TEXAS
INSTRUMENTS

Digital Audio Measurements

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ABSTRACT

To perform correct measurements on a true digital audio amplifier (TDAA), it is important to understand the characteristics of the device, especially in comparison with conventional linear amplifiers such as class-A and class-AB. This report provides an overview of TI's proprietary digital amplification technology and information about how measurements on TDAA's must be performed in order to obtain correct results.

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Introduction

Q How are digital amplifiers different?

A A digital amplifier incorporates a switching output stage that operates according to a principle known as pulse width modulation (PWM). In contrast to the low-frequency control signal used in a conventional analog amplifier output stage, the switching output stage of a TDAA is controlled by a high-frequency digital signal. The high-frequency digital signal is a PWM signal, in which the music is modulated into a fixed-frequency carrier signal. In a TDAA, the modulation process is based on a digital pulse code modulated (PCM) representation of the music signal (or other signal that needs amplification) from a CD player, a DVD player, or some other device.

Q What is a true digital audio amplifier?

A A TDAA is the combination of (i) a very high performance algorithm that enables artifact free PCM to PWM conversion, and (ii) a switching output stage (see Figure 1).The digital amplification technology causes the amplifier to have a different out-of-band spectrum compared to conventional amplifiers, i.e., a spectrum that requires attention to produce correct measurements of such audio parameters as dynamic range (DR) and distortion plus noise (THD+N).
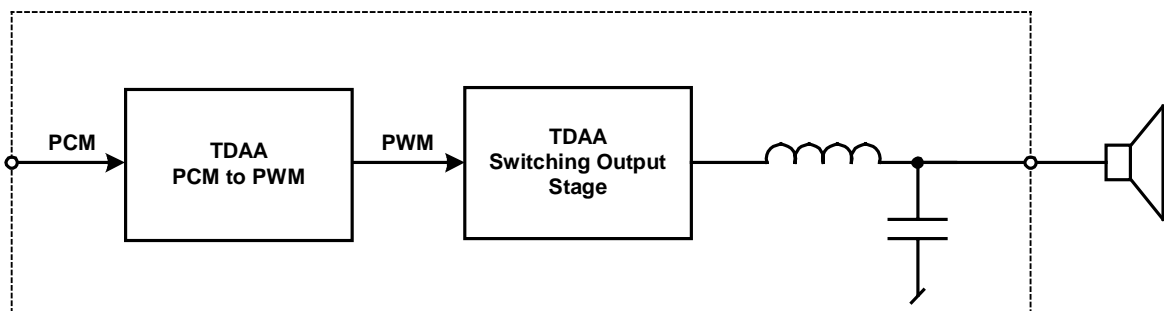

Figure 1.   TDAA With Low-Pass Filter

In principle, a full-bridge TDAA output stage consists of two sets of switches that are controlled on/off by the PWM signal. This way, the small-signal PWM control signal is converted into a power signal with an amplitude corresponding to the output-stage supply voltage. The resulting output is demodulated in a low-pass filter which reconstructs an amplified version of the original input signal waveform, for example, music. The typical demodulation filter is a simple, passive, second-order filter located just prior to the speaker connector. It is interesting to note that the *only* analog signal path in a TDAA is the demodulation filter.

Q What are the features of TI's proprietary digital amplification technology?

A
- High power efficiency, i.e., no heat sinks, no fans

- All digital, easy to integrate with digital front-end

- SMD technology

- Low weight

- Small size

- EMC compliant

- High sonic capability

How to Perform Measurements on a TDAA

This section deals with the output spectrum of a digital amplifier, audio analyzers, and measurement procedures.

Q How does the switching topology affect the output spectrum?

