SIGNAL MEASUREMENT, ANALYSIS, and TESTING

edited by

Jerry C. Whitaker

SIGNAL MEASUREMENT, ANALYSIS, and TESTING

edited by
Jerry C. Whitaker

CRC Press
Boca Raton London New York Washington, D.C.

Library of Congress Cataloging-in-Publication Data

Signal measurement, analysis, and testing / Jerry C. Whitaker, editor.
p. cm.
Includes bibliographical references.
ISBN 0-8493-0048-7 (alk. paper)
1. Signal processing—Digital techniques. 2. System analysis.
3. Fourier analysis. 4. Phase distortion (Electronics)
I. Whitaker, Jerry C.
TK5102.9.S537 1999
621.382'2—dc21 99-44876
CIP

The material in this book was taken from The Electronics Handbook, edited by Jerry D. Whitaker.

International Standard Book Number 0-8493-0048-7
Library of Congress Card Number 99-44876
Printed in the United States of America 1 2 3 4 5 6 7 8 9 0
Printed on acid-free paper

Preface

Signal Measurement, Analysis, and Testing is intended for engineers and technicians involved in the design, production, installation, operation, and maintenance of electronic devices and systems. This publication covers a broad range of technologies with emphasis on practical applications. In general, the level of detail provided is limited to that necessary to design electronic systems based on the interconnection of operational elements and devices. References are provided throughout the book to direct readers to more detailed information on important subjects.

The purpose of this book is to provide in a single volume a comprehensive reference for the practicing engineer in industry, government, and academia. It deals with the prominent aspects of this process for any electronic design, including a thorough treatment of computer-aided analysis emphasizing MATLAB®. Most modern signal analyses make use of computer-aided analysis in one form or other, including peak detection, time base measurement, Fourier analysis, and display. While most sections concentrate on distortion mechanisms and analysis, other sections present extensive tables and data of properties of materials, frequency bands and assignments, international standards and constants, conversion factors, general mathematics, and abbreviations of communications terms.

Contributors

Samuel O. Agbo
California Polytechnic University
San Luis Obispo, California

W.F. Ames
Georgia Institute of Technology
Atlanta, Georgia

George Cain
Georgia Institute of Technology
Atlanta, Georgia

Robert D. Greenberg
Federal Communications
Commision
Washington, DC

Jerry C. Hamann
University of Wyoming
Laramie, Wyoming

John W. Pierre
University of Wyoming
Laramie, Wyoming

James F. Shackelford
University of California
Davis, California

Jerry C. Whitaker
Technical Press
Beaverton, Oregon

Rodger E. Ziemer
University of Colorado
Colorado Springs, Colorado

Contents

I

Signal Measurement, Analysis, and Testing

1

Introduction

Rodger E. Ziemer
University of Colorado, Colorado Springs

THE CHARACTERIZATION, analysis, trouble shooting, and repair of modern complex electronic systems requires sophisticated techniques. This section addresses these aspects of the electronics engineer's profession. Different techniques are required for different regimes of operation. A convenient categorization for this purpose is audio, video, and radio frequency. In addition, it is convenient to subdivide further into analog and digital.

Computers have become pervasive throughout society, and the electronics engineer perhaps finds computer-based analysis more important than any other branch of engineering. Indeed, most modern signal analysis instruments now make use of computer-aided signal analysis in one form or other, including peak detection, time base measurement, Fourier analysis, and display.

This section consists of six major chapters. Four deal with instrumentation aspects of signal measurement, analysis, and testing. Two deal with the mathematics of signal analysis.

Chapter 130 is concerned with audio distortion mechanisms and analysis, mainly signal level, phase, and frequency, or combinations of these, such as signal-to-noise ratio.

Chapter 131 deals with video signal distortion mechanisms and analysis. Because of the unique characteristics of video display systems, special considerations are required for characterization of distortion in them. For example, number of lines scanned per second is an important consideration in this area.

Chapter 132 addresses radio frequency distortion mechanisms and analysis. The considerations are similar to those for audio distortion mechanisms and analysis, except the frequency ranges are higher and are bandpass.

Chapter 133 considers digital test equipment and measurement systems. A universal instrument in this category is the logic analyzer, and this chapter gives an easily understood overview of this important instrument and its capabilities.

