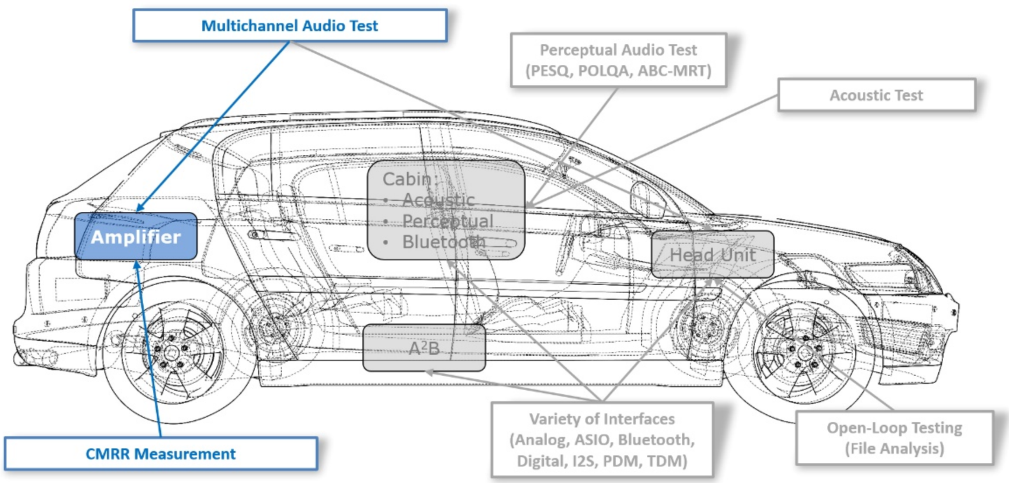
Application Note: Automotive Audio Testing – Amplifiers

Introduction

Even when viewed through an "audio lens," an automobile remains a complex system. As such, it makes sense to break down any discussion around testing into key subsystems and this paper is the first in a series from Audio Precision covering automotive audio measurement applications. Here we will focus on automotive audio amplifiers, and the types of tests that are normally made on these devices.



Background

Most car audio amplifiers today are class A/B or D, due to a desire to keep power consumption to a minimum while still maintaining high power output and an acceptable (low) distortion level. Auto manufacturers do not publish their proprietary test methods used for evaluating amplifier performance, but the kinds of measurements that one would want to make on automotive audio amplifiers are readily identifiable. Here we discuss the typical measurements made and identify specific areas of concern related to automotive systems.

Typical audio amp tests include gain and level, common mode rejection, power supply rejection, frequency response, output power and harmonic distortion, intermodulation distortion, noise floor, crosstalk, DC offset, and click and pop.

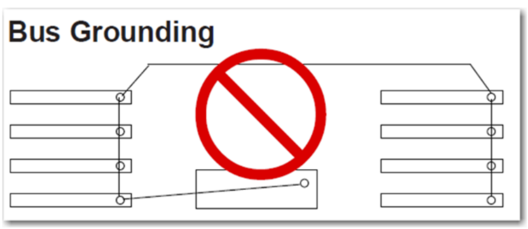
The instrumentation required to perform these tests include an audio analyzer, DC power supply capable of 9 to 16 VDC output, power meter (used for efficiency measurements), and a multimeter (used for checking loads and power voltage). Additional needed test items include non-inductive load resistors of the rated impedance and output power for the amplifier being tested, plus ground cables to connect the DUT (device under test) and the test equipment to a common ground.

From a “black box” perspective, this testing is consistent from one class of audio amplifier to another. Special conditioning of D class amplifier outputs is required, however, to address out-of-band noise that is unique to these amps. This is discussed further in the last section of this paper.

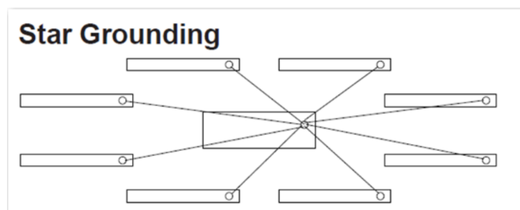
Grounding

Good grounding practice is important for optimizing audio amplifier performance, but it is also critical for achieving optimal measurement results. Small ground-potential differences between devices in the test system (such as switchers, the device under test, and the test instrument) can couple into the signal path and cause undesirable interference or noise due to the inherent stray capacitance between signal conductors and the chassis.

Bus grounding can sometimes seem a convenient method, but this technique often produces the worst results. The resistance in each leg of the chain puts the devices at different ground potentials and is not as effective as a star grounding method.



Bus Grounding – Not Recommended



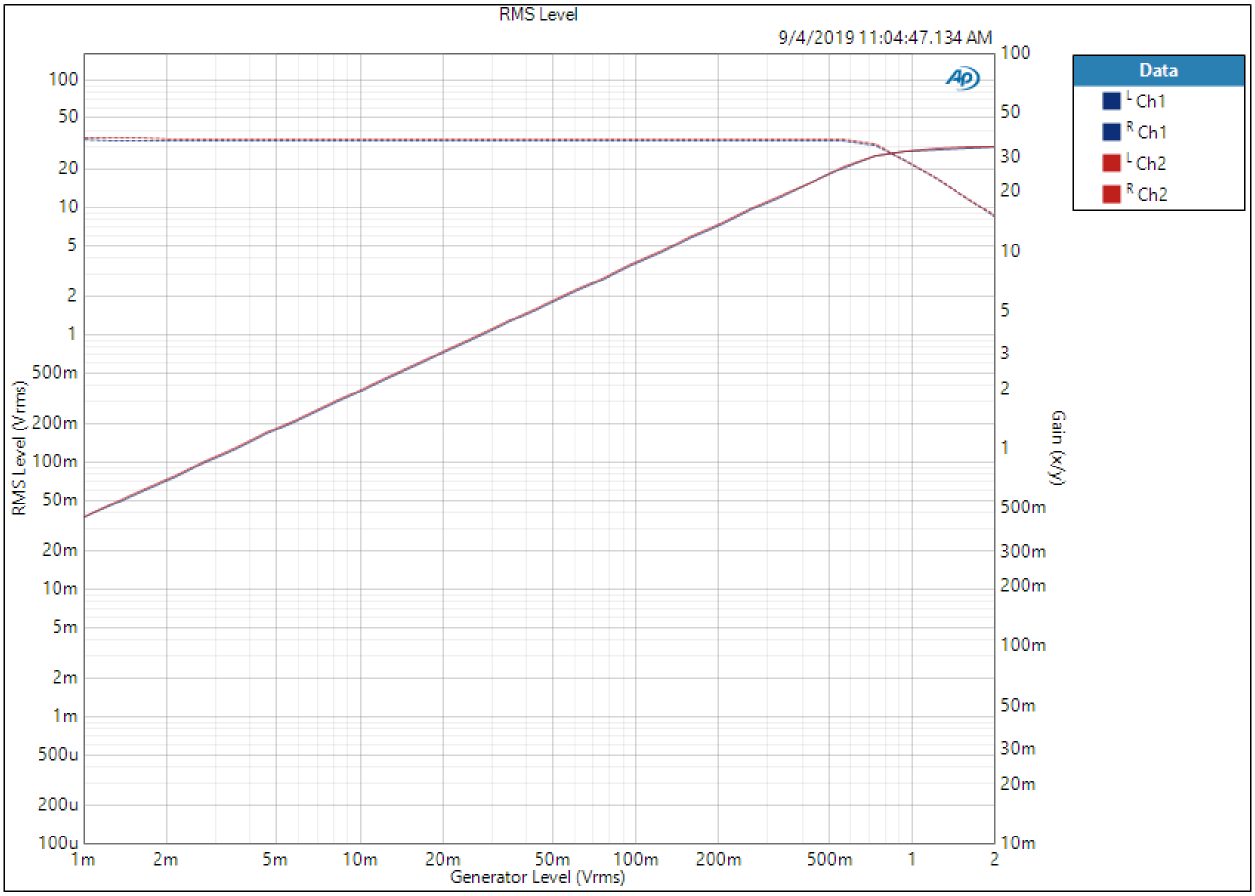
Star Grounding - Recommended

AP strongly recommends connecting the chassis ground of each device directly to the ground of the test instrument via low-impedance wires.