A As already described, a TDAA incorporates a PWM-controlled switching output stage and includes a simple low-pass filter for demodulation. In a high-performance amplifier, it is necessary to keep an adequate margin between the audio bandwidth and the PWM switching frequency. Typically, the ratio is around 20, corresponding to a switching frequency of about 400 kHz. Depending on the order and cut-off frequency of the demodulation filter, the output spectrum from a PWM-based amplifier always contains some level of frequency intermodulation (IM) components around the switching frequency and/or its harmonics. Figure 2 shows an approximation to the output spectrum with the IM components shown as vertical blue lines in the high-frequency part of the plot. The IM components are greatly attenuated by the demodulation filter and will, therefore not cause any problems when fed to a speaker. However, in contrast to measuring on a conventional class-A or class-AB amplifier, these IM components require revised measurement procedures.
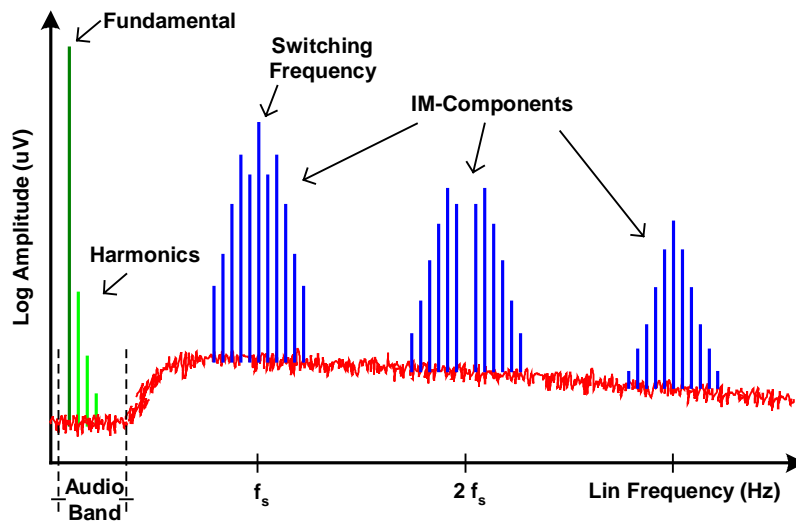
![Figure 2 showing approximate output spectrum with log amplitude versus linear frequency, displaying Fundamental, Harmonics, Switching Frequency, and IM-Components]

Figure 2.   Approximate Output Spectrum

Q   How does TDAA technology affect the output spectrum?

A   To accommodate the common problem of converting a discretely-sampled analog waveform (e.g., PCM) to PWM, the conversion algorithm uses over-sampling and noise shaping. Over-sampling is used to expand the signal bandwidth, typically from 22 kHz or 48 kHz to about 200 kHz (4–8 times over-sampling). At this sampling rate, noise shaping is used to obtain a very low noise level in the audio band (< 20 kHz) at the expense of a higher noise level in the out-of-band region, plotted in red on the output spectrum shown in Figure 2. Like the out-of-band PWM spectrum, the out-of-band noise shaping characteristics do not cause any problems when fed to a speaker, because the noise level is more than 80 dB below maximum output. However, to obtain correct audio performance measurement results, traditional measurement techniques must once again be modified.

Q   Why does the out-of-band region affect the audio analyzer?

A   Many audio analyzers are designed to measure conventional analog amplifiers. Although different kinds of bandwidth-limiting filters can often be utilized, the design of the input section of the analyzer can still create a potential problem.

