
Chapter

13.3

Nonlinear Audio Distortion

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13.3.1 Introduction

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. Distortion of a device is measured by feeding it one or more sine waves of various amplitudes and frequencies. In simplistic terms, any frequencies at the output which were not present at the input are distortion. However, strictly speaking, components due to power line interference or other spurious signals are not distortion. There are many methods of measuring distortion in common use: harmonic distortion and at least three different types of intermodulation distortion. These are different test procedures rather than different forms of distortion in the device under test.

13.3.2 Harmonic Distortion

The transfer characteristic of a typical device is shown in Figure 13.3.1. This 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.

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 Figure 13.3.2. For a 1-kHz input, the output consists of 1 kHz, 2 kHz, 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.

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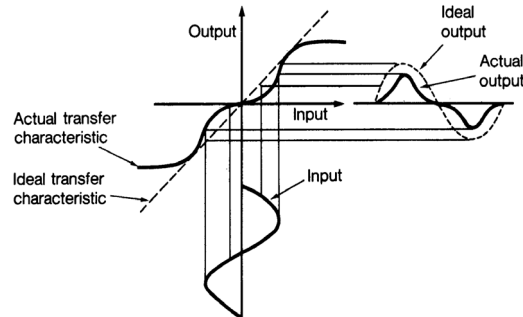


Figure 13.3.1 Total harmonic distortion (THD) test of transfer characteristic.

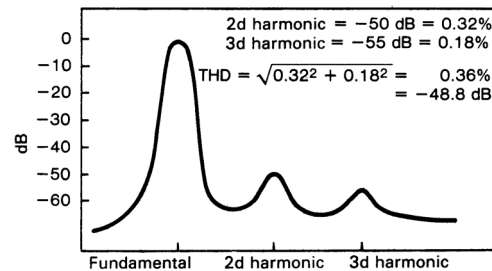


Figure 13.3.2 Reading THD from a spectrum analyzer.

To measure harmonic distortion with a spectrum analyzer the procedure illustrated in Figure 13.3.2 is used. 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 and represents the THD. This procedure is time-consuming and difficult for an unskilled operator. Even skilled operators have trouble in obtaining accuracies better than 2 dB in the final result because of equipment limitations and the problems inherent in reading numbers off a trace on the screen of an analyzer.

A simpler approach to the measurement of harmonic distortion is 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 Figure 13.3.3. The fundamental is removed with a notch filter, and its output is then measured with an ac voltmeter. Since distortion is normally presented as a percentage of the fundamental level, 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 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

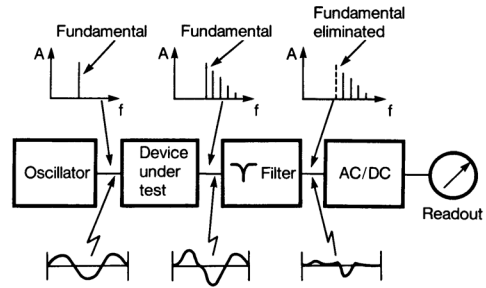


Figure 13.3.3 Basic block diagram of a harmonic distortion analyzer.

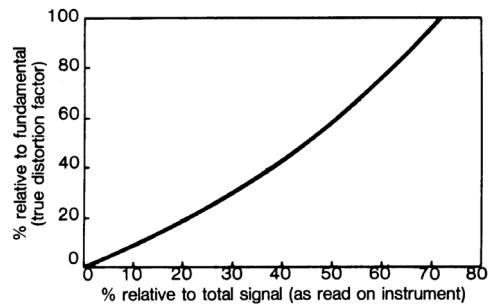


Figure 13.3.4 Conversion graph for indicated distortion and true distortion. (From [1]. Used with permission.)

the attenuators and ranges. This provision significantly reduces the effort and skill required to make a measurement.

The correct method of representing percentage distortion is to express the level of the harmonics as a fraction of the fundamental level. However, 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 relationship between the measured distortion and the true distortion is given in Figure 13.3.4. The errors are negligible below 10 percent measured distortion and are not significant until 20 percent measured distortion.

