LIMITATIONS IN MAKING AUDIO BANDWIDTH MEASUREMENTS IN THE PRESENCE OF SIGNIFICANT OUT-OF-BAND NOISE

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August 2005

Introduction

There once was a time (before the 1980s) when the word "audio" referred to an analog signal having virtually all of its energy content below 20 kHz. Any energy above 20 kHz consisted of residual wide-band noise, signal harmonics, and perhaps the unintentional pickup of local radio and TV broadcast stations. Audio analyzers typically provided one or more bandwidth limiting filters to reduce the effects of out-of-band energy on measurement results. These filters were usually 3-pole in design with an 18 dB/octave roll-off. The most common selections were 80 kHz, 30 kHz, and/or 22 kHz; and few problems were ever encountered.

But times and technology have changed! The word "audio" has now come to include signals that can have gross amounts of energy just above the audio band. For example, the noise shaping commonly employed in most sigma-delta D/A converters and the pulse width modulators of many class-D amplifiers gives an exceptionally low noise floor within the 20 kHz audio band; the tradeoff, however, is an out-of-band noise floor that rises very rapidly just above 20 kHz. Out-of-band energy may also include high frequency artifacts and fast slewing transients related to the over-sampling converter or the amplifier modulator.

The traditional 3-pole bandwidth limiting filter is virtually useless for rejecting these forms of out-of-band energy. Indeed, the AES17 standard for testing D/A converters specifies a very sharp low-pass characteristic that can be realized only with an elliptic filter of at least 7th order. This type of filter is much more costly and difficult to implement, and it can contribute higher levels of its own distortion and noise to the measurement. However, regardless of its design, the tacit assumption is made that all stages in the analyzer *prior* to its bandwidth limiting filter will continue to respond linearly to both the audio signal and the undesirable out-of-band components. ***This can be a very wrong assumption!***

Where is the Bandwidth Limiting Filter?

The architecture of a typical high quality audio analyzer consists of the following blocks:

1) A selectable high impedance input attenuator to enable the measurement of large amplitude signals. Typical attenuator step sizes are usually 10 or 12 dB for a total range of 0 dB (no attenuation) up to perhaps 36–40 dB in total attenuation.

2) A high impedance differential input stage having selectable gain. Typical gain increments are usually 5, 6, 10, or 12 dB for a total gain range of 0 dB to 36 dB (or even higher in some designs). The purpose of this stage is to amplify small signals to the point where the noise contributions of following stages become insignificant when compared to the residual noise floor of the input stage.

3) One or more sets of auto-ranging comparators and means for automatically controlling the input attenuator and input stage gain for optimum performance.

4) An analog signal processing stage such as a tunable notch filter, followed by additional gain stages, bandwidth limiting filters, and measurement detectors;

   or

   A high performance A/D converter followed by DSP processing to extract the desired measurements. A/D conversion inherently limits the alias-free measurement bandwidth to <50 % of its sampling rate. Additional bandwidth limiting is implemented via DSP processing.

Regardless of how the last block is implemented, it is important to note that measurement bandwidth limiting occurs at or very near the end of the overall signal path. This has the advantage of limiting residual noise contributions from the analyzer itself. However all stages before this point in the analyzer must still process the full bandwidth input signal, including any out-of-band energy it may contain. As long as the out-of-band energy does not (1) exceed the amplitude of the in-band signal or (2) contain high frequency components that provoke slew rate limiting, the measurement system will behave predictably and linearly.

Auto-Ranging and Measurement Dynamic Range Problems

If the out-of-band energy exceeds the amplitude of the in-band audio signal, the analyzer auto-ranging circuits will no longer pick the same input attenuator and measurement path gain states as if there were no out-of-band energy. The in-band audio signal will be lower than its optimum level within each stage of the measurement path causing degraded residual noise performance and increased error. Depending upon the relative amplitude of the out-of-band noise compared to the in-band signal, the effects can be profound.

For example, suppose an audio analyzer is attempting to measure a 10 mV, 1 kHz signal in the presence of 1 V of out-of-band noise. The input stage auto-ranging circuits will seek to pick the range that optimizes the measurement of the 1 V out-of-band noise, not the 10 mV signal otherwise clipping and gross non-linearity would occur. The result is the in-band signal is approximately 40 dB lower than it would have been if no out-of-band noise was present.

At some point in the signal path the bandwidth limiting filter removes the out-of-band noise leaving just the in-band signal to be passed on to the detector or A/D converter for measurement. In analyzers with analog detectors, a 40 dB lower than normal signal will seriously degrade measurement accuracy. Most analog detectors have only a 40–50 dB useable dynamic range due to quantization and residual dc offsets in their circuits. This is roughly equivalent to reading an old-style galvanic meter where the signal is only 1 % or 1/100 of the full scale of the meter. Although the bandwidth limiting filter has successfully removed the undesired out-of-band energy before being measured, the amplitude of the desired in-band component is now so small in comparison to the full scale of the detector that the resultant measurement is subject to large error. In the extreme, the detector may not even register a reading above 0 (or –999 dB)!