The signal chain in a typical audio analyzer is shown in Figure 3. This block diagram shows that although bandwidth-limiting filters can be applied, the filters are located too late in the signal chain to ensure that the gain-scaling and tracking-notch filter circuitry functions optimally. Typically, the dynamic range of the audio analyzers is not fully used unless precautions are taken to avoid overloading the gain-scaling section.
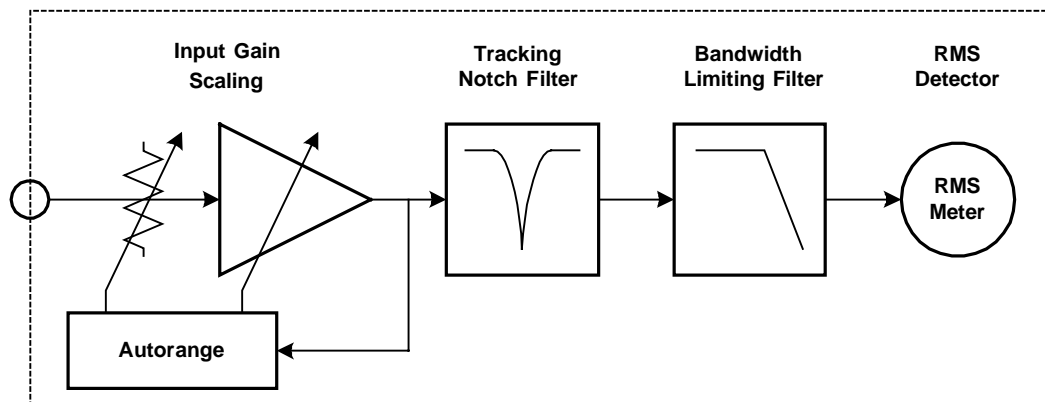
Figure 3.   Simplified Audio-Analyzer Block Diagram

Q   How can the effect of out-of-band components be minimized?

A   To deal with this general measurement problem, (which, incidentally, is accentuated when making measurements on sigma-delta DACs) the Audio Engineering Society (AES) has specified a new measurement standard *AES Standard Method for Digital Audio Engineering—Measurement of Digital Audio Equipment* (AES17). Audio Precision (AP) has implemented a new low-pass filter type in accordance with AES17 §4.2.1.1 which, in AP's terminology, is referred to as the AES17 filter. In an AP audio analyzer, this new filter can be placed quite early in the signal chain, thereby minimizing the effect of out-of-band components.

Q   What can be done to make the measurements correctly?

A   Using an AP system featuring the AES17 filter is strongly recommended. However, if this is not an option, an existing 20-kHz bandwidth-limiting filter, together with a passive low-pass filter between the device under test (DUT) and the audio analyzer is recommended.

Q   How should the measurement method be modified?

A   In simple terms, a conventional analog amplifier not only amplifies the input signal, but it also adds harmonics that result from its own nonlinearities. The noise floor is typically somewhat flat in and beyond the audio band. A common standard for measuring THD+N specifies a measurement procedure in which the fundamental is removed with a tunable notch filter. The remaining harmonics plus noise are then measured with an analog RMS voltmeter. The voltmeter reading is compared to the RMS value of the fundamental to give the THD+N ratio. Typically, this measurement standard gives the correct result for a conventional analog amplifier. However, when used for a *switching amplifier*, the measurement result can be wrong.

The problem with the method lies in the measurement bandwidth. For conventional amplifiers, this bandwidth is not a major factor, at least not as long as the ratio of the noise-floor level to the magnitude of the harmonics is large enough. However, for a PWM-based digital amplifier, the inaudible out-of-band frequency components affect the measurement. To make correct measurements on digital amplifiers, a bandwidth-limiting filter (e.g., 20 kHz BW) must be used before the audio analyzer. This prevents the audio analyzer from being overloaded by out-of-band noise, and it ensures correct gain scaling in the region of interest.

Q How can the efficacy of measured parameters be evaluated?

A It is recommended that any parameter always be evaluated using two different measurement procedures. As an example of this, a THD+N value can be obtained using the measurement method just described, and also by a procedure that uses an FFT analysis followed by post-processing. In fact, performing an FFT analysis is advantageous, because it shows each of the contributing components of both the THD and the N (i.e., noise shaping, hum, and IM-components) of the THD+N value.

Q How should the FFTs and post-processing be performed?

A A number of factors are important with regard to performing FFT analysis. These factors fall into one of two categories:

- Considerations on the analyzer input and dynamic range

- General FFT parameters

As already discussed for audio analyzers, to obtain the correct gain-scaling, and thereby retain the full dynamic range of the analyzer, care must be taken not to overload the FFT analyzer.

Depending on the analyzer's dynamic range, it may not be possible to perform full-scale analysis in which the levels of both the fundamental and noise floor are correct. Typically, insufficient dynamic range is seen as noise-floor modulation at different power output levels. If the dynamic range in the audio analyzer is inadequate, it is recommended that full-scale measurements be performed by using a notch filter to remove the fundamental, thereby ensuring correct levels of the noise floor and harmonic components.