The need to tune the notch filter to the correct frequency can also make this a very tedious measurement. Some manufacturers have circumvented this problem by including the measurement oscillator and analyzer in one package, placing the analyzer and oscillator frequency controls on the same knob or button. This eliminates the problem only when the signal source used for the test is the internal oscillator. A better approach, used by some manufacturers, is to measure the input frequency and tune the filter to the measured frequency. This eliminates any need to adjust the analyzer frequency.

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 Figure 13.3.5. Any practical signal contains some hum and 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 factor does limit the readings of equipment for very low distortion, it is not necessarily bad. Indeed it can be argued that the ear hears

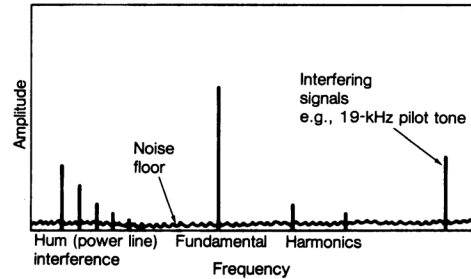


Figure 13.3.5 Interference in distortion and noise measurements.

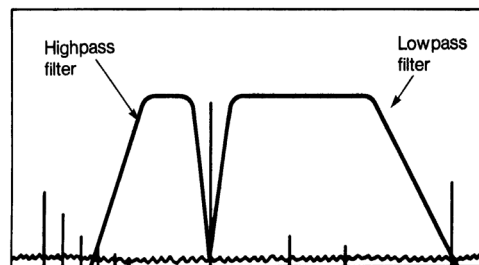


Figure 13.3.6 Use of filters to reduce interference in THD measurement.

all components present in the signal, not just the harmonics. Some interfering signals, such as the 19-kHz pilot tone used in frequency-modulation (FM) stereo, may be outside the range of audibility and therefore totally undesirable.

Additional filters are included on most distortion analyzers to reduce unwanted hum and noise as illustrated in Figure 13.3.6. 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 kHz, 30 kHz, and 80 kHz. Better equipment will include filters at all these frequencies to ease the tradeoff between limiting bandwidth to reduce noise and the reduction in reading accuracy from removing desired components of the signal. When used in conjunction with a good differential input on the analyzer, these filters can solve most practical measurement noise problems.

The use of a sine-wave test signal and a notch-type distortion analyzer has the distinct advantage of simplicity in both instrumentation and use. This simplicity has an additional benefit in ease of interpretation. The shape of the output waveform from a notch-type analyzer indicates the slope of the nonlinearity. Displaying the residual components on the vertical axis of an oscilloscope and the input signal on the horizontal gives a plot of the transfer characteristics' deviation from a best-fit straight line. This technique is diagramed in Figure 13.3.7. The trace will be a horizontal line for a perfectly linear device. If the transfer characteristic curves upward on positive input voltages, the trace will bend upward at the right-hand side. Examination of the distortion components in real time on an oscilloscope will show such characteristics as oscillation on the peaks of the signal, crossover distortion, and clipping. This is a valuable tool in the design and development of audio circuits and one which no other distortion measurement method can fully match. Viewing the residual components in the frequency domain using a spectrum ana-

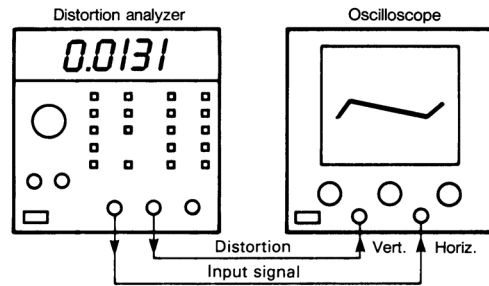


Figure 13.3.7 Transfer function monitoring.

lyzer also yields considerable information about the distortion mechanism inside the device under test.

Both the frequency and the amplitude of the sine-wave stimulus are adjustable parameters in harmonic-distortion testing. This often proves to be of great value in investigating the nature of a distortion mechanism. By measuring at low frequencies, thermal distortion and related effects may be examined in detail. Using frequencies near the power line frequency and looking for beats in the distortion products can reveal power supply limitations and line-related interference. Measurements at high frequencies can reveal the presence of nonlinear capacitances or slew-rate limiting. By examining the slope of the distortion change with frequency, several mechanisms which are active in the same frequency range may be isolated.