The situation is much better in A/D based analyzers because the DSP implemented "converter" or "detector" has a much broader dynamic range, typically well over 100 dB. However, the converter noise floor, distortion, and spurious contributions will be 40 dB higher compared to the signal amplitude than had there been no out-of-band energy in the first place. Numeric round-off and truncation errors within the DSP algorithms can also lead to increased error when the signal being measured is below its optimum level.

Measurement Path Non-linearity and Input Slew Rate Problems

When the out-of-band signal contains high frequency components beyond the specified bandwidth of the analyzer, the ability of the analyzer input stages to linearly respond to the total signal must be considered. The analog signal paths of all high quality audio and FFT analyzers contain ultra-low distortion operational amplifiers or "op-amps" such as the AD797, OPA627, or even the venerable 5534 to provide buffering, gain, and active filtering. These devices have a typical maximum slew rate limit of 20–50 V/µsec depending upon their compensation. If the undesirable out-of-band energy components cause an op-amp to hit its slew rate limit, the in-band audio signal will no longer be processed in a linear manner. All subsequent measurements will then be subject to potentially gross errors.

For a given op-amp peak slew rate ("SR"), the maximum allowable output amplitude at a given frequency (or the maximum frequency at a given amplitude) can be calculated using the formula:

$$\text{Peak SR} \quad = \quad 2\pi \cdot f \cdot (\text{Vrms} \cdot \sqrt{2}) \quad = \quad 8.89 \cdot f \cdot \text{Vrms}$$

where "SR" is in units of V/µsec and "f" is in megahertz.

For example, the Audio Precision System Two analyzer uses AD797 op-amps in its input stage. The “full-scale” operating voltage at this point is 2.5 Vrms. Given the slew rate of the AD797 is about 20 V/µsec, the maximum full-scale signal frequency cannot exceed about 900 kHz without *hard* slew limiting. Unfortunately the distortion performance of an op-amp degrades long before its slew rate reaches its limiting value. One design “rule-of-thumb” is to avoid signal conditions that push an op-amp beyond 50–60 % of its maximum slew rate. Thus, the maximum rated full-scale frequency should not exceed ≈500 kHz.

An equivalent way to think about this problem is in terms of an input signal slew rate limitation. Since the slew rate of an op-amp always refers to its *output* signal, the maximum allowable *input* signal slew rate will scale in inverse proportion to the gain (or attenuation) of the input stage. Thus the maximum input slew rate will necessarily decrease with increasing sensitivity (decreasing voltage range). Using the Audio Precision System Two again as the example, the following table lists the overall gain and maximum allowable input signal slew rate versus input range setting due to this factor:

Range	**Gain**	**Max Input SR**
160 V	–36 dB	640 V/µsec
80 V	–30 dB	320 V/µsec
40 V	–24 dB	160 V/µsec
20 V	–18 dB	80 V/µsec
10 V	–12 dB	40 V/µsec
5 V	–6 dB	20 V/µsec
2.5 V	0 dB	10 V/µsec
1.25 V	+6 dB	5 V/µsec
600 mV	+12 dB	2.5 V/µsec
300 mV	+18 dB	1.25 V/µsec
160 mV	+24 dB	0.62 V/µsec
80 mV	+30 dB	0.31 V/µsec
40 mV	+36 dB	0.16 V/µsec

This table is illustrative only, and it shows the limitation caused only by the input stage op-amps. The actual situation is somewhat more complex because there are other design factors that can more severely limit the maximum input slew rate, especially in the higher input voltage ranges. Similar tables can be constructed for other models and brands of audio analyzers based upon their choices for input voltage ranges, op-amps, and maximum rated signal bandwidths.

The key point is that *all* audio analyzers exhibit a decreasing ability to tolerate fast-slewing out-of-band components as the input voltage range is decreased. No amount of bandwidth filtering or processing in a later stage can repair the damage to the in-band signal if slew rate limiting has occurred in the input stage.

Input and External Filters

Audio analyzers often contain an internal radio frequency interference ("RFI") filter in series with their input stages. A properly designed RFI filter can provide good rejection above about 5 MHz and prevent input stage demodulation of FM and TV station pickup. RFI filters effectively increase the maximum allowable input signal slew rate at these higher frequencies. Unfortunately that still leaves the input stage susceptible to slew rate problems from out-of-band signals from just above the analyzer's maximum specified bandwidth up through about 5 MHz.

Input or RFI filters can be designed to be effective down to much lower frequencies, but only with performance tradeoffs that are usually unacceptable for general purpose audio analyzers. Such tradeoffs can include a significantly higher input capacitance (or lower input impedance), higher input noise floor, degraded common mode rejection, and/or degraded flatness within the intended measurement bandwidth.

In certain applications where one or more of the above tradeoffs is acceptable, a passive *external* filter can provide the needed rejection of out-of-band energy. External filters can be highly effective because they prevent the offending high frequency energy from entering the analyzer input stage in the first place. The Audio Precision AUX-0025 accessory for testing class-D amplifiers is a good example. It is a dual passive low-pass filter with a 20 kHz usable bandwidth, very steep roll-off above 100 kHz and a stop-band attenuation >50 dB at 250 kHz and higher. However, it also has an input capacitance of 10 nF (10,000 pF).