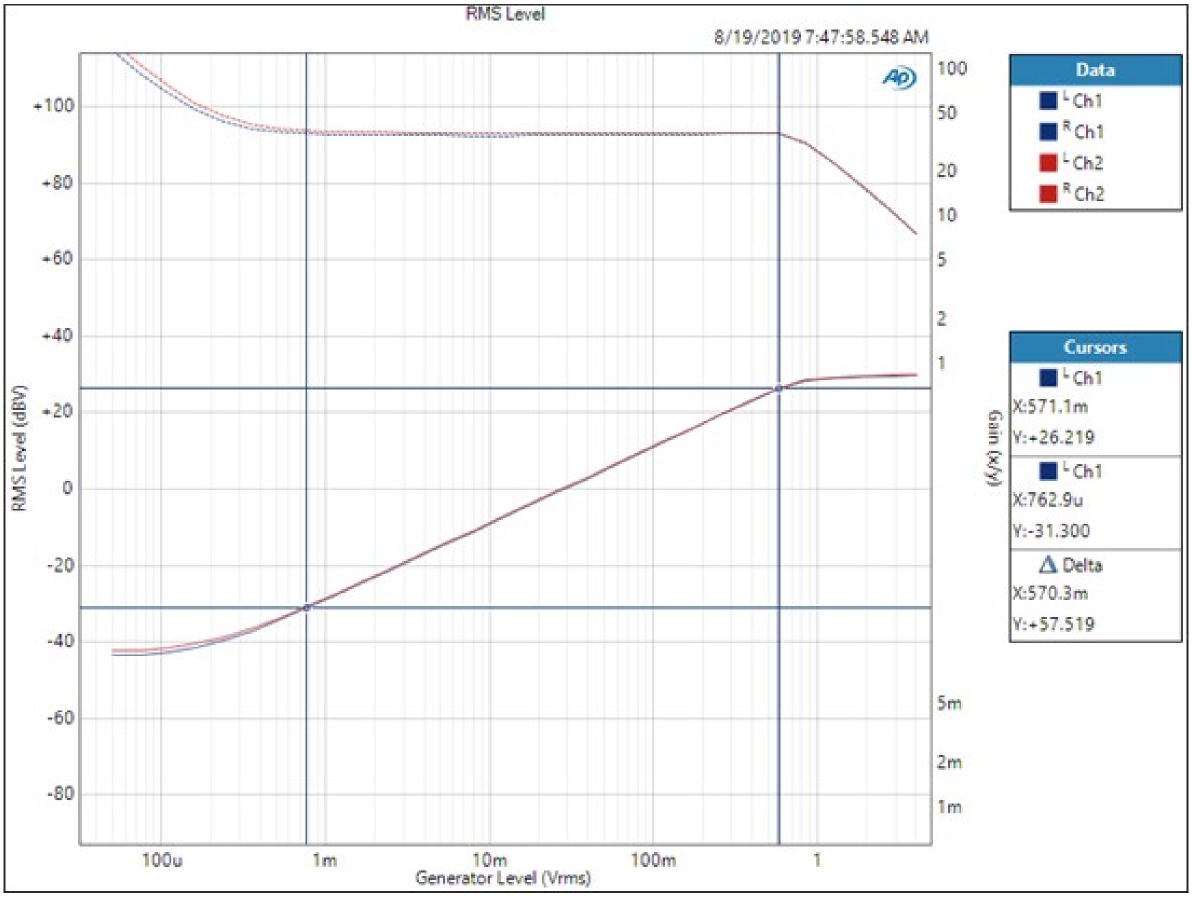
Measurements

Gain and Level

The gain and level tests can be conveniently performed by applying a stepped input level sweep, while measuring the output level and gain simultaneously on two channels. As you can see from the figure below, this amplifier has a gain of about 35 as shown on the upper trace (using the right vertical axis) and has a linear response with an input amplitude (horizontal scale on the plot) from below 2 mVrms to about 600 mVrms of input amplitude before it begins to clip at the output. The left axis is showing the measured output level in Vrms. Note that the Left and Right channels of this DUT are very closely matched.



If the DUT is tested over a broader input range, we can use this same measurement to visualize linear dynamic range. The following graph shows that this DUT was swept beyond the linear range at both the low and high end. This device demonstrates a gain of about 35, with a linear dynamic range of about 57 dBV. Note that the placement of the cursors that show the 57 dB range is somewhat ambiguous, and for this reason the SNR measurement is commonly used with the max amplitude applied to the system to produce a single-value dynamic range measurement.



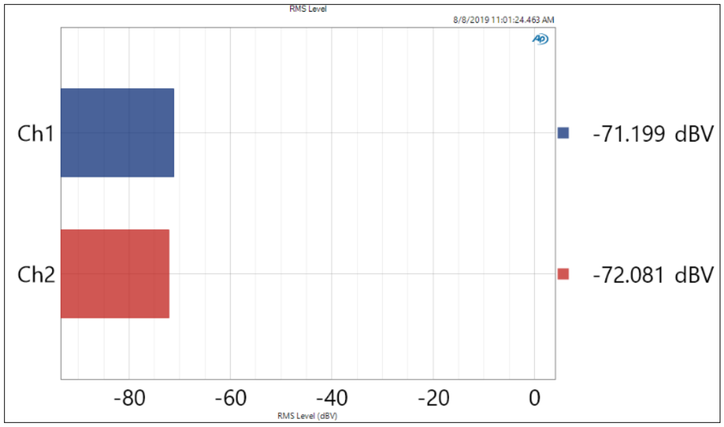
Noise Floor

Noise is the nemesis of a good audio system. Any unintended signal in a channel is often referred to as noise. This might be random noise which occurs anytime current is flowing in a circuit, or it might be a deterministic (non-random) signal that appears in the channel due to crosstalk, lack of adequate power supply isolation and filtering, poor grounding, or radiated as EMI from electric motors (e.g., windshield wipers, seats, sunroof) or ignition systems. If this noise occurs in front of a gain stage, the noise will be amplified.

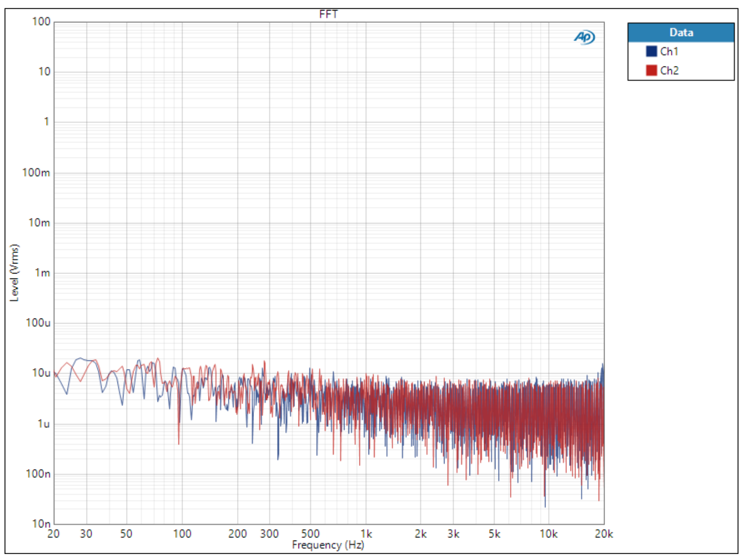
Knowing exactly how much noise is on the amplifier channel and where it might be coming from are critical bits of information in assessing the quality of an amp or working to troubleshoot a specific unit or installation.

Measuring the noise floor demonstrates how “quiet” the channel is with no signal applied. Therefore, the measurement is normally done by terminating the input of the DUT with a matched impedance and measuring the residual RMS amplitude with no signal applied to the input. Limiting the bandwidth of the measurement on the analyzer is critical, as more bandwidth means more noise in the measurement. For most automotive audio amplifiers, 20 Hz to 20 kHz is appropriate.

Here is an example of measurements made on an automotive audio power amp:



A look at the spectral view of this data shows that there is no dominant deterministic noise source (as it would show up as specific lines or spurs in the spectrum) and that the noise is primarily random in nature. If deterministic spurs appear in the spectrum when the inputs are terminated, the frequency of these spurs can provide a hint to the nature and source of the signal, which is the first step in troubleshooting the system or design.