Limitations when measuring distortion at high frequencies are the major problem with THD testing, as illustrated in Figure 13.3.8. Because the components being measured are harmonics of the input frequency, they may fall outside the pass-band of the device under test. A tape recorder with a cutoff frequency of 22 kHz (typical for a good machine) will allow measurement only up to the third harmonic of a 7-kHz input signal. THD measurements on a 20-kHz input are impossible because all the distortion components are filtered out by the recorder.

13.3.3 Intermodulation Distortion

The commonly accepted solution to the limitations of THD analysis is to measure distortion by the interaction or intermodulation (IM) of two or more signals passing through a device simultaneously. Many methods have been devised to measure this interaction. 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 may be done with a spectrum analyzer or with a dedicated distortion-analysis instrument. The modulation components of the upper signal appear

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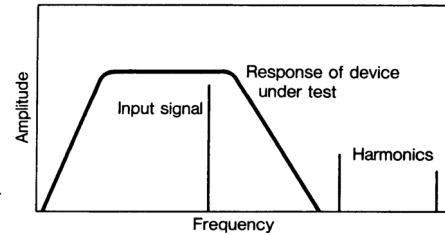


Figure 13.3.8 Problems in measuring harmonic distortion in band-limited systems.

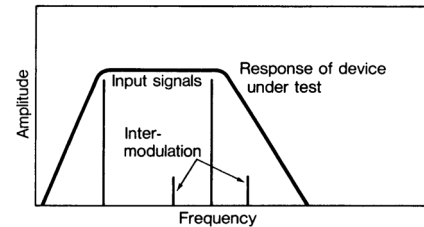


Figure 13.3.9 Measuring intermodulation in band-limited systems.

as sidebands spaced at multiples of the lower-frequency tone as illustrated in Figure 13.3.9. The amplitudes of the sidebands are rms-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 rms-sum them together. The spectrum-analyzer approach has a drawback in that it is sensitive to frequency modulation of the carrier as well as amplitude modulation. Because Doppler effects cause frequency modulation, this approach cannot be used on loudspeakers. Similar problems result from the wow and flutter in tape recorders.

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 Figure 13.3.10. The high-frequency tone is then demodulated as if it were an amplitude-modulation (AM) radio signal to obtain the sidebands, which are low-pass-filtered to remove any remaining high-frequency energy. The resultant demodulated low-frequency signal will follow the envelope of the high-frequency tone. This low-frequency fluctuation is the distortion and is displayed as a percentage of the high-frequency tone's amplitude. Since this low-pass filtering sets the measurement bandwidth, noise has little effect on SMPTE measurements. The analyzer is very tolerant of harmonics of the two input signals, allowing fairly simple oscillators to be used to generate the test signal. Indeed, early analyzers used a filtered version of the power line for the low-frequency tone: hence the 60-Hz low tone frequency in the SMPTE standard. It is important, however, that no harmonics of the low-frequency signal generator extend into the measurement range of the high-frequency tone. The analyzer will be unable to distinguish these from sidebands. After the first stage of filtering in the analyzer, there is little low-frequency energy left to create IM in the analyzer. This considerably simplifies the remaining circuitry.

As shown in Figure 13.3.11, when this composite signal is applied to the test device, the output waveform is distorted. As the high-frequency tone is moved along the transfer characteristic by the low-frequency tone, its amplitude changes. The high-frequency tone is being used to measure the gain at each point of the transfer characteristic. This results in low-frequency amplitude