The important area of Fourier waveform analysis is dealt with in Chapter 134. In the not too distant past, Fourier analysis was strictly an analytical tool. The discovery of the fast Fourier transform (FFT) in the late 1960s and the development of fast hardware to perform the FFT in the 1980s have made Fourier analysis a mainstay of signal analysis instrumentation, and it is difficult to tell an oscilloscope (with digital signal processing capability) from a spectrum analyzer with FFT-based spectral analysis capability.

Many software tools are now available for computer-based signal analysis, which is the subject of Chapter 135. This chapter concentrates on only one general purpose analysis tool called MATLAB®, which is a vector-based high-level programming language. It has many capabilities including general number crunching, signal analysis, graphical interface, and symbolic manipulation.

This section, written by several experts in the disciplines represented, provides important information for any electronic design. Whether a component in an overall system or a stand-alone box, distortion mechanisms and their characterization are important in the design of high-quality instrumentation.

0-8493-0048-1/00/$0.00+$.50

2

Audio Frequency Distortion Mechanisms and Analysis

Jerry C. Whitaker
Editor-in-Chief

2.1 Introduction

Most measurements in the audio field involve characterizing fundamental parameters.[1] These include signal level, phase, and frequency. Most other tests consist of measuring these fundamental parameters and displaying the results in combination by using some convenient format. For example, signal-to-noise ratio (SNR) consists of a pair of level measurements made under different conditions expressed as a logarithmic, or decibel, ratio. When characterizing a device, it is common to view it as a box with input terminals and output terminals. In normal use a signal is applied to the input and the signal, modified in some way, appears at the output. Instruments are necessary to quantify these unintentional changes to the signal. Some measurements are *one-port* tests, such as impedance or noise level, and are not concerned with input/output signals, only with one or the other.

Purpose of Audio Measurements

Measurements are made on audio circuits and equipment to check performance under specified conditions and to assess suitability for use in a particular application. The measurements may be used to verify specified system performance or as a way of comparing several pieces of equipment for use in a system. Measurements may also be used to identify components in need of adjustment or repair. Whatever the application, audio measurements are a key part of audio engineering.

Many parameters are important in audio devices and merit attention in the measurement process. Some common audio measurements are frequency response, gain or loss, harmonic distortion, intermodulation distortion, noise level, phase response, and transient response.

[1]Portions of this chapter were adapted from: Benson, K.B. and Whitaker, J.C. 1990. *Television and Audio Handbook for Technicians and Engineers*. McGraw–Hill, New York. Used with permission.

0-8493-0048-1/00/$0.00+$.50

Measurement of level is fundamental to most audio specifications. Level can be measured either in absolute terms or in relative terms. Power output is an example of an absolute level measurement; it does not require any reference. SNR and gain or loss are examples of relative measurements; the result is expressed as a ratio of two measurements. Although it may not appear so at first, frequency response is also a relative measurement. It expresses the gain of the device under test as a function of frequency, with the midband gain, typically, as a reference.

Distortion measurements are a way of quantifying the amount of unwanted components added to a signal by a piece of equipment. The most common technique is total harmonic distortion (THD), but others are often used. Distortion measurements express the amount of unwanted signal components relative to the desired signal, usually as a percentage or decibel value. This is another example of multiple level measurements that are combined to give a new measurement figure.

2.2 Level Measurements

The simplest definition of a level measurement is the alternating current (AC) amplitude at a particular place in the audio system. In contrast to direct current measurements, however, there are many ways of specifying AC voltage. The most common methods are average, root-mean-square (RMS), and peak response. Strictly speaking, the term *level* refers to a logarithmic, or decibel, measurement. However, common parlance employs the term for an AC amplitude measurement, and that convention will be followed in this chapter.

Root-Mean-Square Measurements

The RMS technique measures the effective power of the AC signal. It specifies the value of the DC equivalent that would dissipate the same power if either were applied to a load resistor. This process is illustrated in Fig. 2.1 for voltage measurements. The input signal is squared, and the average value is found. This is equivalent to finding the average power. The square root of this value is taken to translate the signal from a power value back to a voltage. For the case of a sine wave the RMS value is 0.707 of its maximum value.