The general FFT parameters that affect the appearance of the FFT plot are as follows:

- Numbers of sampling points in the data sequence ($FFT_{(length)}$)

- Data sampling frequency ($FFT_{(freq)}$)

- FFT window function

In general, great care must be exercised when comparing FFT plots made with unequal parameters. For comparison purposes, it is recommended that the FFT measurements be used in conjunction with post-processing to calculate the desired parameter value, e.g., DR from 20 Hz to 20 kHz, RMS noise in a specific frequency range, THD for 2nd to 7th harmonics. Such post-processed values can be compared without further consideration or concern.

The following is an example of the post-processing of FFT data.

To find the 20 – 20 kHz RMS noise ($V_{(rms, noise)}$) for an amplifier running with no signal, an FFT analysis with a length of 16K samples, a sampling frequency of 44.1 kHz and Blackman-Harris window is performed, and post-processing is undertaken according to the steps:

1. Perform the FFT analysis to achieve a noise floor measurement.

2. Based on the FFT, calculate the $V_{(rms, noise)}$ as follows:

The FFT parameters give a bin width of:

$$\frac{FFT_{(freq)}}{FFT_{(length)}} = \frac{44100}{16384} = 2.69165 \ \ Hz/bin$$

The target band bin (integer) values can be found:

$$\text{Start bin:} \frac{20 \ Hz}{2.69165 \ Hz/bin} \approx 7 \qquad\qquad \text{Stop bin:} \ \frac{20\,000 \ Hz}{2.69165 \ Hz/bin} \approx 7430$$

Calculate the RMS noise in the target band, taking the FFT window energy loss into account:

$$V_{(rms,noise)} = K_{(WindowCorrection)} \cdot \sqrt{\sum_{n=7}^{7430} V_{(bin,n)}^2}$$

Correction factors for various types of windows are given in Table 1

Table 1.   Correction Factors for Various FFT Window Types

FFT Window Type	K$_{(WindowCorrection)}$
Blackman-Harris	0.7610
Hanning	0.8165
Hamming	0.8566
Bartlet	0.8660
No window	1.0000

Q   How should the balanced speaker output be handled?

A   In a conventional amplifier, the black speaker output terminal (–) usually represents ground or chassis level and the red terminal (+) potential swings between positive and negative. In a TDAA, the speaker output is balanced and has a common-mode voltage relative to ground. This means that the potential of both output terminals varies out of phase, centered around a voltage potential equal to half the output-stage supply voltage. This mode of operation requires that, unlike conventional amplifiers, *the black output terminal must never be tied to ground or the chassis*. Measurements on the speaker output from a TDAA must always be performed as balanced measurements using a differential probe and measurement equipment with a common mode rejection greater than half the output-stage supply voltage.

Measurement Setup

To produce correct measurement results, it is important to have a well-controlled measurement setup that eliminates or minimizes potential error sources. It is also important to have the following measurement equipment:

- Audio analyzer, preferably Audio Precision (AP) System Two or Cascade

- Coaxial cable for SPDIF input to DUT

- Shielded twisted-pair (balanced) measurement cable for AP

- Adjustable power supply with low noise, preferably with adjustable current limit

- Dummy loads for speaker-output channels (e.g., 4–8 Ω)

- Oscilloscope, including differential probe

- Six high-precision multimeters (RMS voltage, current)