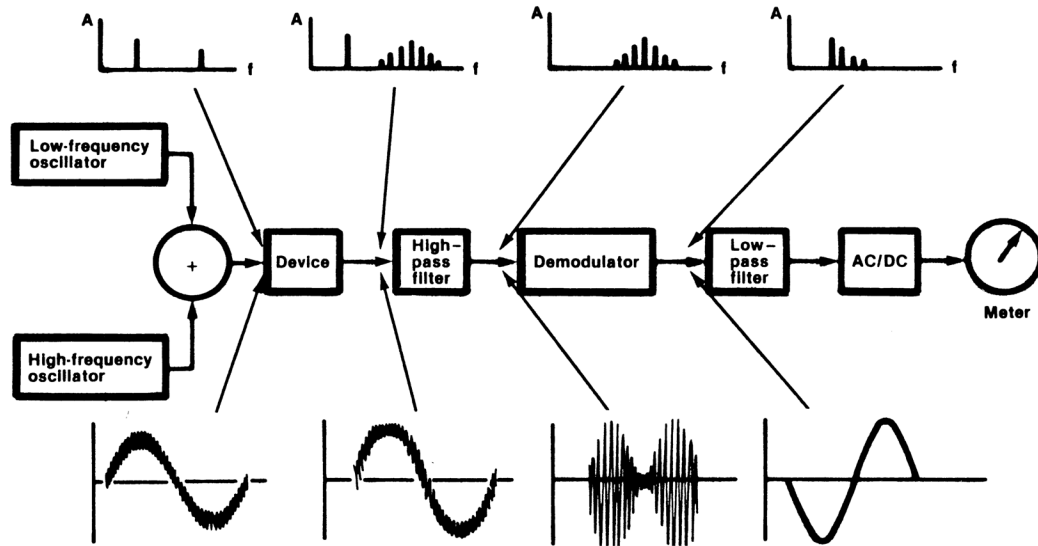


Figure 13.3.10 Basic block diagram of an SMPTE intermodulation analyzer.

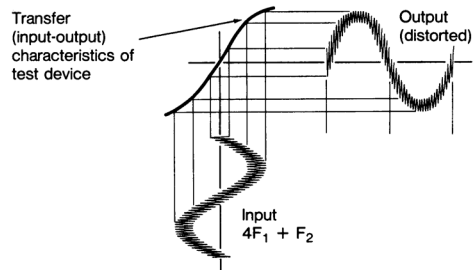


Figure 13.3.11 SMPTE intermodulation test of transfer characteristic.

modulation of the high-frequency tone. This modulation results in sidebands around the high-frequency lone as described previously. This test is therefore quite sensitive to such conditions as crossover distortion and clipping. High-order nonlinearities create bumps in the transfer characteristic which produce large amounts of SMPTE IM.

SMPTE testing is also useful for exciting low-frequency thermal distortion. As the low-frequency signal moves around, exciting thermal effects, the gain of the device changes, creating modulation distortion. Another excellent application is the testing of output inductance-capacitance (LC) stabilization networks in power amplifiers. Low-frequency signals may saturate the output inductor, causing it to become nonlinear. Because the frequency is low, very little voltage is dropped across the inductor, and there is little low-frequency harmonic distortion. The high-frequency tone will develop a signal across the inductor because of the rising impedance with frequency. When the low-frequency tone creates a nonlinear inductance, the high-frequency tone becomes distorted. This illustrates an important advantage of SMPTE IM testing. The sensitivity

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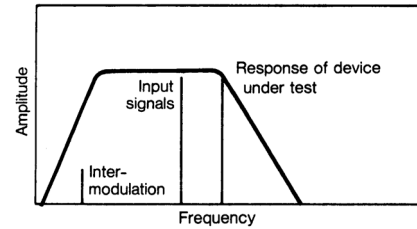


Figure 13.3.12 CCIT intermodulation in band-limited systems.

may be quite high to low-frequency distortion mechanisms in that the components occur at high frequencies. In most electronic audio circuits there is feedback which decreases at high frequencies. This lower feedback allows the distortion components to escape much of the reduction otherwise achieved at low frequencies by such feedback.

The inherent insensitivity of SMPTE IM to wow and flutter has fostered widespread use of the SMPTE test in applications which involve recording audio signals. Much use is made of SMPTE IM in the film industry, for example.

It is often claimed that because the distortion components in an SMPTE test are not harmonically related to either input they will be more objectionable to the ear. Musical instruments are rich in harmonics but contain few if any components which are inharmonic. With the typical 60-Hz low-frequency tone used in SMPTE measurements, it is doubtful that the sidebands could be outside the masking range of the ear. However, it is quite possible for the test to be indicative of audible performance even if the test-signal distortion is not audible.

A number of studies have compared SMPTE IM readings with harmonic-distortion readings. For most classic transfer-function nonlinearities, the SMPTE test is approximately 12 dB more sensitive than THD. However, when heavy feedback is employed or when dynamic distortion effects are present, the difference becomes considerably less predictable.