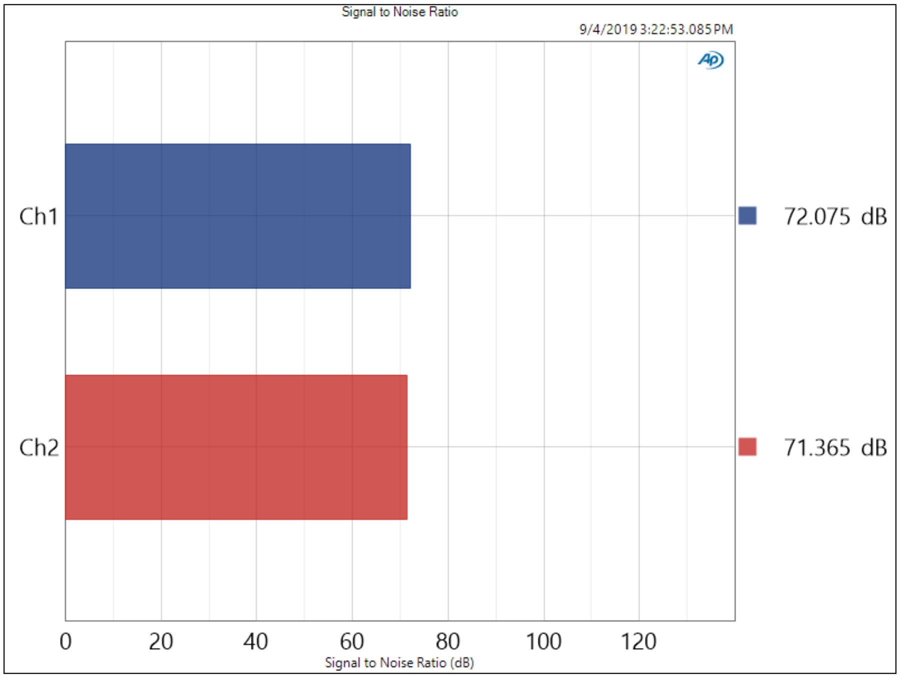


Noise is an important aspect of the automotive listening experience. Listeners can become quite sensitive when listening in a quiet vehicle with engine and other accessories switched off. Occupants can become acutely aware of the audio system's noise floor when turning the system on and/or off.

Noise measurements are sometimes made using an A-weighting filter, which attempts to factor in human hearing response.

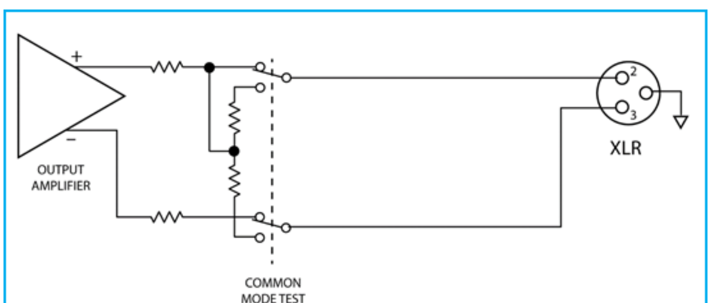
SNR and Common Mode Rejection

The method used for signal-to-noise ratio measurement is to inject a single tone into an amplifier input (at a specified input amplitude for the DUT) and measure the amplitude of the single tone at the output, then remove the input signal and measure the amplitude of the noise floor on that output, which provides the needed data points for the SNR. The following graph shows the measured SNR on a two-channel amplifier. This shows that there is over 70 dB difference at the output between the amplitude of the applied signal and the noise floor.



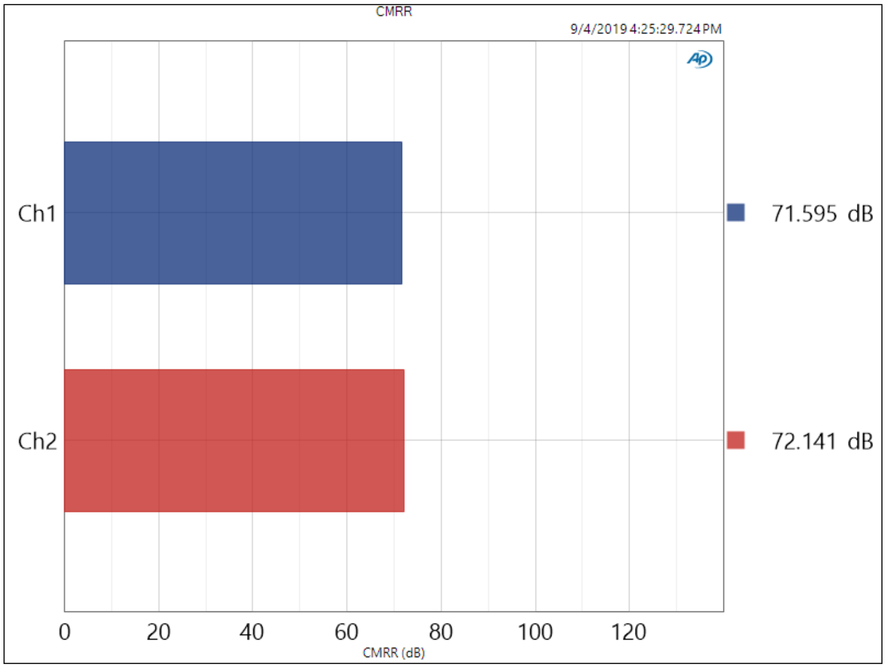
The common mode rejection ratio (CMRR) is a comparison of the output noise amplitude of an amplifier configured as a balanced (differential) output versus that same amplifier configured as an unbalanced (single-ended) output. This test demonstrates the effectiveness of the balanced output to provide random noise cancellation. CMRR is calculated as CMRR(dB) = $20*\log_{10}(V_{DM}/V_{CM})$.

The following test circuit is implemented within the Audio Precision APx analyzers which have two-channel generators (e.g., APx515, APx525, APx526, APx555, and APx582):



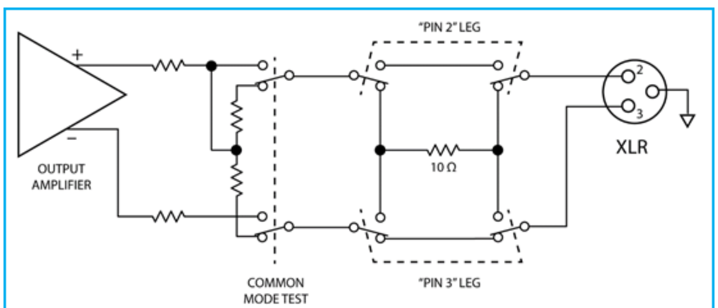
CMRR SIMPLIFIED OUTPUT SWITCHING

Here is an example output from this test:



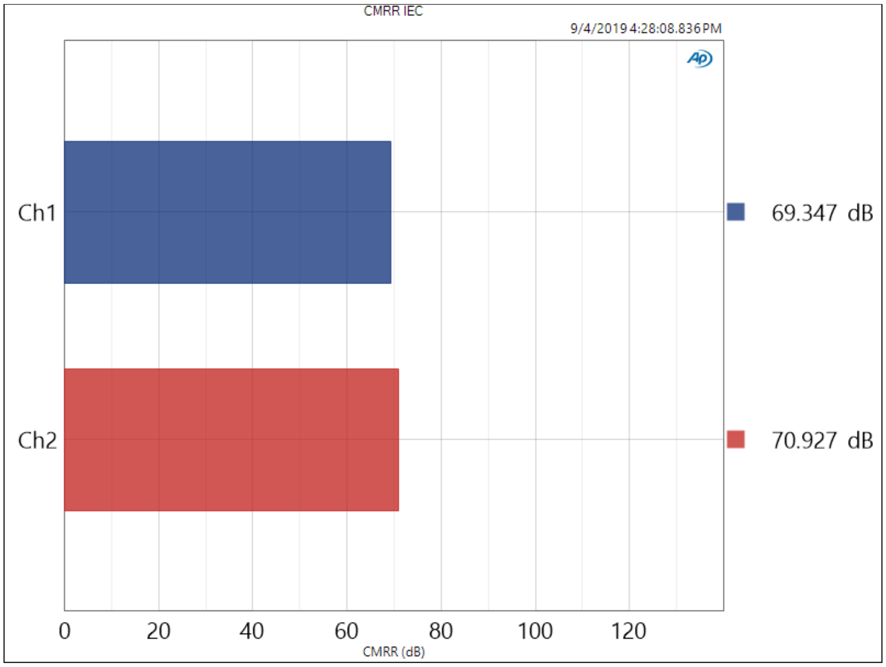
The IEC 60268-3 standard defines a slightly different implementation for the CMRR test. The differential signal is first measured and the valued stored. Next, each output leg is separately measured as a common mode output with a 10-ohm source resistance in series with each leg as defined by the IEC 60268-3 specification. The higher measured common mode level of the two outputs is used as the common mode value, and finally the calculation is performed as described in the previous paragraph.

The following is the test circuit prescribed by the IEC 60268-3 standard, and is also implemented in the Audio Precision APx analyzers that provide two-channel generators:



CMRR-IEC SIMPLIFIED
OUTPUT SWITCHING

Here is an example output from this test, using the same DUT that was tested in the prior CMRR measurement:



For a well-designed amplifier, these CMRR test results are typically in the 60 dB or greater range. You can see that the IEC CMRR method is slightly more pessimistic than the basic CMRR test. The Audio Precision APx500 audio analyzers (B Series or Legacy) are designed to perform either of these CMRR tests, which can be selected from the list of available measurements.

