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Testing Challenges in Personal Computer Audio Devices

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Abstract

The Personal Computer audio environment has evolved over the years to become a significant entity within the field of acquisition and rendering of audio information. The personal computer is a highly sophisticated interactive environment that is much more complex than a conventional dedicated home audio device, leading to new problem areas. These include, but are not limited to, stochastic interrupts, network accesses, disc I/O and disparate hardware qualities. While the environment of a highly matrixed multi-tasking concurrent operating system offers many opportunities to overcome quality issues, the PC, due to the media-rich tools and feature sets, is becoming the entertainment capture and rendering device of choice for future generations.

Many of the quality issues have been focused on hardware, such as converter quality, power supply quality and component metrics. We will be focusing on software performance metrics which are, by definition, much more difficult to ascertain. We will address conventional audio measurements such as distortion, frequency response and signal-to-noise ratio, but will extend these to new depths and address the unique difficulties the PC environment adds to these tests. The tests will also include "glitch verification", throughput latency, and MIDI latency.

1. Background

Testing the audio performance of a PC audio device or "sound card" can present some unique challenges. Unlike conventional audio devices, all of the inputs and outputs may not be easily accessible—many of the digital signals exist only on high-speed digital buses internally, with no direct access. Although the basic structure of a PC audio device is similar to a conventional tape recorder, there is no "record level meter", making establishment of levels difficult. Finally, the PC environment can introduce new quality problems not experienced in analog audio systems, for example, the integrity of the audio stream. Other demands within the PC can cause signal dropouts or repeats during an audio playback. Conventional audio system tests generally will not detect these errors. We will examine the challenges faced in evaluating PC audio devices, present some techniques to meet these challenges, and present some typical measurements to illustrate problem areas.

1.1 Unique Challenges

From a traditional audio industry perspective, the personal computer may be thought of as a “tape recorder”, with the tape actually being a hard drive or CD-ROM. The PC is able to record audio signals and play these back on demand. With the prevalence of Internet connectivity, program material is easily transported between computers. Although practices for the maintenance of quality of a conventional tape recorder have been well established, these do not easily migrate to the measurement of PC audio devices. One immediate difference is the absence of the conventional record level meter. Establishment of operating and full-scale levels in the PC environment is a necessary but difficult task. We will explore solutions to this problem.

2. What is a PC Audio Device?

We use the phrase *Personal Computer Audio Device* to identify the portion used for acquiring, storing, and rendering audio program material using the sophisticated data handling facilities built into the PC. Such systems started out in the early days of PCs as simple “sound cards” primarily to provide sound effects for games. Early systems were not much more than noisemakers with 8-bit resolution, mostly monaural implementation, and miniature limited-range speakers. The technology has evolved over more than two decades to the point where contemporary PC sound systems are typically 16 to 24 bit resolution, often include surround sound, and exhibit performance levels on a par with high quality home audio systems. Small speakers have been replaced with full-range speaker systems, some even with sub-woofers, resembling those supplied with home stereo systems. In many cases, the PC audio device is connected to an external sound system. This evolution has been enabled by technological advancement, particularly in converters, but is also driven by the increasing use of the PC as an entertainment center. DVD videos, streaming audio, and Internet distribution of coded audio files (MP3, WMA) have all contributed to the development of improving the PC audio facilities. The PC has a lot to offer in ease of content distribution, storage, and organization.

While the technology of PC audio devices enables parity with conventional high quality audio systems, it is not without its potential problems. The PC must share its resources with many other tasks and applications. The electronic environment is inherently hostile and noisy compared to conventional audio components, which are packaged to optimize the rejection of interference sources and other contributors to signal degradation. Much of the development in PC audio has been by designers whose background and training is in areas other than audio, a change from the classic stereotype audio designer who would be typically involved in conventional home audio component development. And finally, the architecture of a PC audio device creates challenges when conventional audio testing practices are attempted. Limited access to internal signal points, wide variations in control mechanisms, and absence of specifications and definitions can make characterization difficult. This paper will examine many of these difficulties and suggest solutions.

2.1 PC audio device Control

Most automated PC audio device testing procedures require that elements of the device be controlled by software. This would include setting various gains by controlling the sliders in the interface. There is much variation in how this control is done, what the control law is and what the resolution of the control is.

2.2 Form factor

A PC audio device is not always a plug-in card—many systems implement the circuitry on the motherboard. However, the architecture is the same; only the package changes. We will not differentiate between plug-in sound cards and on-board designs in this paper since the testing techniques are the same.

Figure 1 is a block diagram of a typical PC audio device. This diagram shows a representative collection of inputs. Any particular system may have more or fewer inputs and outputs. Line In and Line Out are present on virtually all systems with the exception of some notebook computers that have only a microphone input with no line input. Some sound cards include small power amplifiers and therefore have speaker outputs, although most systems generally provide only line level outputs and rely on powered speakers. It is not uncommon today to find 5.1 surround sound outputs as well—rear channel line outputs, center channel outputs and a sub-woofer output. It is also increasingly more common for the better quality cards to include an SPDIF (IEC60958) digital output or even an SPDIF input. Note that the mic and telephone (TAD) inputs are single channel that split to the two stereo channels.