13.3.3a CCITT IM Test

Twin-tone intermodulation or International Telephone Consultative Committee (CCIT) *difference-frequency distortion* is another method of measuring distortion by using two sine waves. The test signal consists of two closely spaced high-frequency tones as shown in Figure 13.3.12. When these are passed through a nonlinear device, IM products are generated at frequencies related to the difference in frequency between the original tones. For the typical case of signals at 14 kHz and 15 kHz, the IM components will be at 1 kHz, 2 kHz, 3 kHz, etc., and 13 kHz, 16 kHz, 12 kHz, 17 kHz, 17 kHz, and so on. Even-order, or asymmetrical, distortions produce low-frequency difference-frequency components. Odd-order, or *symmetrical, nonlinearities* produce components near the input-signal frequencies. The most common application of this test measures only the even-order components since they may be measured with only a multipole low-pass filter. Measurement of the odd-order components requires a spectrum analyzer or a selective voltmeter. The measurement residuals in the two cases are approximately 110 dB and 85 dB below the input signals, respectively. The CCIT test has several advantages over either harmonic or SMPTE IM testing. The signals and distortion components may almost always be arranged to be in the passband of the device under test. This method ceases to be useful below a few hundred hertz when the required selectivity in the spectrum analyzer or selective voltmeter becomes

excessive. However FFT-based devices can extend the practical lower limit substantially below this point.

The distortion products in this test are usually far removed from the input signal. This positions them outside the range of the auditory system's masking effects. If a test which measures what the ear might hear is desired, the CCIT test is the most likely candidate.

13.3.3b Specialized IM Tests

Ladegaard [2] discusses the use of swept-frequency two-tone IM tests to study *transient intermodulation* (TIM), or slope-induced distortion. This approach has numerous advantages over the specialized tests which have been developed for the same purpose. The ability to adjust the test frequency enables qualitative study of the underlying distortion mechanism. Factors such as the steepness of the change in distortion with frequency and the frequency at which the change begins are useful in separating static from dynamic distortion mechanisms. The dominant order of the nonlinearity also gives useful information about its origins.

The totally in-band character of the test is one of its most attractive attributes. The difference product at $f_1 - f_2$ is widely separated in frequency and is easy to measure with a low-pass filter. This provides a simple implementation of the measurement and is the one most commonly used. However, the difference-frequency product is responsive only to even-order, or asymmetrical, nonlinearities in the device under test. It is possible to measure the odd-order products without a spectrum analyzer by using a precision squarer and bandpass filters. This approach has limitations in dynamic range because of the multiplier or squarer and its attendant inaccuracies.

Another approach, suggested by Thiele [3] and Small [4], involves a clever choice of test frequencies to place both even- and odd-order components near each other. The idea is to choose the two test frequencies in almost a 3:2 frequency ratio. If they were exactly a 3:2 ratio, the second-order and third-order products would both fall at the 1 position in frequency; i.e., for tones of 10kHz and 15 kHz, both distortion components would fall at 5 kHz. Both components may then be measured with a single multipole bandpass filter tuned to this frequency. To prevent possible cancellation of the two distortion products if they should happen to be of opposite phase, the frequencies are offset slightly. The distortion components will then be at slightly different frequencies, yielding an unambiguous measurement.

The sine-wave-square-wave IM test was originally proposed by Leinonen, Otala, and Curl [5]. It consists of summing a square wave at 3.18 kHz and a sine wave at 15 kHz (of one-fourth of the square-wave amplitude) and applying them to the device under test. The signal is low-pass-filtered at either 30 kHz or 100 kHz and is described as DIM (dynamic intermodulation) 30 or DIM 100, respectively. As the amplifier slews on the corners of the square wave, it becomes nonlinear and distorts the sine wave. The resulting IM components are measured with a spectrum analyzer, rms-summed, and expressed as a percent of the 15-kHz sine-wave amplitude.