Consider the case when the signal is not a sine wave, but rather a sine wave and several of its harmonics. If the RMS amplitude of each harmonic is measured individually and added, the resulting value will be the same as an RMS measurement on the signals together. Because RMS voltages cannot be added directly, it is necessary to perform an RMS addition, as follows:

$$V_{\mathrm{rms}} = \sqrt{V_{\mathrm{rms1}}^2 + V_{\mathrm{rms2}}^2 + V_{\mathrm{rms3}}^2 + V_{\mathrm{rms}n}^2}$$

Note that the result is not dependent on the phase relationship of the signal and its harmonics. The RMS value is determined completely by the amplitude of the components. This mathematical predictability is powerful in practical applications of level measurement, enabling measurements made at different places in a system to be correlated. It is also important in correlating measurements with theoretical calculations.

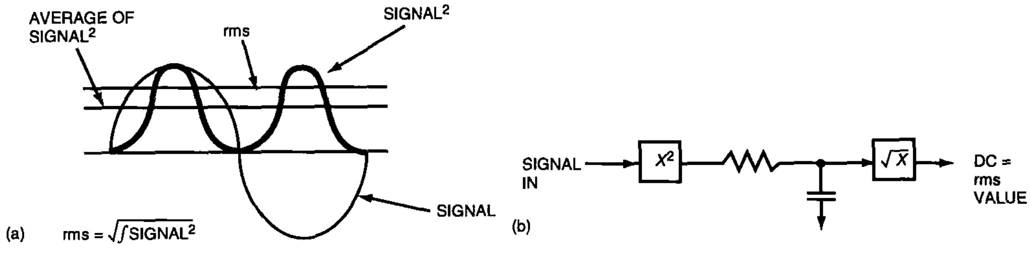

FIGURE 2.1 Root-mean-square (RMS) voltage measurements: (a) the relationship of RMS and average values, (b) RMS measurement circuit.

WAVEFORM	rms	Avg.	rms avg.	CREST FACTOR
SINE WAVE	$\frac{V_m}{\sqrt{2}}$ 0.707 V_m	$\frac{2}{\pi} V_m$ 0.637 V_m	$\left\lvert\frac{\pi}{2\sqrt{2}}\right\rvert = 1.111$	$\sqrt{2} = 1.414$
SYMMETRICAL SQUARE WAVE OR DC	V_m	V_m	1	1
TRIANGULAR WAVE OR SAWTOOTH	$\frac{V_m}{\sqrt{3}}$	$\frac{V_m}{2}$	$\frac{2}{\sqrt{3}} = 1.155$	$\sqrt{3} = 1.732$

FIGURE 2.2 Comparison of RMS and average voltage characteristics.

Average-Response Measurements

Average-responding voltmeters were common in audio work some years ago principally because of their low cost. Such devices measure AC voltage by rectifying and filtering the resulting waveform to its average value, which can then be read on a standard DC voltmeter. The average value of a sine wave is 0.637 of its maximum amplitude. Average-responding meters are usually calibrated to read the same as an RMS meter for the case of a single-sine-wave signal. This results in the measurement being scaled by a constant K of 0.707/0.637, or 1.11. Meters of this type are called average-responding, RMS calibrated. For signals other than sine waves, the response will be different and difficult to predict. If multiple sine waves are applied, the reading will depend on the phase shift between the components and will no longer match the RMS measurement. A comparison of RMS and average-response measurements is made in Fig. 2.2 for various waveforms. If the average readings are adjusted as described previously to make the average and RMS values equal for a sine wave, all of the numbers in the *average* column should be increased by 11.1%, whereas the *RMS-average* numbers should be reduced by 11.1%.

Peak-Response Measurements

Peak-responding meters measure the maximum value that the AC signal reaches as a function of time. (See Fig. 2.3.) The signal is full-wave rectified to find its absolute value and then passed through a diode to a storage capacitor. When the absolute value of the voltage rises above the value stored on the capacitor, the diode will conduct and increase the stored voltage. When the voltage decreases, the capacitor will maintain the old value. Some means for discharging the capacitor is required to allow measuring a new peak value. In a true peak detector, this is accomplished by a switch. Practical peak detectors usually include a large resistor to discharge the capacitor gradually after the user has had a chance to read the meter.