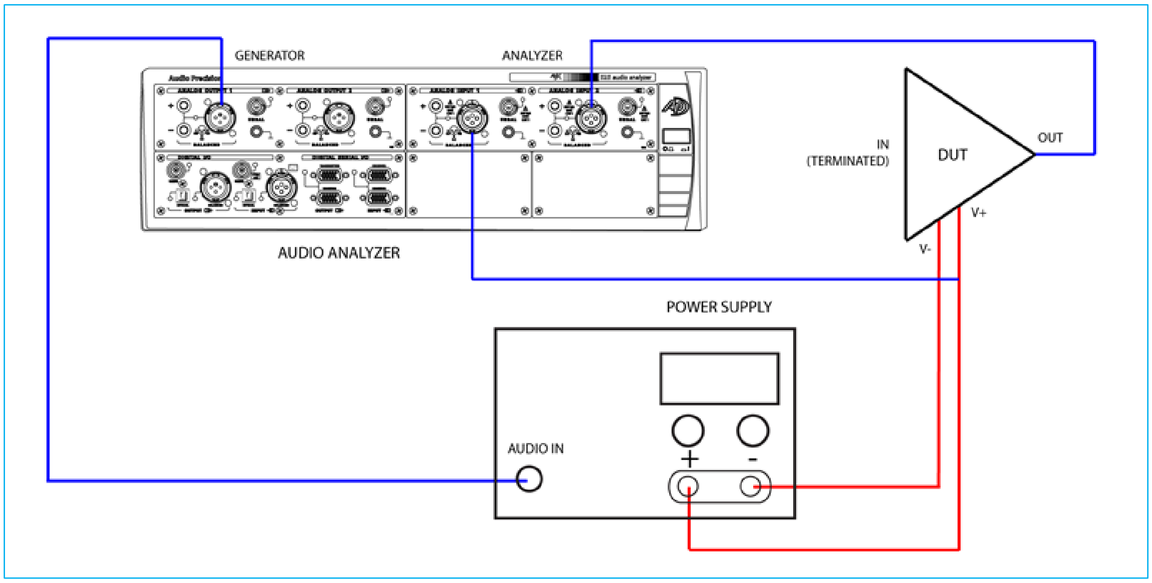
Power Supply Rejection Ratio (PSRR)

Electrical noise in the vehicle's power system can be coupled into the audio system, resulting in audible noise. The Power Supply Rejection Ratio (PSRR) test measures the ability of the amplifier to prevent power system noise from affecting the audio output. This is critical in systems powered directly from the vehicle battery. For systems designed to be supplied from a switch mode power supply, the supplies are typically designed with filters to remove the higher levels of power supply ripple typically associated with switching supplies, and therefore exhibit a higher level of power supply AC rejection.

One of the test challenges here is to configure a DC power source capable of driving the DUT at full power level, but that can also have the supply lines modulated by a prescribed level of AC signal riding on the DC level. One example of an instrument-grade power supply designed for this type of testing is the KEPCO BOP 20-20MC. Another potential DC-modulated power source is the Accel TS200. The Picotest J2121A is a Power Line Injector which can be used for PSRR measurements and is designed to be used in conjunction with an external fixed DC source. It is also possible to configure a DC-coupled power amplifier to be used as the supply to perform the PSRR test, provided it has enough power to properly supply the DUT.

A typical limit for this test is ≥ 60 dB across the entire sweep.

Here is an example of a PSRR test setup:

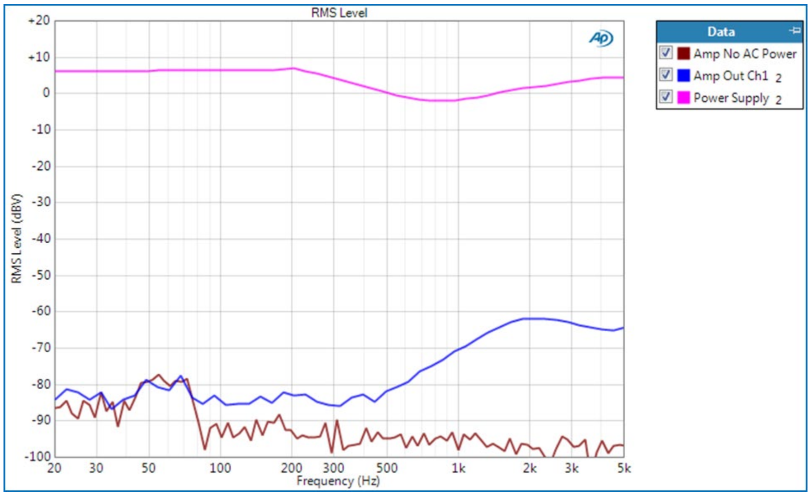


From this diagram, you can see that the output of one of the generators is connected to the modulator input (shown in the figure as AUDIO IN) of the power supply, providing the modulation for the output of the supply. Two inputs are used on the audio analyzer; one for measuring the amplitude of the signal on the DC supply output, and one to measure the related frequency on the output of the DUT. For this test, the DUT inputs are terminated.

The audio analyzer is set up to produce a swept single-tone output on the generator using the “Bandpass Frequency Sweep” selection in the Sequence Navigator. The sequence starts with an “RMS level” measurement, followed by the PSRR measurement. The final test data set is produced through a derived output which is the calculation of the PSRR. Note that the “window width” selection is used with the analyzer to create the narrowest bandpass filter possible, providing the most accurate representation of the output level based on the specific input frequency applied to the DUT for each step. This swept output is applied to the modulation input on the power supply.

The test starts with a measurement of the DUT RMS output with a clean (non-modulated) DC supply. This becomes the measurement “noise floor” for the PSRR at any specific frequency. If the DUT output does not exceed that noise floor, we can only measure the lower bound of the PSRR; the true PSRR value will be even higher. Next, the generator output is set up to sweep across a range typically from about 20 Hz to 5 kHz. At each frequency step, the RMS level is measured.

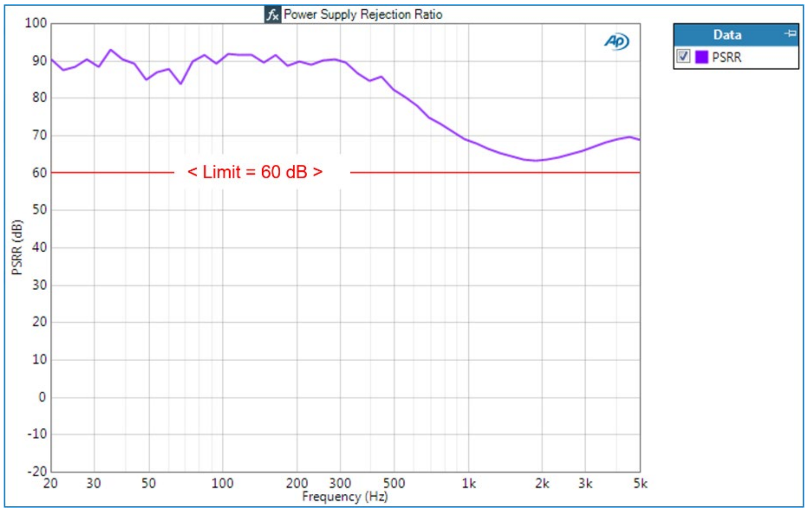
Here is an example raw data output from this test:



From this, the PSRR can be calculated as:

$$PSRR(f) = Power\ Supply(f) - Amp\ Out\ Ch1(f)$$

The resulting derived data set calculated by the analyzer is shown below:

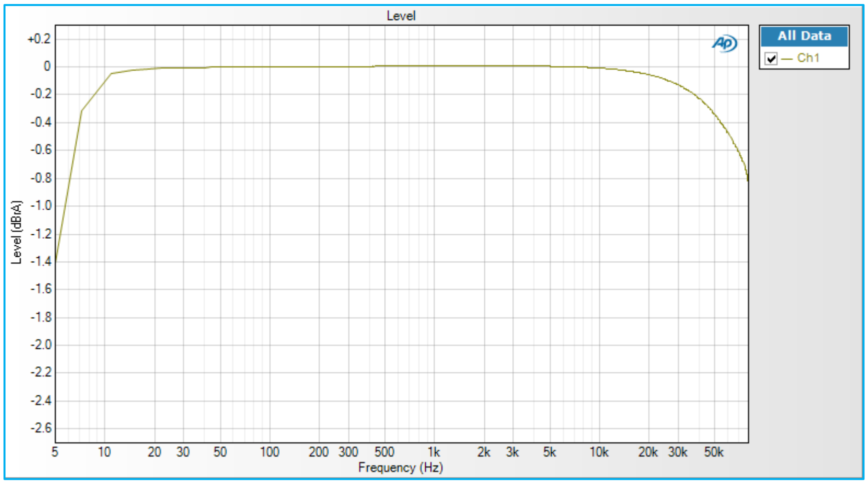


This plot shows that the DUT clearly meets the test limit of ≥ 60 dB across the 20 Hz to 5 kHz range. Note also that because the “Amp Out Ch1” level was not higher than the “Amp No AC Power” in the range of 20 Hz to about 70 Hz, we are only able to say that the DUT was as good or better than the performance shown. Ideally, the power supply AC voltage will be high enough to cause a peak in the DUT output at all given stimulus frequencies. This might not be possible under all conditions. An example setup for the power supply might include a DC level of 13.8 V and an AC level of approximately 2 Vrms.