The test-signal spectrum is shown in Figure 13.3.13. Leinonen and associates [5] define nine IM components in the audio band to be summed when making the measurement. If the square wave is not exactly 50 percent duty cycle, there will be even-order harmonics of 3.18 kHz in the test signal. These even-order-harmonic components of the square wave, when introduced by the device under test, are not included in the measurement. However, these components may be as large as the IM components with some test devices and do provide valuable insight into the device nonlinearities. Making this measurement, as originally defined, requires a sine-wave generator, a square-wave generator, a single-pole low-pass filter, a spectrum analyzer, a calculator,

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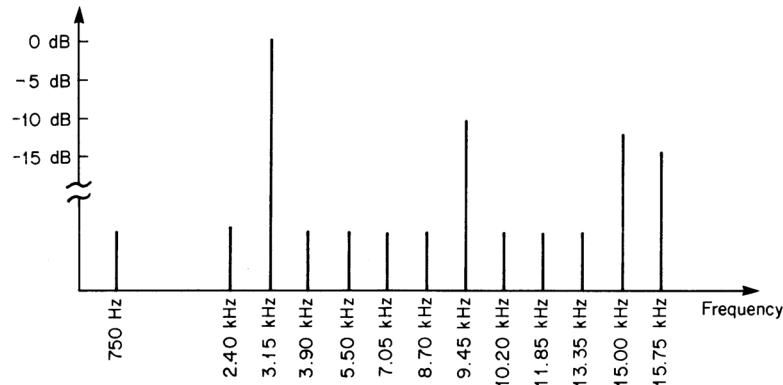


Figure 13.3.13 Spectrum of sine-wave-square-wave TIM test.

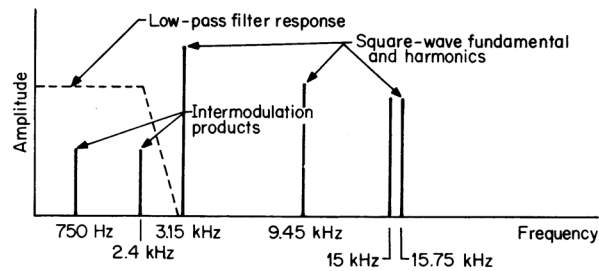


Figure 13.3.14 Simplified TIM test-measurement system.

and a considerable amount of time. The procedure is a fixed-frequency test, for there is no practical way to adjust the test frequencies to examine intricacies of the device under test.

Owing to its complexity in measurement, the test was slow to be adopted by the audio industry despite widespread interest in transient forms of distortion. Skritek [6] and later Hofer [7] developed a procedure that simplifies this measurement considerably. The basic approach is illustrated in Figure 13.3.14. Both an even-order and an odd-order IM product fold down to the 2-kHz region from interaction of the square wave and sine wave. By using a square wave of 3.15 kHz and a sine wave at 15 kHz, these appear at 750 Hz and 2.4 kHz. The 3.18-kHz square wave will produce components at 900 Hz and 2.28 kHz. The amplitude of these two components may be measured with a high-order elliptic low-pass filter at approximately 2.5 kHz. Their amplitude is expressed as a percentage of the 15-kHz tone amplitude. Owing to the practical difficulties of measuring the 15-kHz signal directly, the rms amplitude of the total signal is measured and a correction factor is applied. Additional high-pass filtering at approximately 400 Hz may be used to eliminate the effects of hum on the measurement. A block diagram of a DIM distortion analyzer using this measurement approach is shown in Figure 13.3.15. Extensive testing by Hofer demon-

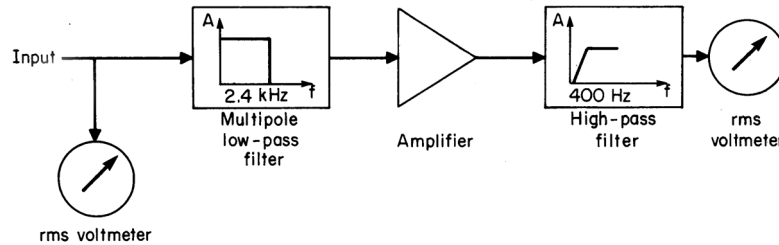


Figure 13.3.15 Block diagram of a simplified DIM distortion meter.

strated that these two products correlate adequately with the results obtained by adding all nine IM components.

13.3.4 Distortion-Measurement Hardware

Distortion measurements should be performed with an rms-responding meter in the distortion analyzer. This is necessary to make the reading represent the true power in the distortion. With most practical distortion measurements, the rms response will read about 2 dB higher than the average response. Some arguments can be made for measuring distortion with peak or quasi-peak detectors. The averaging time of the ear is between 50 and 250 ms, depending on test conditions and procedure. This has led to the definition of the quasi-peak meter for measuring telephone noise as described previously.