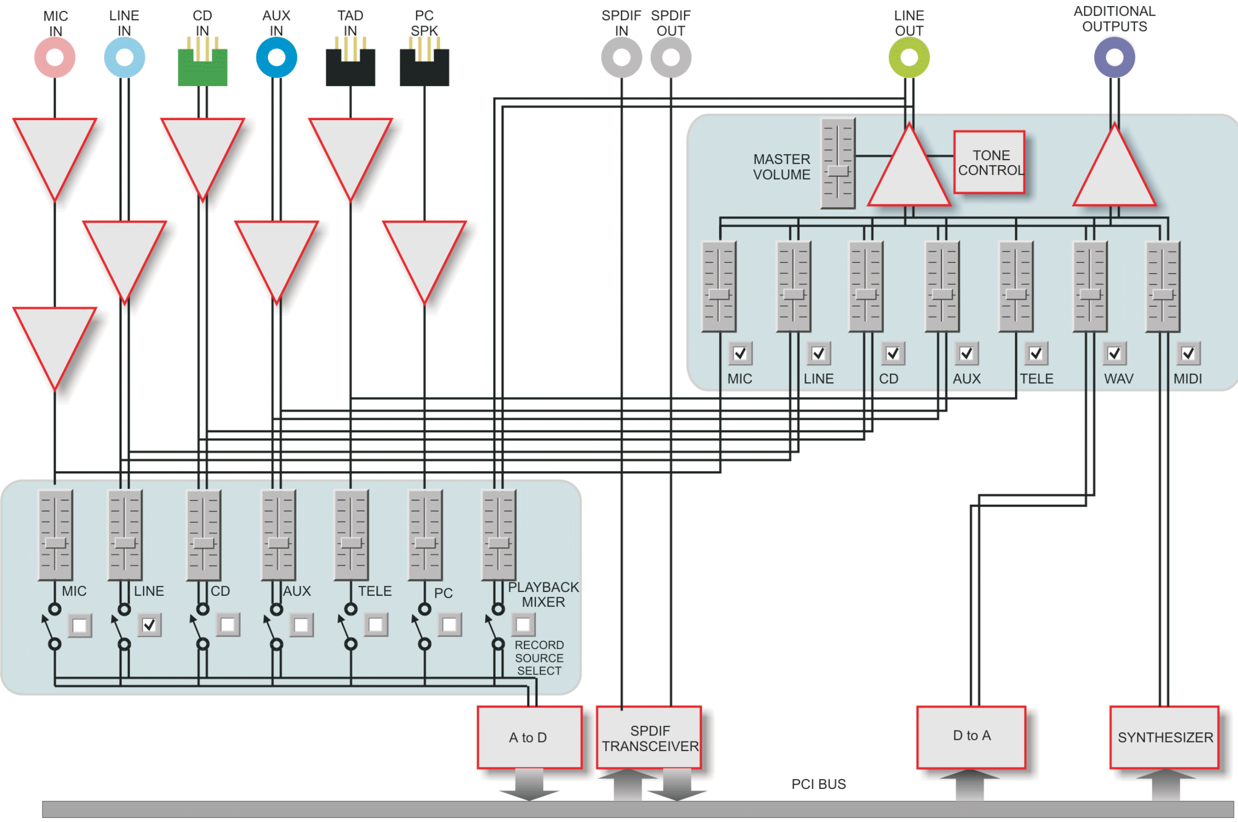

Figure 1 Block diagram of a typical PC audio device

The user control of the audio system is typically by means of a "mixer panel", a series of software slider controls that control the gain of various parts of the circuit. See Figure 2. There is typically a "Record" and a "Playback" panel each with several sliders. In many cases, some sliders on the two panels are tied together or ganged.

The block diagram in Figure 1 shows an equivalent representation of the actual operation but may not be truly representative of actual signal flow. A block diagram such as this is able to accurately represent conventional analog systems constructed from discrete building blocks. Much of the PC audio device is virtual with actual signal flow and gain settings defined by software. Nevertheless, the block diagram is a reasonable representation of equivalent operation.

3 Signal levels

Signal levels and signal handling capability are subjects of high importance to an audio engineer. Any audio system designer must pay careful attention to signal levels throughout the signal path and signal handling capacity of the respective circuitry if good performance is to be attained. This requires attention to clipping or "full scale" level, noise floor, and nominal program level. Any evaluation and testing process must also recognize these issues.