Frequency Response

Frequency response provides a record of the output level variation of a DUT when stimulated with a range of input frequencies at a known level. For audio devices, this is generally done over the range of 20 Hz to 20 kHz, but for “high definition” audio it may extend to 45 or 90 kHz bandwidths.

The following is an example output from the frequency response test. This DUT shows very good amplitude flatness across the passband, with typical cutoff response at the high end.



Traditionally, this test has been done with a stepped frequency source. However, especially in environments such as production test, much of this work is now done using a logarithmically-swept sine (chirp) signal to substantially reduce test time. A 23-second stepped-frequency test, for example, can be replaced with a 4-second chirp test with comparable results.

Output Power and Harmonic Distortion

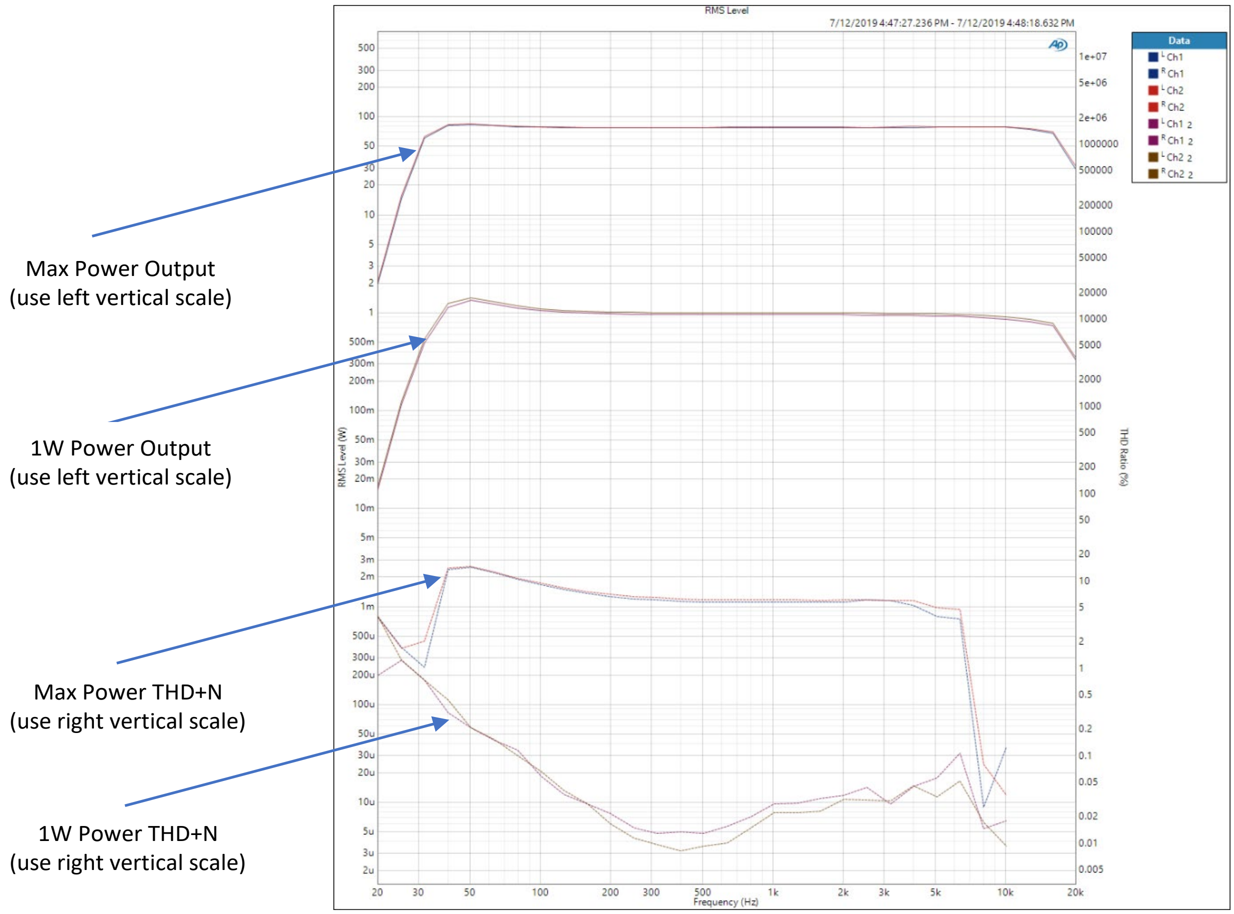
Output power and harmonic distortion are related; distortion generally increases as the output power is increased to full-rated power.

Two THD+N measurements are commonly done when testing an audio amplifier. One is done at 1 W output power, with all channels driven simultaneously. This focuses on small signal distortion of the entire amplifier. The other is a full-power test, with only one channel driven. This verifies that the DUT can reach the intended max power level design target and at the same time meet the distortion performance design target. The THD+N percentage can increase by up to 10x or more as the power of an amplifier is increased from a 1 W output reference level to the maximum rated power output. Here is an example of the target THD+N performance specs versus grade of equipment:

No.	Power Level	Channels Driven	Expected THD+N by Grade		
			Entry	Mid	Premium
1	1 W	all	≤ 1.0%	≤ 0.1%	≤ 0.01%
2	Rated power	1	≤ 10 %	≤ 1.0 %	≤ 0.1%

It is a general practice to perform the THD+N test with the analyzer bandwidth set to the full bandwidth of the DUT (e.g., 20 Hz to 20 kHz), but not wider, as this would add additional noise to the measurement from an irrelevant portion of the spectrum. However, there are differing views on the range of the applied frequency range from the analyzer source that should be used. It is common for the input to cover a range of 20 Hz to 6 kHz for the THD+N test. The reasoning here is that at 6 kHz, there are only the first and second harmonics still in band, and that testing above 6 kHz provides less useful data.

The following plot shows the measured output power from a DUT (using the left vertical scale to read the output power), and the corresponding THD+N measurement result (using the right vertical scale to read the THD+N percentage). The output power curves are in the upper portion of the graph, and the THD+N curves are in the lower portion. The curve nearest the bottom is the THD+N at 1 W output level, and the next curve up from that is the THD+N at maximum rated output power of 75 Watts.



We see here that the shape of the output power curve is consistent between the power levels, but that the THD+N curves are dramatically different. Also note that both THD+N curves drop precipitously at about 6 kHz, which demonstrates that the value of this measurement is diminished when harmonics are beyond the bandwidth cutoff.

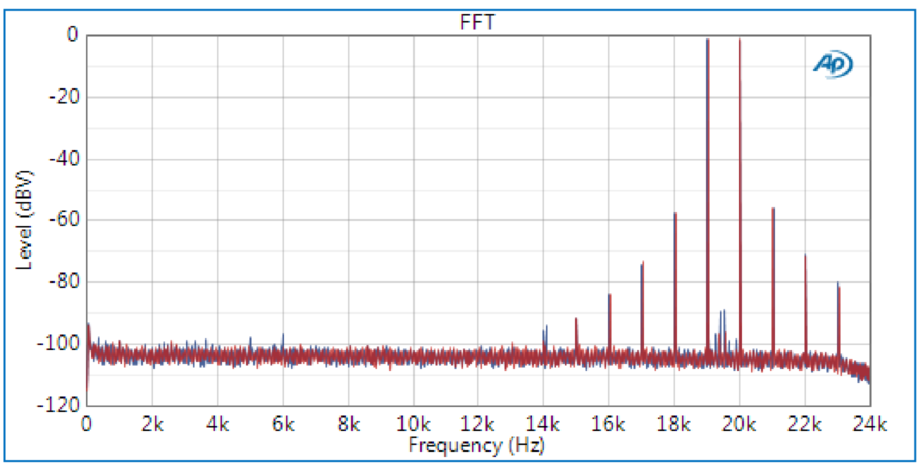
Intermodulation Distortion

As discussed in the previous section regarding THD+N, it is difficult to get useful data about harmonic distortion above 10 kHz because the harmonic products fall above the audio bandwidth and are often beyond the typical THD+N measurement bandwidth.