- Test cables with low series resistance

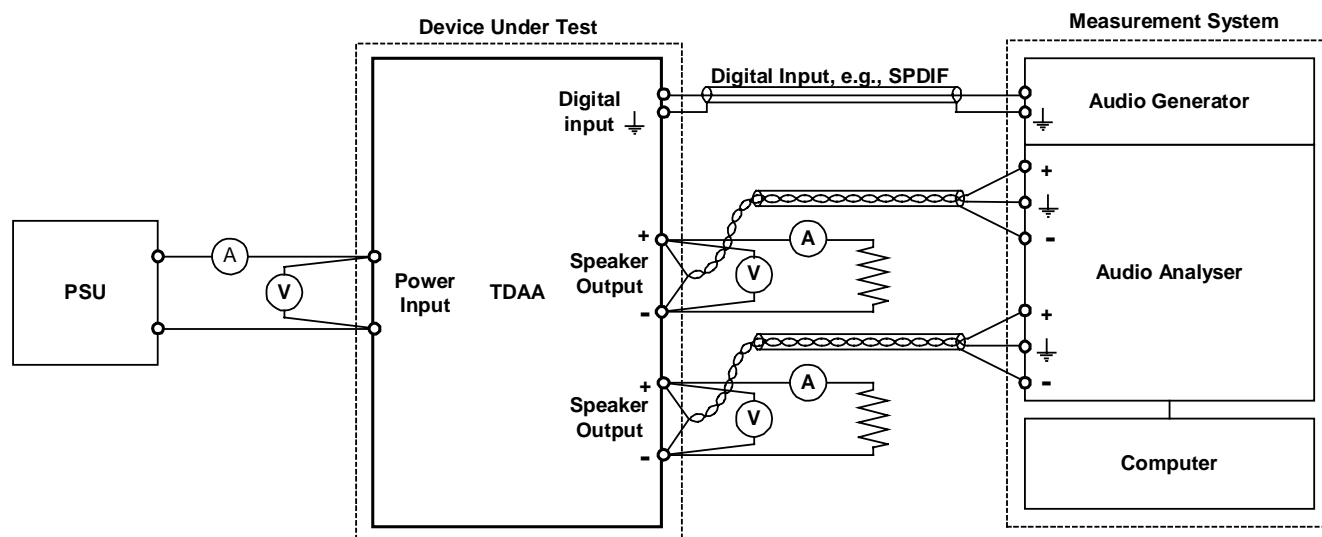The basic test setup is shown in Figure 4.

Figure 4.   Test Setup

A number of aspects need careful consideration, and it is recommended that these general guidelines be followed:

1. Measurements on speaker output terminals *must* be performed using a balanced measurement technique. Use a differential probe or AP balanced input. Single-ended measurements using floating measurement of any kind are *not* recommended. For measurements done with an Audio Precision (or similar) analyzer, always use a high-quality balanced cable, e.g., a shielded twisted-pair type. Keep all wires short.

2. Always use a regulated power supply. Before starting the measurement session, validate the quality of the PSU by performing an audio-bandwidth analysis on the PSU using, for example, FFT analysis.

3. Use short, heavy-gauge cable from the PSU to the DUT. The high resistance of light-gauge cable can sometimes affect the low-frequency response of THD+N measurements.

4. Be careful not to create any ground loops in the measurement setup. The audio analyzer equipment must have only *one* ground connection to the DUT. Typically, good results can be obtained if the ground connections are made via the measurement cable from the audio analyzer to the DUT; for example, via the shield around the twisted-pair cable. Depending on the measurement equipment, good results can often be obtained when grounds are connected via the SPDIF cable, and the shield of the measurement cable is only connected at the analyzer end. It is recommended that a number of different setups be tested and compared before the actual measurement session begins. FFT analysis is an excellent tool for this task because hum components can be easily distinguished.

5. All measurement equipment must be connected directly to the terminals of the DUT, *not* via the PSU or a dummy load. Especially during efficiency measurements, it is important to eliminate the influence of resistance in both the PSU wires and the speaker wires.

6. To validate the measurements, be sure to know the functionality of the measurement equipment —especially the input section—including gain-scaling electronics and the band-limit filter cutoff frequency and steepness. Depending on the measurement equipment, it is sometimes beneficial to use a passive prefilter prior to the analyzer.