Accuracy of most distortion analyzers is specified at 1 dB, but this can be misleading. Separate specifications are often put on the bandwidth and ranges, as is common for voltmeters. A more important specification for distortion measurements is the residual distortion of the measurement system. Manufacturers of distortion analyzers often specify the oscillator and the distortion analyzer separately. A system in which the oscillator and the analyzer are each specified at 0.002 percent THD can have a system residual distortion of 0.004 percent. If the noise of the analyzer and/or the oscillator is specified separately, this must be added to the residual specification to find the residual THD + N of the system. It is not uncommon to find this limiting system residual at most input voltages. For example, an analyzer specified at 0.002 percent distortion and 20- μ V input noise will have a 0.003 percent residual at 1-V input and 0.02 percent at 0.1-V input. These voltages are common when measuring analog mixing consoles and preamplifiers, resulting in a serious practical limitation with some distortion analyzers.

Many commercial units specify the residual distortion at only one input voltage or at the full scale of one range. The performance usually degrades by as much as 10 dB when the signal is at the bottom of an input range. This is true because THD + N measurements are a ratio of the distortion components and noise to the signal level. At the full-scale input voltage, the voltage in the notch filter is a maximum and the filter's noise contribution will be minimized. As the level drops, the residual noise in the notch filter becomes a larger percentage of the reading. When the next input range occurs, the residuals will improve again. This limitation is in addition to the

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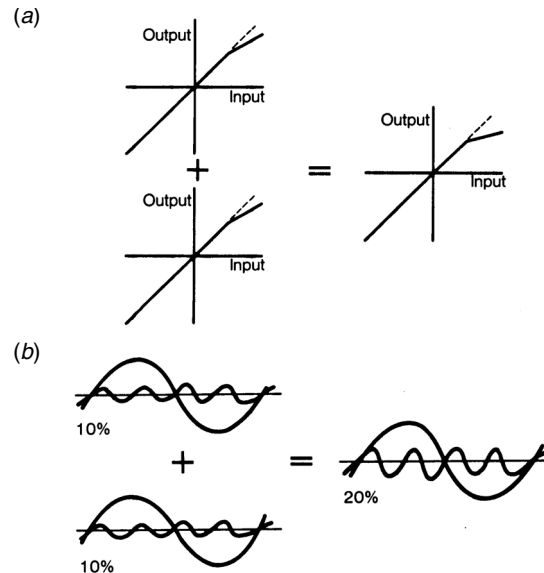


Figure 13.3.16 Addition of distortion: (a) addition of transfer-function nonlinearities, (b) addition of distortion components.

input-noise problem discussed previously because it results from noise in a later portion of the instrument.

13.3.4a Addition and Cancellation of Distortion

Another often-overlooked problem is that of distortion addition and cancellation in the test equipment or the device under test. Consider the examples in Figures 13.3.16 and 13.3.17. Suppose one device under test has a transfer characteristic similar to that diagramed at the top of Figure 13.3.16a and another has the characteristic diagramed at the bottom. If they are cascaded, the resulting transfer-characteristic nonlinearity will be magnified as shown. The effect on sine waves in the time domain is illustrated in Figure 13.3.16b. The distortion component generated by each nonlinearity can be seen to be in phase and will sum to a component of twice the magnitude. However, if the second device under test has a complementary transfer characteristic as shown in Figure 13.3.17, we obtain quite a different result. When the devices are cascaded, the effects of the two curves will cancel, yielding a straight line for the transfer characteristic. The corresponding distortion products are out of phase with each other, resulting in no distortion components in the final output.