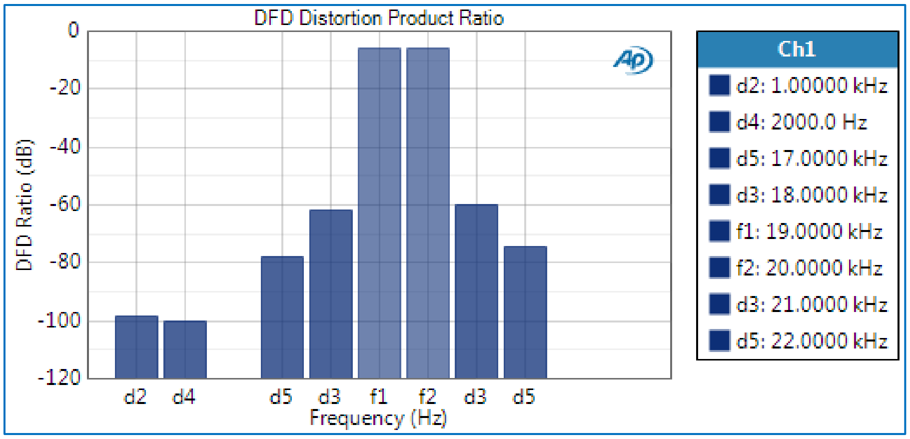
One technique used to investigate distortion in the upper range of the audio bandwidth is to simultaneously apply two relatively pure tones to the DUT, which produces beat frequencies (intermodulation products) in a non-linear device. The primary mechanism producing IMD in most devices is AM (amplitude modulation), which creates sidebands that are at the sum and difference of the frequencies of the original audio tones. The modulation products may also beat with each other and with the original audio signal, creating more modulation products.

Difference Frequency Distortion (DFD) is one example of a high-frequency two-tone IMD test, which was standardized in the IEC 60268-3 specification. The DFD test applies to both Class AB and Class D amplifiers. This test is typically done as part of a design and development cycle but is not generally done as part of validation or production test.

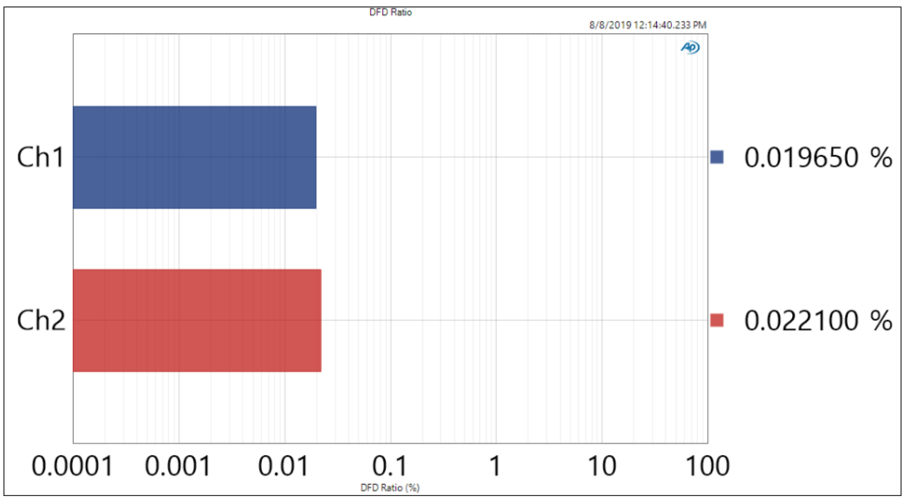
The following is an example output of a DFD test. In this case, the two equal-amplitude tones applied to the DUT are 19 kHz and 20 kHz. The analysis is done by calculating the ratio of the sum of the intermodulation harmonic amplitudes to the sum of the amplitudes of the two fundamental tones applied. Only the 2nd and 3rd order intermodulation products are considered in this ratio.



The test results for the DFD test are shown in the following figure. The 4th and 5th order products (shown as d4 and d5) are displayed in the plot for reference but are not part of the calculated ratio.



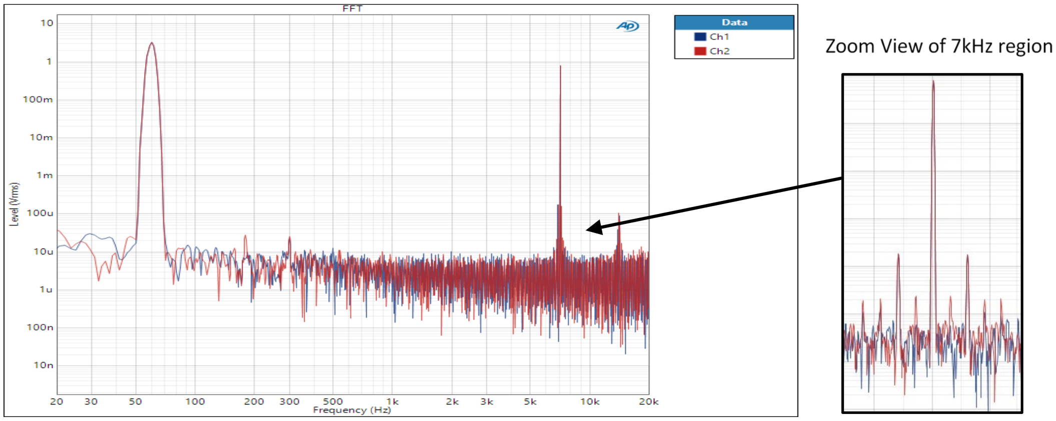
Here are the actual measured values for this test:



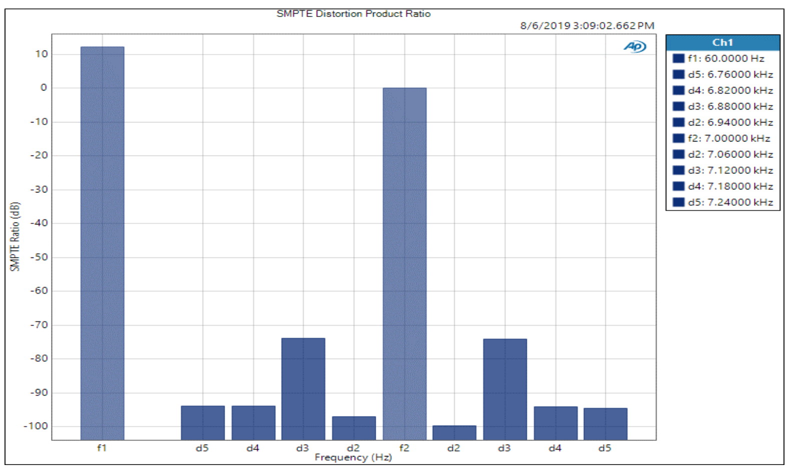
For more extensive analysis, it is possible to sweep the mean frequency or the difference frequency to check for sensitivities in the DUT to specific frequency combinations that might exhibit higher DFD levels.

Another intermodulation test technique is the SMPTE IMD test, which was originally standardized by the Society of Motion Picture and Television Engineers (SMPTE) but is now managed by IEEE. As with the DFD test, this is likely to be done as a part of the design and development cycle rather than production test.

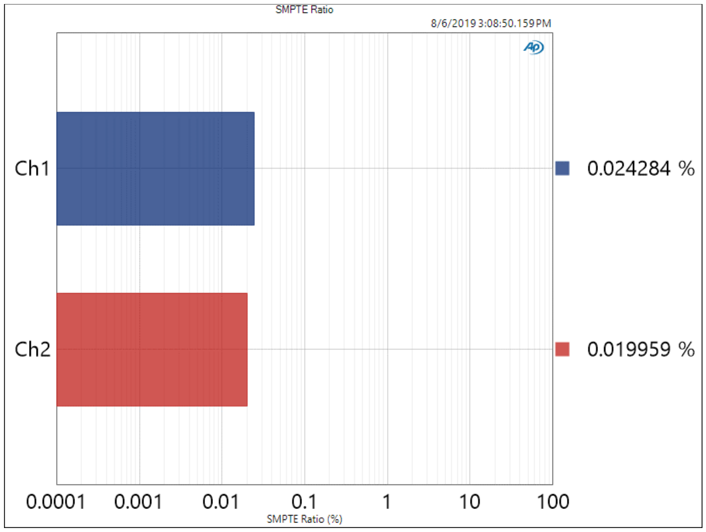
In concept, this is not too different than the DFD test described above; two tones are applied, and the resulting harmonics are measured. However, in this case one tone (f_1) is a strong low-frequency interfering signal, and the typically weaker higher frequency (f_2) represents the signal of interest. A common setting is f_1 at 60 Hz, and f_2 at 7 kHz, with an amplitude ratio of 4:1.