How to Obtain Typical Audio Performance Parameters

Total Harmonic Distortion Plus Noise (THD+N)

The THD+N measurement gives a good indication of how well the DUT functions because it includes harmonic components (THD), as well as quantization noise, intermodulation distortion, hum, and so on (N). Depending on the measurement bandwidth, the N part of the THD+N varies. As already described, a switching amplifier has some out-of-band components that, if not correctly filtered, affect measurements—in other words, the noise (N) part will completely dominate the THD part. To produce valid measurement figures, the bandwidth must be specified to be in the audible region, from 20 Hz to 20 kHz.

If adequate bandwidth-limiting filters are available, the measurement can be performed using a notch filter to remove the fundamental, and an RMS voltmeter to measure the harmonics plus noise. THD+N can then be calculated as the ratio between the RMS value of the fundamental and the RMS value of the harmonics plus noise. If the available band-limiting filters are inadequate, it is recommended that THD+N be measured by FFT analysis be followed by post-processing. This approach is also recommended for measuring the total harmonic distortion (THD).

Dynamic Range:

The dynamic range (DR) test is based on a THD+N measurement made on a signal of –60 dB, followed by adding 60 dB to the result to give the range between full output (0 dB) and the noise floor of the amplifier. To provide a valid result, the measurement bandwidth must be in the audible region (20 Hz–20 kHz).

Example 1

If the measurement with a –60 dB input signal level gives a THD+N of 53 dB, the dynamic range of the amplifier is (53 + 60) dB = 113 dB.

If the availability of band-limiting filters is inadequate, an FFT analysis followed by post-processing is an excellent way to measure DR. Based on an FFT, the DR can be calculated as 60 dB plus the ratio in dB between the energy in a narrow window around the tone at –60 dB, and the combined energy of the harmonics and noise floor. The reason for calculating the ratio based on energy rather then RMS voltages is that the result will be independent of the FFT length and the FFT window function used.

Example 2

After performing the FFT, post processing can be done as follows: all bins in a narrow window around the fundamental are squared and summed. All other bins in the range

20 Hz to 20 kHz are squared and summed. Thereafter, the DR can be calculated as the ratio in dB between these two numbers, plus 60dB.

Signal-to-Noise Ratio

The signal-to-noise ratio (SNR) for an amplifier is the ratio between the maximum output signal and the noise level at zero output. An SNR value is meaningful only when it is accompanied by a specification of the measurement bandwidth. The measurement bandwidth is usually 20 Hz to 20 kHz, with or without A-weighing.

Using a band-limiting filter (e.g., 20 Hz to 20 kHz) and a RMS voltmeter, the SNR can be calculated as the ratio between the voltmeter readings of two measurements: the full-scale output measurement and the output measurement of the amplifier at zero input.

Example 1

To determine the SNR of an amplifier using the RMS voltmeter in an AP system:

1. Adjust the AP generator until maximum output from the DUT is obtained, then note the voltmeter reading.

2. Set the AP generator to zero and note the voltmeter reading.

3. Calculate SNR as the ratio of these readings.

Another way to measure SNR is by using FFT analysis data and post-processing to calculate the RMS noise in the desired frequency band.

Example 2

To find the 20-Hz to 20-kHz SNR, an FFT analysis is performed on the amplifier running with no signal. Before performing the FFT, the maximum RMS output signal capability of the DUT is measured using an RMS voltmeter as follows:

1. Adjust DUT to maximum output signal and measure the RMS value ($V_{(rms,max)}$).

2. Calculate the $V_{(rms,noise)}$ as shown in the example in the FFT section of this document.

3. Calculate SNR as the ratio between $V_{(rms,max)}$ and $V_{(rms,noise)}$

THD+N vs Output Power

A plot of THD+N vs output power is developed via a series of THD+N measurements performed on the DUT running with a constant frequency of the fundamental sine wave, and made at increasing values of the output power. The plot gives a reliable indication of how the distortion varies across the entire output range.