The ratio of the true peak to the RMS value is called the **crest factor**. For any signal but an ideal square wave the crest factor will be greater than 1, as illustrated in Fig. 2.4. As the measured signal become more peaked, the crest factor will increase.

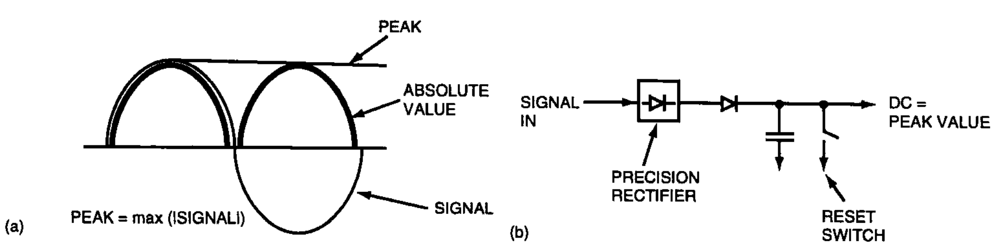

FIGURE 2.3 Peak voltage measurements: (a) illustration of peak detection, (b) peak measurement circuit.

By introducing a controlled charge and discharge time, a *quasipeak* detector is achieved. The charge and discharge times may be selected, for example, to simulate the ear's sensitivity to impulsive peaks. International standards define these response times and set requirements for reading accuracy on pulses and sine wave bursts of various durations. The gain of a quasipeak detector is normally calibrated so that it reads the same as an RMS detector for sine waves.

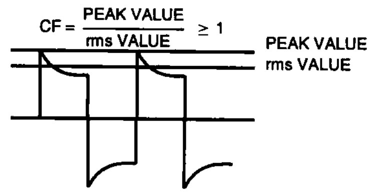

FIGURE 2.4 Illustration of the crest factor in voltage measurements.

Another method of specifying signal amplitude is the **peak-equivalent sine**, which is the RMS level of a sine wave having the same peak-to-peak amplitude as the signal under consideration. This is the peak value of the waveform scaled by the correction factor 1.414, corresponding to the peak-to-RMS ratio of a sine wave. This is useful when specifying test levels of waveforms in distortion measurements.

Decibel Measurements

Measurements in audio work are usually expressed in decibels. Audio signals span a wide range of level. The sound pressure of a live concert band performance may be one million times that of rustling leaves. This range is too wide to be accommodated on a linear scale. The logarithmic scale of the decibel compresses this wide range down to a more easily handled range. Order-of-magnitude changes result in equal increments on a decibel scale. Furthermore, the human ear perceives changes in amplitude on a logarithmic basis, making measurements with the decibel scale reflect audibility more accurately.

A decibel may be defined as the logarithmic ratio of two power measurements or as the logarithmic ratio of two voltages:

$$\text{dB} = 20\log\frac{E_1}{E_2} \quad \text{for voltage measurements}$$

$$\text{dB} = 10\log\frac{P_1}{P_2} \quad \text{for power measurements}$$

There is no difference between decibel values from power measurements and decibel values from voltage measurements if the impedances are equal. In both equations the denominator variable is usually a stated reference. A doubling of voltage will yield a value of 6.02 dB, whereas a doubling of power will yield 3.01 dB. This is true because doubling voltage results in a factor-of-four increase in power.

Audio engineers often express the decibel value of a signal relative to some standard reference instead of another signal. The reference for decibel measurements may be predefined as a power level, as in decibels referenced to 1 mW (dBm), or it may be a voltage reference, as in decibels referenced to 1 V (dBV). When measuring dBm or any power-based decibel value, the reference impedance must be specified or understood. Often it is desirable to specify levels in terms of a reference transmission level somewhere in the system under test. These measurements are designated dBr, where the reference point or level must be separately conveyed.