Testing D/A Converters with Significant Out-of-Band Energy

The nature of D/A out-of-band energy varies considerably from manufacturer to manufacturer, and with different designs. General statements regarding the amplitude and frequency distribution are difficult to make because of the wide range of noise profiles encountered. Some out-of-band signals are basically impulsive in nature, while others show the appearance of sine-bursts of one or more cycles at frequencies of several MHz or higher. Although these signals can be filtered on chip, competitive pressures often force the omission of such filters.

The best tool to view these out-of-band signals is the digital storage oscilloscope. Looking at this signal using a spectrum analyzer can give very misleading indications. Significant impulsive or sine-burst forms of out-of-band signals can appear to be random noise if the time domain position of these artifacts varies in a random or pseudo-random fashion.

Because D/A out-of-band noise tends to be constant and relatively independent of the in-band signal, an interesting situation occurs when it is measured with an auto-ranging audio analyzer. As the amplitude of the in-band audio signal is decreased, the analyzer will switch to progressively more sensitive ranges thus reducing its input slew rate tolerance. Measurements of noise or THD+N will show a sudden jump or increase if the input stage gain switches to the critical point where the analyzer input slew rate capability drops below the slew rate of the out-of-band components. Once this happens, all subsequent analyzer measurements are subject to potentially serious error. One needs to be especially careful when making the Dynamic Range THD+N test at –60 dBFS. Because the signal is so small, the analyzer will usually attempt to pick its most sensitive input range (with the lowest slew rate tolerance) if left in its auto-ranging mode of operation.

For example, a certain D/A has an out-of-band noise component that resembles a 4 MHz sine-burst with an amplitude of about 60 mVpp. The slew rate of this signal is calculated to be 0.75 V/µsec. Using the table of input slew rate capabilities given on Page 4, it is seen that linear analyzer operation can be expected down through the 300 mV range of an Audio Precision System Two before input stage slew rate limiting occurs. Input range settings below 300 mV should be avoiding when making measurements of this D/A.

Testing Class-D Amplifiers

Class-D amplifiers operate by rapidly switching their outputs between two or more supply potentials using pulse width modulation techniques to control the power of the audio band signal. The raw output waveform consists of a series of variable width pulses whose amplitude is determined by the power supply rails. Typical switching frequencies are in the range of 250 kHz to 750 kHz. In the frequency domain, the raw amplifier output contains the desired in-band audio signal plus high frequency artifacts related to the switching frequency and its harmonics. The modulators of many class-D amplifiers also employ noise-shaping techniques to favor low noise within the audio band. The resultant noise floor often resembles that of a sigma-delta D/A converter showing a rapidly rising characteristic just above 20 kHz.

The amplifier's raw output waveform can have an incredibly high slew rate, typically 1000 V/µsec or even higher. It is relatively independent of the actual audio signal itself. As can be seen from the table of input signal slew rate limits on page 4, there is virtually *NO* analyzer range setting that will process this signal without serious non-linearity. Indeed, the slew rate is so high it will attempt to cause potentially damaging 200 mA current spikes to flow into the 200 pF input capacitance of the analyzer. Directly connecting an audio analyzer to the raw output signal of a class-D amplifier is strongly discouraged!

To make valid measurements of class-D amplifiers one must insert a suitable low-pass filter between its output and the analyzer input to attenuate the switching artifacts and their peak slew rate. Some class-D amplifiers contain internal LC filters to minimize radio frequency interference. Although they dramatically reduce the amplitude of the switching artifacts and the corresponding peak slew rates, they may not be adequate to prevent input slew rate problems in all ranges of an audio analyzer. The Audio Precision AUX-0025 accessory referred to earlier was specially designed for this application.

Because most class-D amplifier modulators employ noise shaping, the analyzer's AES17 low-pass filter should also be enabled to reject the effects of rapidly rising noise above

20 kHz. Otherwise, making measurements on class-D amplifiers is very similar to measuring a D/A.

Recommendations & Summary

Audio analyzers are optimized to make high quality measurements up through their maximum specified bandwidths. The presence of significant energy above these bandwidths can cause serious problems with linearity within the input stages of the analyzer and introduce gross measurement errors. The test engineer must carefully consider the nature and amplitude/frequency profiles of out-of-band energy sources when setting up audio measurements. Unfortunately there is no automatic way to sense if out-of-band energy is compromising a measurement by provoking slew rate non-linearity within the audio analyzer.

Whenever possible specify a fixed or minimum input range for analyzer operation when testing devices that have significant out-of-band energy. This prevents the analyzer autoranging feature from picking too sensitive a range where the input slew rate capability can drop below the actual out-of-band signal slew rate. If the slew rate of the out-of-band content is simply too high for any reasonable input range setting, an external low-pass filter (e.g. the AUX-0025) *must* be inserted between the device output and the analyzer input.

Good grounding and signal interconnection practices should also be observed to minimize common mode potentials (which can also have a very high slew rate) between the device under test and the analyzer inputs.

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