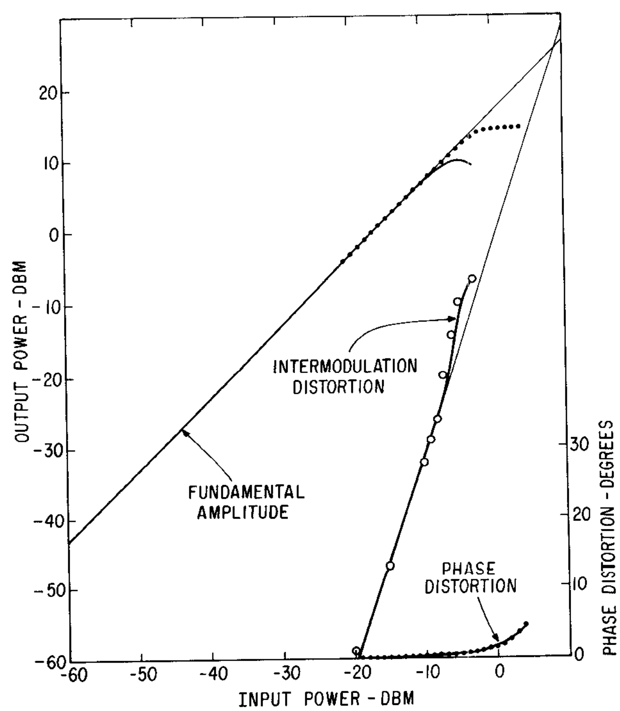

Figure 5 Gain and Phase Distortion