2.3 Noise Measurement

Noise measurements are simply specialized level measurements. It has long been recognized that the ear's sensitivity varies with frequency, especially at low levels. This effect was studied in detail by Fletcher and Munson and later by Robinson and Dadson. The Fletcher–Munson hearing-sensitivity curve for the threshold of hearing and above is given in Fig. 2.5. The ear is most sensitive in the region of 2–4 kHz, with rolloffs above and below these frequencies. To predict how loud something will sound it is necessary to use a filter that duplicates this nonflat behavior electrically. The filter *weights* the signal level on the basis of frequency, thus earning the name *weighting filter*. Various efforts have been made to do this, resulting in several standards for noise measurement.

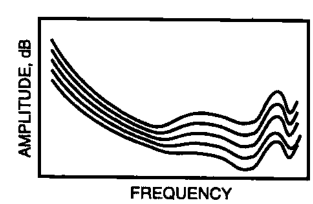

FIGURE 2.5 The Fletcher–Munson curves of hearing sensitivity vs frequency.

Some of the common weighting filters are shown overlaid on the hearing threshold curve in Fig. 2.6.

The most common filter used in the United States for weighted noise measurements is the A-weighting curve. An average-responding meter is often used for A-weighted noise measurements, although RMS meters are also used for this application.

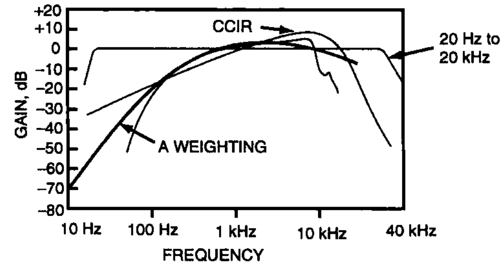

FIGURE 2.6 Response characteristics of several common weighting filters for audio measurements.

European audio equipment is usually specified with a CCIR filter and a quasipeak detector. The CCIR curve is significantly more peaked than the A curve and has a sharper rolloff at high frequencies. The CCIR quasipeak standard was developed to quantify the noise in telephone systems. The quasipeak detector more accurately represents the ear's sensitivity to impulsive sounds. When used with the CCIR filter curve, it is supposed to correlate better with the subjective level of the noise than A-weighted average-response measurements do.

Some audio equipment manufacturers specify noise with a 20 Hz–20 kHz bandwidth filter and an RMS-responding meter. This is done to specify noise over the audio band without regard to the ear's differing sensitivity with frequency. The International Electrotechnical Commission (IEC) defies the audio band as all frequencies between 22.4 Hz and 22.4 kHz. Measurements over such a bandwidth are referred to under IEC standards as unweighted.

2.4 Phase Measurement

Phase in an audio system is typically measured and recorded as a function of frequency over the audio range. For most audio devices phase and amplitude responses are closely coupled. Any change in amplitude that varies with frequency will produce a corresponding phase shift. A fixed time delay will introduce a phase shift that is a linear function of frequency. This time delay can introduce large values of phase shift at high frequencies that are of no significance in practical applications. The time delay will not distort the waveshape of complex signals and will not be audible. There can be problems, however, with time delay when the delayed signal is used in conjunction with an undelayed signal.

When dealing with complex signals, the meaning of phase can become unclear. Viewing the signal as the sum of its components according to Fourier theory, we find a different value of phase shift at each frequency. With a different phase value on each component, the question is raised as to which one should be used as the reference. If the signal is periodic and the waveshape is unchanged passing through the device under test, then a phase value may still be defined. This may be done by using the shift of the zero crossings as a fraction of the waveform period. If there is differential phase shift with frequency, however, the waveshape will be changed. It is then not possible to define any phase-shift value, and phase must be expressed as a function of frequency.

Group delay is another useful expression of the phase characteristics of an audio device. Group delay is the slope of the phase response. It expresses the relative delay of the spectral components of a complex waveform. If the group delay is flat, all components will arrive together. A peak or rise in the group delay indicates that those components will arrive later by the amount of the peak or rise. Group delay ϕ is computed by taking the derivative of the phase response vs frequency:

$$\phi = \frac{-(\alpha_{f2} - \alpha_{f1})}{f_2 - f_1}$$

where α_{f1} is the phase at f_1 and α_{f2} phase at f_2. This requires that phase be measured over a range of frequencies to yield a curve that can be differentiated. It also requires that the phase measurements be performed at frequencies close enough together to provide a smooth and accurate derivative.