The harmonics of interest are now located near the 7 kHz tone. Any harmonics of the 60 Hz tone that are seen at 120 Hz, 180 Hz, etc., are not relevant to this test. The SMPTE IMD is then determined as the RMS level of IMD products expressed as a ratio to the RMS level of f_2. The following is a representation of the SMPTE IMD test result.



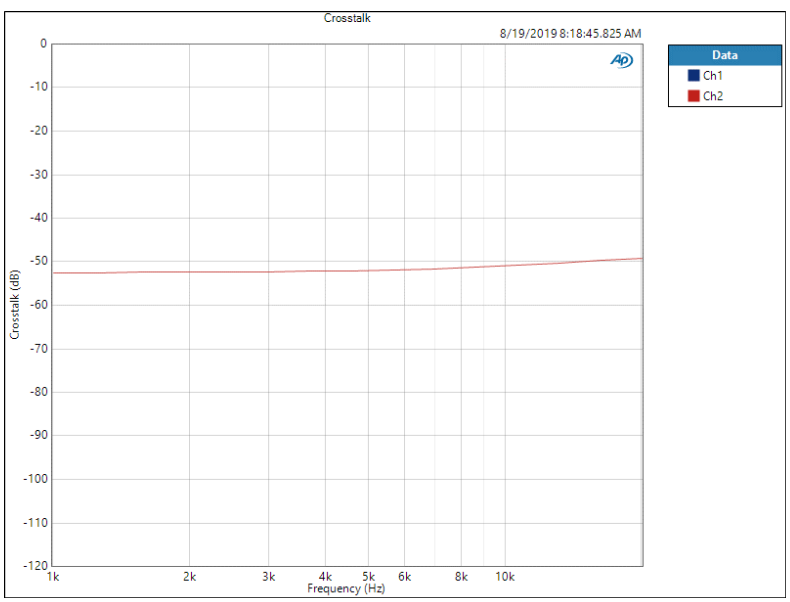
The actual measured values are seen in the following chart. Note that ≤ 0.5% is considered good performance for standard performance products, and ≤ 0.1% is considered good performance for premium performance products. This DUT is showing particularly good performance at about 0.02%, as shown in the following chart.



The APx500 Series analyzers provide for easily sweeping the lower frequency interfering tone to check for IMD sensitivities at various frequency combinations of f_1 and f_2.

Crosstalk

Crosstalk is the unintentional coupling of a signal from one channel to another in a multichannel system. It is generally caused by stray capacitance, inductive coupling, shared power supplies, and shared ground returns. Performing crosstalk measurements helps ensure that signals on one channel do not leak significantly into other channels. Today, performance targets for crosstalk testing limits of automotive amplifiers are in the range of ≤ -40 dB for standard performance products, and ≤ -50 dB for premium performance products. Often the performance is specified to be tested with a 1 watt output level. The following plot shows the performance of one premium performance audio amplifier.



Crosstalk should be measured as a function of frequency, as it commonly varies substantially with frequency. If a single-measurement test is desired, it is commonly measured at 10 KHz. This is for two reasons. First, the

A-Weighting curve indicates that our hearing is attenuated in the range of -10 dB to -12 dB at 20 kHz, compared with 10 kHz attenuation in the range of -3 dB to -5 dB. This suggests that crosstalk effects beyond 10 kHz will be substantially less noticeable. Second, if the crosstalk is caused by a single-pole stray capacitive coupling, and the other impedances are relatively constant, the crosstalk amplitude will increase at a 6 dB per octave rate; assessing at 10 kHz will demonstrate the highest level of crosstalk at the point where it is likely to still be heard.

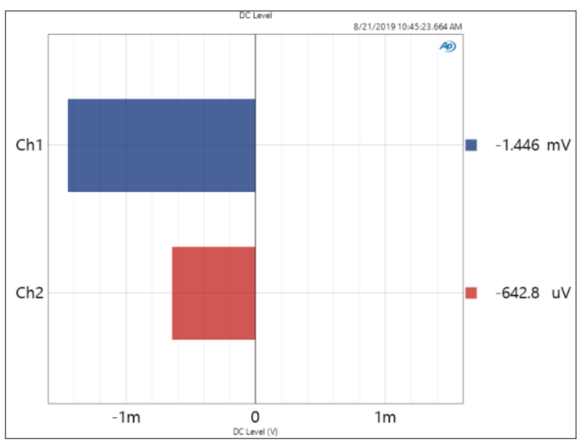
The APx500 audio measurement software provides several crosstalk tests, including single-frequency measurements, stepped-frequency sweep sine measurements, and continuous sweep (chirp) sine measurements. For each of these modes, the user has the choice of driving one channel and measuring the undriven channels or driving all channels except the one that is being measured.

DC Offset, Click and Pop

Amplifiers are normally AC-coupled, meaning there should be no DC voltage at the outputs. A small DC Offset voltage is acceptable, but it must be measured and verified to be within limits. Large DC offsets will affect system performance and can damage components.

The test for DC Offset is related to the Output turn on/off pop-noise test, since the DC offset ramp-up and ramp-down of the power amplifier is a large contributor to how objectionable the sound is.

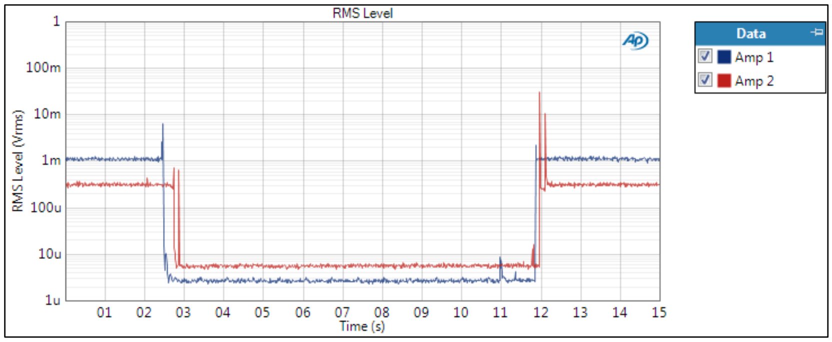
The procedure for testing DC Offset is relatively simple using an audio analyzer. After connecting loads, the amplifier is turned on, and the DC Level measurement is used to measure the DC level. This does not use an AC signal applied to the inputs of the amp. The following plot shows an example of the measurement results for this test.



An absolute value less than or equal to 100 mV is considered good for standard performance products, and an absolute value less than or equal to 10 mV is considered good for a premium performance product.

When amplifiers are turned on or off, transient voltage spikes can occur. Depending on the level and duration of the transient spikes, these may become audible as annoying pops, clicks or thumps. Analysis tools can help determine if transients will be audible, but the final assessment will likely involve listening tests in the vehicle cabin.

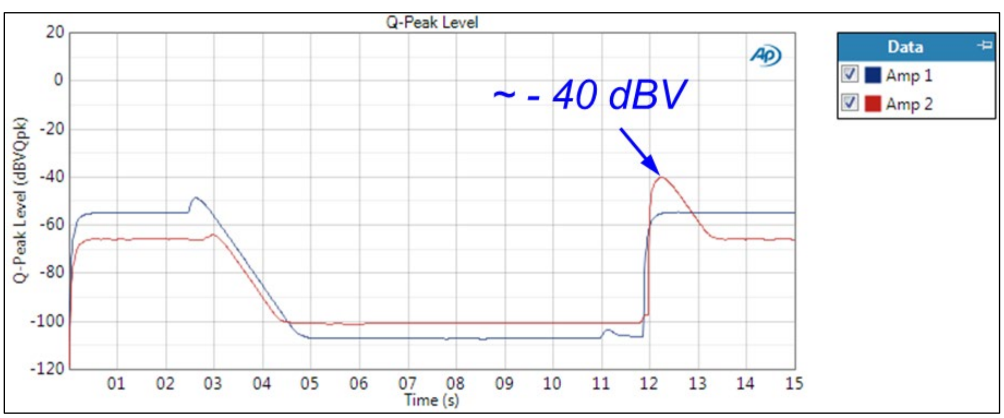
It is relatively easy to connect an audio analyzer to the output of a properly loaded audio amp and acquire a signal that demonstrates if spikes in the signal occur at turn-on or turn-off. Here is an example of such an acquisition using the Measurement Recorder in an APx500 analyzer showing transient spikes occurring at and/or immediately after at the turn-off and turn-on portion of the test.



However, this acquired waveform gives us no sense of how objectionable those spikes might or might not be to the listener. Because perceived loudness depends on the level of the event, the sensitivity of the loudspeaker, and the human perception of sound, these items must be accounted for in the evaluation.