In a TDAA, the plot of THD+N vs output power is pieced together from three different contributions, each one characteristic of the digital amplification technology. The three contributions typically look like the simplified curves shown in Figure 5 where each contribution is shown in a different color.
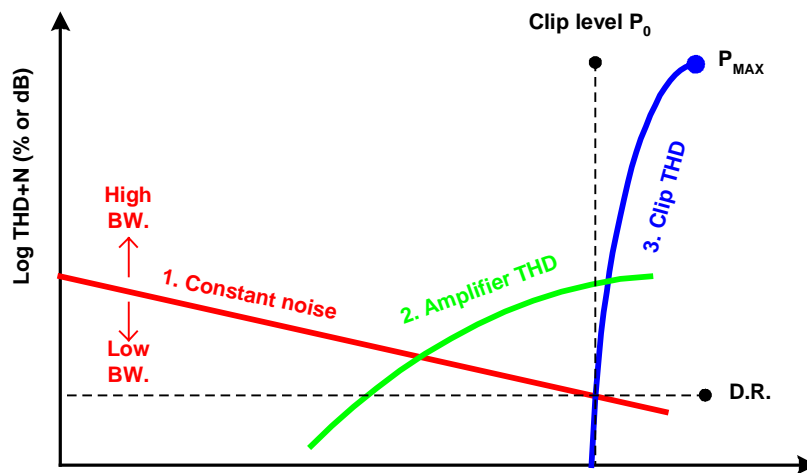
Figure 5   Three Contributions to the THD+N vs Output Power Curve

1. *Constant Noise*

   In the low to moderate output power range, the THD+N curve is dominated by noise (N) rather than THD. Since the noise floor energy is constant and does not change with respect to output power, the curve is approximately a straight line with a slope of -0.5 on a log-log plot, i.e., N depends roughly on the inverse square root of output power. Depending on the measurement bandwidth used, the vertical position of the curve varies as indicated by the small red arrows on the plot. This is caused by a higher bandwidth allowing more noise energy to be included in the measurement, and visa-versa. If the noise curve is extrapolated to intersect the amplifier clip-level power, P_O, the ordinate value at that point indicates the amplifier's dynamic range.

2. *Amplifier THD*

   At moderate to high power output levels, the THD+N curve is dominated by THD, i.e., by the harmonics of the fundamental that result from nonlinearities in the amplifier topology.

3. *Clip THD*

   For a true digital amplifier, this particular curve exists only if digital gain has been implemented as part of the signal processing. At outputs beyond the clipping level (indicated on the plot by P_O), the THD+N curve is dominated by THD. However, unlike the curve for amplifier THD, the harmonic distortion now results from the power-supply voltage being insufficient for the TDAA to produce an output signal with the desired amplitude, i.e., the waveform itself clips. If digital gain has been used, the curve rises steeply at P_O, indicating a fast raise in THD followed by a gradually-decreasing slope until the maximum output power is reached. Theoretically, the maximum output power is twice the clip-level power (P_O) and the THD level at that point is approximately 42%, which is the THD of a square wave.

Efficiency

Measuring efficiencies in the >80% range demands a well-controlled test setup in which error sources such as resistances in speaker/power supply wires are eliminated. To keep the tolerance on the measured efficiency tight, it is of great importance that the voltmeters and ammeters used are high precision types, preferably with 6½ digits of precision.

The recommended setup consists of:

- Two RMS voltmeters connected at the speaker output of the DUT

- Two RMS ammeters connected in series with the load resistors

- One dc voltmeter connected at the power-supply connector of the DUT

- One dc ammeter connected in series with the power-supply cable.

Note that when measuring efficiency at high output power, and depending on the type of ammeter used, the drop resistor inside the meter can heat up and result in inaccurate measurement. If such problems are discovered, avoid the use of ammeters altogether. Instead, use dedicated high-precision measurement-resistors in conjunction with accurate voltmeters.

Example

Using voltmeters and ammeters, the percent efficiency, η, is calculated as follows:

$$\eta = 100 \times \frac{V_{(rms,\ ch1)}\ I_{(rms,\ ch1)}\ +\ V_{(rms,\ ch2)}\ I_{(rms,\ ch2)}}{V_{(dc,\ input)}\ I_{(dc,\ input)}}$$

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