2.5 Nonlinear Distortion Mechanisms

Distortion is a measure of signal impurity. It is usually expressed as a percentage or decibel ratio of the undesired components to the desired components of a signal. There are several methods of measuring distortion, the most common being harmonic distortion and several types of intermodulation distortion.

Harmonic Distortion

The transfer characteristic of a typical amplifier is shown in Fig. 2.7. The transfer characteristic represents the output voltage at any point in the signal waveform for a given input voltage; ideally this is a straight line. The output waveform is the projection of the input sine wave on the device transfer characteristic. A change in the input produces a proportional change in the output. Because the actual transfer characteristic is nonlinear, a distorted version of the input waveshape appears at the output.

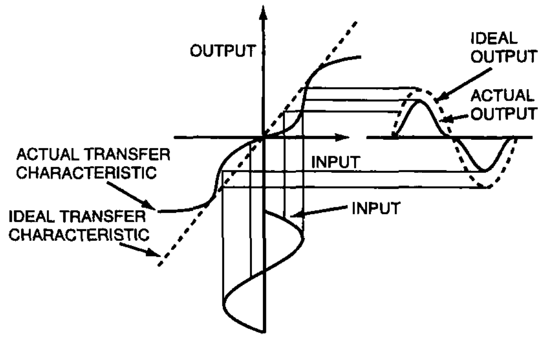

FIGURE 2.7 Illustration of total harmonic distortion (THD) measurement of an amplifier transfer characteristic.

Harmonic distortion measurements excite the device under test with a sine wave and measure the spectrum of the output. Because of the nonlinearity of the transfer characteristic, the output is not sinusoidal. By using Fourier series, it can be shown that the output waveform consists of the original input sine wave plus sine waves at integer multiples (harmonics) of the input frequency. The spectrum of the distorted signal is shown in Fig. 2.8 for a 1-kHz input, and output signals consisting of 1, 2, 3 kHz, etc. The harmonic amplitudes are proportional to the amount of distortion in the device under test. The percentage harmonic distortion is the RMS sum of the harmonic amplitudes divided by the RMS amplitude of the fundamental.

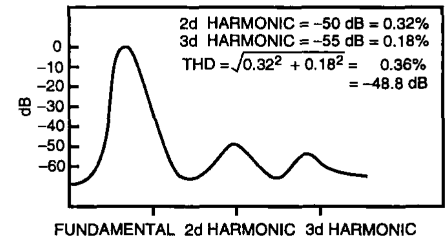

FIGURE 2.8 Example of reading THD from a spectrum analyzer.

Harmonic distortion may also be measured with a spectrum analyzer. As shown in Fig. 2.8, the fundamental amplitude is adjusted to the 0-dB mark on the display. The amplitudes of the harmonics are then read and converted to linear scale. The RMS sum of these values is taken, which represents the THD. This procedure is time consuming and can be difficult for an unskilled operator.

Notch Filter Analyzer

A simpler approach to the measurement of harmonic distortion can be found in the notch-filter distortion analyzer. This device, commonly referred to as simply a distortion analyzer, removes the fundamental of the signal to be investigated and measures the remainder. A block diagram of such a unit is shown in Fig. 2.9. The fundamental is removed with a notch filter, and the output is measured with an AC voltmeter. Because distortion is normally presented as a percentage of the fundamental signal, this level must be measured or set equal to a predetermined reference value. Additional circuitry (not shown) is required to set the level to the reference

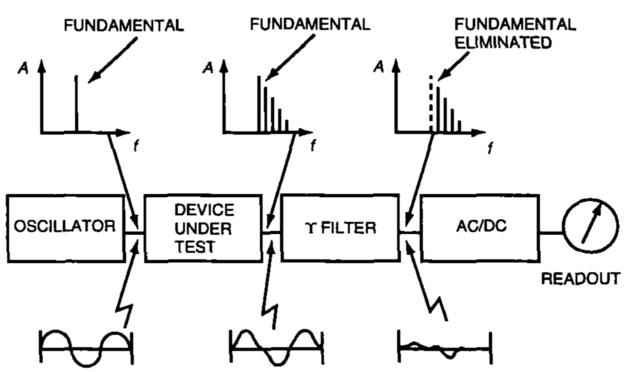

FIGURE 2.9 Simplified block diagram of a harmonic distortion analyzer.

value for calibrated measurements. Some analyzers use a series of step attenuators and a variable control for setting the input level to the reference value. More sophisticated units eliminate the variable control by using an electronic gain control. Others employ a second AC-to-DC converter to measure the input level and compute the percentage by using a microprocessor. Completely automatic units also provide autoranging logic to set the attenuators and ranges, and tabular or graphic display of the results.