It is quite common for this to occur at low levels of distortion, especially between the test equipment and the device under test. For example, if the test equipment has a residual of 0.002 percent when connected to itself and readings of 0.001 percent are obtained from the circuit under test, cancellations are occurring. It is also possible for cancellations to occur in the test

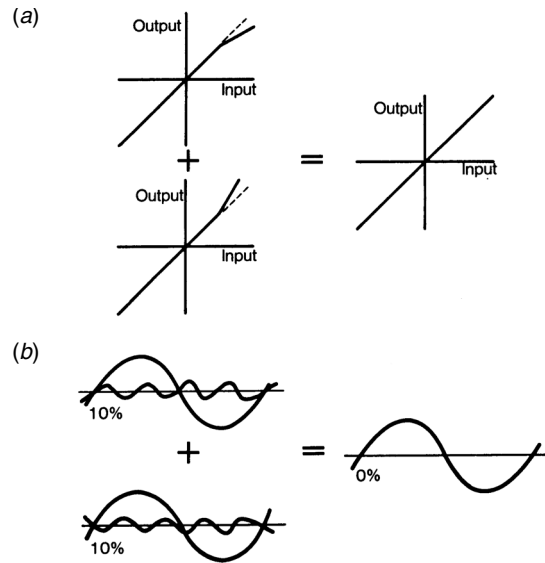


Figure 13.3.17 Cancellation of distortion: (a) cancellation of distortion waveform, (b) cancellation of transfer-characteristic nonlinearity.

equipment itself, with the combined analyzer and signal generator system giving readings lower than the sum of their individual residuals. If the distortion is from an even-order (asymmetrical) nonlinearity, reversing the phase of the signal between the offending devices will change a cancellation to an addition. If the distortion is from an odd-order (symmetrical) nonlinearity, phase inversions will not affect the cancellation.

13.3.5 Signal-Source Effects in Measurements

The signal source is an often-overlooked factor in the accuracy of level measurements. Let us consider the measurement of a low-pass filter's gain above its -3 -dB frequency. This is shown in Figure 13.3.18 for a three-pole low-pass filter. If the frequency of the generator is off by 3 percent, the gain measurement will be off by 1 dB. A higher-order filter or a less accurate generator will produce more error.

Figure 13.3.19 shows the effect of generator distortion on the gain measurement of a multi-pole high-pass filter. The harmonics of the generator are not attenuated as much by the filter as is the fundamental. If the signal-source distortion is high, as with function generators, the gain measurement will be in error. This effect is most important when measuring notch filters in an equalizer, where the distortion will appear as inadequate notch depth.

Figure 13.3.19 suggests that the effect of these errors on distortion measurements is more severe. The gain introduced by a filter on the harmonics of the generator can make them exceed

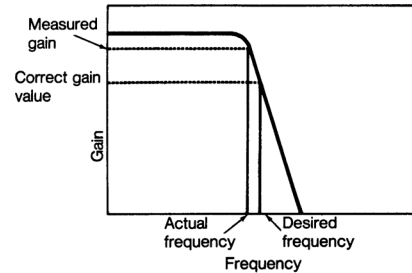


Figure 13.3.18 Frequency-accuracy effects in measuring filters.

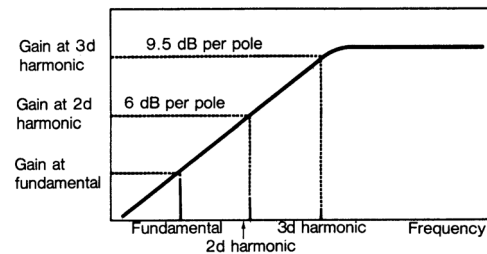


Figure 13.3.19 Source-distribution effects in high-pass filters.

the distortion of the filter itself. A three-pole high-pass filter will introduce 18 dB of gain at the second harmonic and 29dB of gain at the third. Under these conditions an oscillator which has 0.001 percent second- and 0.001 percent third-harmonic distortion will read 0.03 percent when measuring a distortion-free filter. These errors necessitate the use of an oscillator with very low distortion.

Another source of error in measurements is the output impedance of the generator. The amplitude and phase response of a device under test will often be affected by its input impedance interacting with the source impedance of the generator. They form a resistive divider in which the shunt leg is the nonconstant impedance of the device under test. This causes a variation of the voltage at the input to the test device, thus corrupting the measurements. Low-output-impedance generators will suffer less variation with load than high-impedance generators. However, if the system response is being measured, the generator impedance should be equal to the source impedance of the device normally driving that input. Transformer input stages often require a specific source impedance to provide correct damping for optimum high-frequency response. Too large a source impedance will cause excessive rolloff while too low a source impedance will produce an underdamped or peaked response. For example, most microphone inputs are designed to be driven from a 150-Ω source, a value close to the typical microphone source impedance.

13.3.6 References

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