A couple of methods can be used to make a more realistic assessment of these kinds of events using an audio analyzer. If we were making acoustic measurements, we could use a calibrated microphone and an A-weighting filter along with the analyzer to mimic a sound level meter. Unfortunately, this means that we would need to connect a speaker to the amp, and we would likely need to be in an anechoic chamber to get good measurement results. In addition, it has been shown that the human ear responds differently to clicks and bursts of random noise, so the A-weighting filter is less than optimal.

A more appropriate method for analyzing pops and clicks is to use an ITU-R BS.468-4 approach, which includes a CCIR-468 weighting filter, along with a quasi-peak detector with carefully defined attack and decay times that better represents the amplitude of an impulsive noise burst relative to human hearing. APx analyzers provide the ability to apply an ITU-R BS.468-4 Q-peak filter to either an electrical or acoustic measurement. This means that it is possible to measure an electrical signal, and then apply a model for a representative loudspeaker to arrive at a reasonable estimate of the sound pressure level this event represents.



This DUT demonstrates (graph above) a maximum Q-peak level of -40 dBV. From this measurement, we can proceed with the estimation of sound pressure level.

Based on a mid-range loudspeaker (where human hearing is most sensitive), a typical loudspeaker sensitivity would be in the range of 90 dBSPL, based on 1 watt at 1 meter. From this we can do a couple quick calculations that give an indication of the expected sound pressure level with that representative speaker connected to the amp.

With a 4-ohm load, we can calculate the voltage using the formula $\boldsymbol{V} = \sqrt{\boldsymbol{P} * \boldsymbol{R}}$.

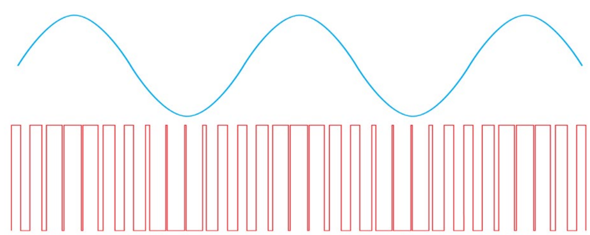
With 1 watt of output power, and a 4-ohm load, we determine that the voltage at 90 dBSPL is 2 Vrms. Since our Q-Peak level measurement is in dBV, and dBV is referenced to 1 V, we need to convert the 90 dBSPL sensitivity to an equivalent level with a reference of 1 V. The dB level calculation as a result of 1 V reference is $\boldsymbol{dB}$ = **20***$\boldsymbol{log}(\frac{\boldsymbol{1V}}{\boldsymbol{2V}})$, which is about -6 dB, resulting in a value of approximately 84 dBSPL. With this value, and the measured ITU-R BS.468-4 Q-peak level, we can determine that the audible level will be 84 dB – 40 dB, resulting in a sound level of 44 dBSPL. In a very quiet environment (e.g., vehicle parked with engine off) a transient with a sound level of 44 dBSPL could be audible.

Though there is no specific maximum dB performance target for click/pop, it is desirable to have no audible click/pop in a quiet environment inside the passenger compartment.

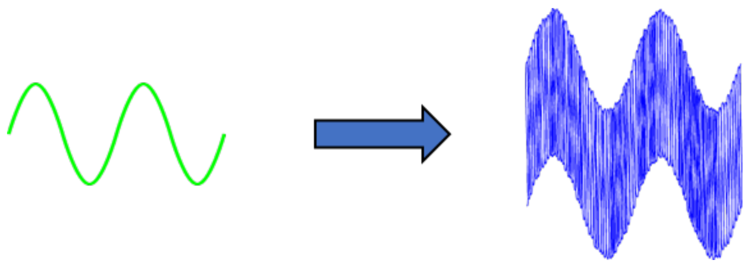
Class D Amplifiers

"Linear" audio amplifiers are Class A or Class AB. These typically have very good fidelity, but relatively low efficiency. For these amps, the active components are constantly dissipating power. Class D amplifiers can have very good fidelity but can also have substantially higher efficiency compared to linear amps. Both are tested using the same tests and techniques described in the previous sections of this paper. However, Class D amps do require a slightly different treatment to get representative test results.

Class D amplifiers (sometimes referred to as switching amplifiers) use pulse width modulation (PWM) or some other type of modulation to maximize the efficiency of the amp. The maximum on-time per pulse occurs at the positive peaks, and the maximum off-time occurs at the negative peaks. At the amplitude mid-point of the sine wave, the duty cycle will be approximately 50%. This is represented in the following diagram.

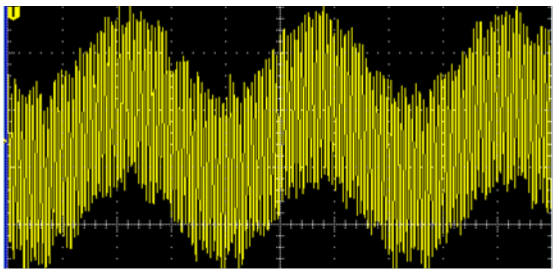


The output signal swings from maximum output to minimum output very quickly (resulting in high frequency content in the output) as can be seen in the following image.

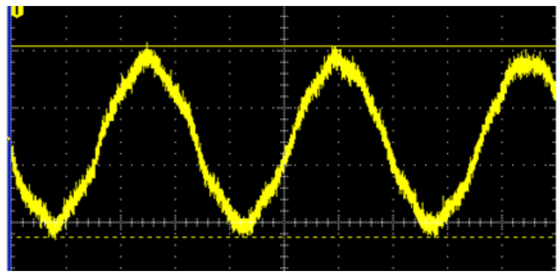


These noise signals have very high slew rates (SR), which can cause erroneous distortion and noise readings in analyzer measurements. In some cases, audio analyzer circuits can be forced into unstable states or even suffer damage.

Fortunately, there is a robust solution for this situation. By applying filtering to the output, the high frequency content can be filtered prior to it arriving at the analyzer for measurement, dramatically reducing the high frequency component. The recommended filtering is done in two stages. The first stage is an external passive filter connected between the amplifier outputs and the analyzer inputs. This eliminates the bulk of the high frequency content before the signal reaches the analyzer. The following signal captures show the pre-filter and post-filter signal from a Class D amp.



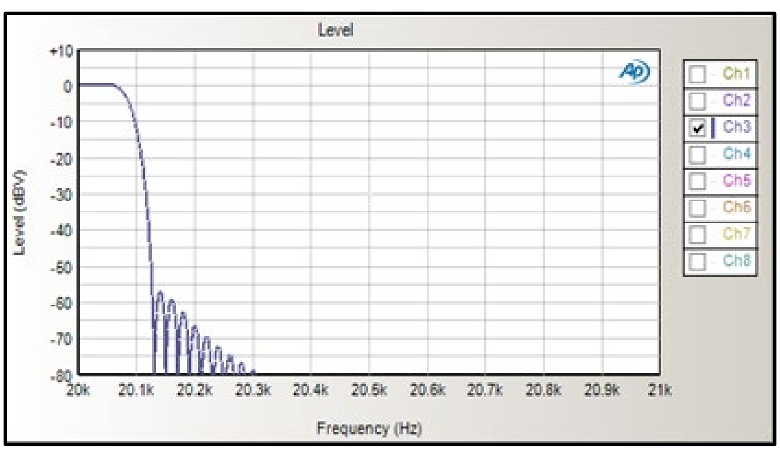
Pre-Filter Output



Post-Filter Output

As an example, AP has three different configurations of external filters that are designed for this purpose. These include the AUX-0025 two-channel 25 kHz low-pass filter, the AUX-0100 eight-channel 25 kHz low-pass filter, and the AUX-0040 two-channel 40 kHz low-pass filter.

The second recommended filtering stage is a sharp roll-off bandwidth-limiting filter in the analyzer. The AES17 standard recommends such a filter for D/A converter measurement, and this AES17 filter is a good option as an internal filter for Class D amplifier measurements. Audio Precision implements this as a brick-wall DSP filter in the APx analyzers. The cutoff response of this filter is shown below.



It is critical that a passive external filter be used in addition to the AES17 filter to assure good measurement quality and avoid potential damage to the analyzer input. The analyzer's internal AES17 filter does nothing to protect the analyzer input stages from effects of the high frequency content of the Class D output.

It is worth noting that some Class D amplifier manufacturers have built-in output filtering, such as the Pioneer Class FD. It is best to know for certain about the nature of the amp under test before making any connections to the test equipment.

Conclusion

This paper has presented the types of measurements commonly made on automotive amplifiers, the typical expected performance levels, some of the industry standards that prescribe specific methods for testing, and features in AP audio analyzers that facilitate making these measurements. Watch for upcoming papers in this series, which will describe the testing of other audio components in today's cars.

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