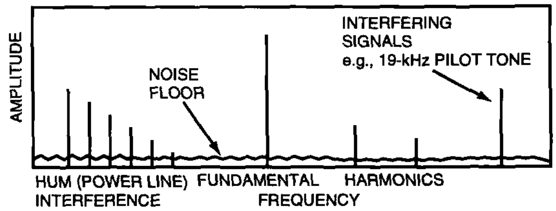

FIGURE 2.10 Example of interference sources in distortion and noise measurements.

The correct method of representing percentage distortion is to express the level of the harmonics as a fraction of the fundamental level. However, most commercial distortion analyzers use the total signal level as the reference voltage. For small amounts of distortion these two quantities are equivalent. At large values of distortion the total signal level will be greater than the fundamental level. This makes distortion measurements on these units lower than the actual value. The errors are not significant until about 20% measured distortion.

Because of the notch-filter response, any signal other than the fundamental will influence the results, not just harmonics. Some of these interfering signals are illustrated in Fig. 2.10. Any practical signal contains some noise, and the distortion analyzer will include these in the reading. Because of these added components, the correct term for this measurement is *total harmonic distortion and noise* (THD+N). Although this fact does limit the reading of very low-THD levels, it is not necessarily bad. Indeed, it can be argued that the ear hears all components present in the signal, not just the harmonics.

Additional filters are included on most distortion analyzers to reduce unwanted hum and noise. These usually consist of one or more high-pass filters (400 Hz is almost universal) and several low-pass filters. Common low-pass filter frequencies are 22.4, 30, and 80 kHz. A selection of filters eases the tradeoff between limiting bandwidth to reduce noise and the reduction in reading accuracy that results from removing desired components of the signal. When used in conjunction with a good differential input stage on the analyzer, these filters can solve most noise problems.

Intermodulation Distortion

Many methods have been devised to measure the intermodulation (IM) of two or more signals passing through a device simultaneously. The most common of these is **SMPTE IM**, named after the Society of Motion Picture and Television Engineers, which first standardized its use. IM measurements according to the SMPTE method have been in use since the 1930s. The test signal is a low-frequency tone (usually 60 Hz) and a high-frequency tone (usually 7 kHz) mixed in a 4:1 amplitude ratio. Other amplitude ratios and frequencies are used occasionally. The signal is applied to the device under test, and the output signal is examined for modulation of the upper frequency by the low-frequency tone. The amount by which the low-frequency tone modulates the high-frequency tone indicates the degree of nonlinearity. As with harmonic distortion measurement, this test may be done with a spectrum analyzer or with a dedicated distortion analyzer.

The modulation components of the upper signal appear as sidebands spaced at multiples of the lower-frequency tone. The RMS amplitudes of the sidebands are summed and expressed as a percentage of the upper-frequency level.

The most direct way to measure SMPTE IM distortion is to measure each component with a spectrum analyzer and add their RMS values. The spectrum analyzer approach has a drawback in that it is sensitive to frequency modulation of the carrier as well as amplitude modulation. A distortion analyzer for SMPTE testing is quite straightforward. The signal to be analyzed is passed through a high-pass filter to remove the low-frequency tone, as shown in Fig. 2.11. The high-frequency tone is then demodulated as if it were an amplitude modulated signal to obtain the sidebands. The sidebands pass through a low-pass filter to remove any remaining high-frequency energy. The resulting demodulated low-frequency signal will follow the envelope of the high-frequency tone. This low-frequency fluctuation is the distortion component and is displayed as a percentage of the amplitude of the high-frequency tone. Because low-pass filtering sets the measurement bandwidth, noise has little effect on SMPTE IM measurements.