This is an area of difficulty in PC audio devices; there is seldom an adequate indication of signal level at any part of the signal path. The minimum or maximum signal levels for inputs are often not defined. Performance measurements require that the test signal level be chosen for an optimal operation below full scale or "clipping" but as close to this level as possible. Some tests, such as dynamic range, don't require the knowledge of the full-scale level to run the test but do require it for computation of the ratio that is the final stated value. With no accurate indication of level or even an error warning of overload, this is hard to do. Analog tape recorders, predecessors to PC-based digital recorders, always had a VU meter that the operator considered essential to proper operation and the maintenance technician used for testing and adjustment. It would have been unthinkable to use such a recorder without a VU meter; even inexpensive consumer recorders had such a meter, although a less accurate version. But that is what a PC audio device is — a recorder without a VU meter. Exacerbating the problem is a wide variance in nominal signal levels and gain structures among commercially

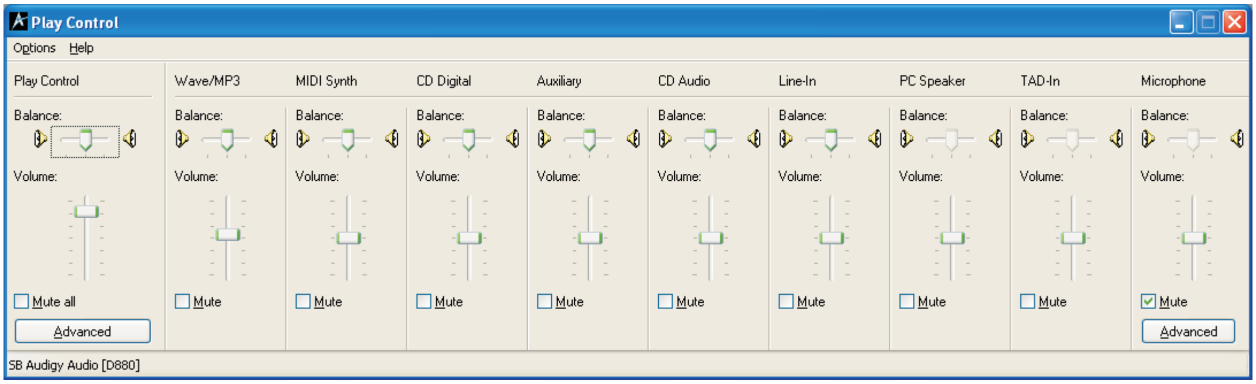

Figure 2 Typical User Interface. The driver associated with the PC audio device will define this mixer panel.

available PC audio devices. Consequently, one of the first steps in characterizing a PC audio device is to establish operating levels and full-scale levels. This can be a tedious and iterative process of trying various levels and looking at the results in the analog domain. A more effective approach is to directly measure the signal level on the PC bus using a software-based digital level meter. This will enable one to determine digital full scale, often expressed as 0 dB FS.

Exceeding 0 dB FS (digital full scale) can have serious audible consequences. Unlike the more gradual compression behavior that is common in conventional analog devices at overload, a program signal that reaches digital full scale produces completely unacceptable audible results (unless for a very short duration). For this reason, many PC audio device manufactures take the precaution of building in a form of peak limiting or compression to prevent the signal from ever attaining absolute 0 dB FS. But this well intended facility adds a burden to the test process by making it difficult to determine what signal level to use for optimal testing. Intuitively, one would determine absolute full scale and then make measurements with this level backed off slightly but still comfortably below the digital overload point to be able to achieve the best dynamic range.

With the use of a digital signal level meter, one can infer the full-scale level by testing at two or more points below full scale but sufficiently above the noise floor. For example, if you find the analog input signal level that will produce a digital signal level of –20 dB FS, then it is easy to predict the analog level that would produce 0 dB FS. But the existence of the limiting or compression algorithms described above will prevent this level from being attained or even approached.

To address such level ambiguities, the AES6id[1] information document and AES17[2] Standard suggest that digital full scale be defined as the level which produces 1% (-40dB) THD+N. In practice, as the signal level approaches the full scale point, the THD+N will sharply rise over a very narrow range near 0 dB FS. The applied signal level that produces a low residual level of THD+N (perhaps –60 dB or lower) and that which produces the magnitude of THD+N that is present at overload or clipping may only differ by 1 dB or less, making determination of the level for –40 dB THD+N a difficult task. For efficiency, the signal level should be rapidly increased in coarse steps to quickly find the approximate full-scale level, and then much finer steps measured close to this value to determine the exact –40dB THD+N signal level.

Another problem that can arise when determining the –40 dB THD+N point is the coarseness of the slider control. On some devices the steps can be as large as 2 dB (increments). They are almost always unpredictable. Although the electronic resolution of a slider level control may be many fractional-dB steps, the actual realizable steps of the control are far less.

4 Standards and Practices

There are several standards and commercial practices that define how tests of PC audio devices should be performed. AES6id specifically addresses testing PC audio devices. AES17 covers digital audio measurements in general and measurements of A-to-D and D-to-A converters in particular.

Intel Corporation and Microsoft Corporation have produced a series of measurement practices to define testing procedures for all hardware used in a PC, including a chapter devoted to testing audio systems. The Microsoft Windows Hardware Quality Labs (WHQL) administers compliance testing based on these documents. The latest document as of this writing is PC2001 [4]. Much of the content of this document and AES 6id is an outgrowth of a white paper written by Dr Steven Harris and Clif Sanchez entitled *Personal Computer Audio Quality Measurements* [3]. We will

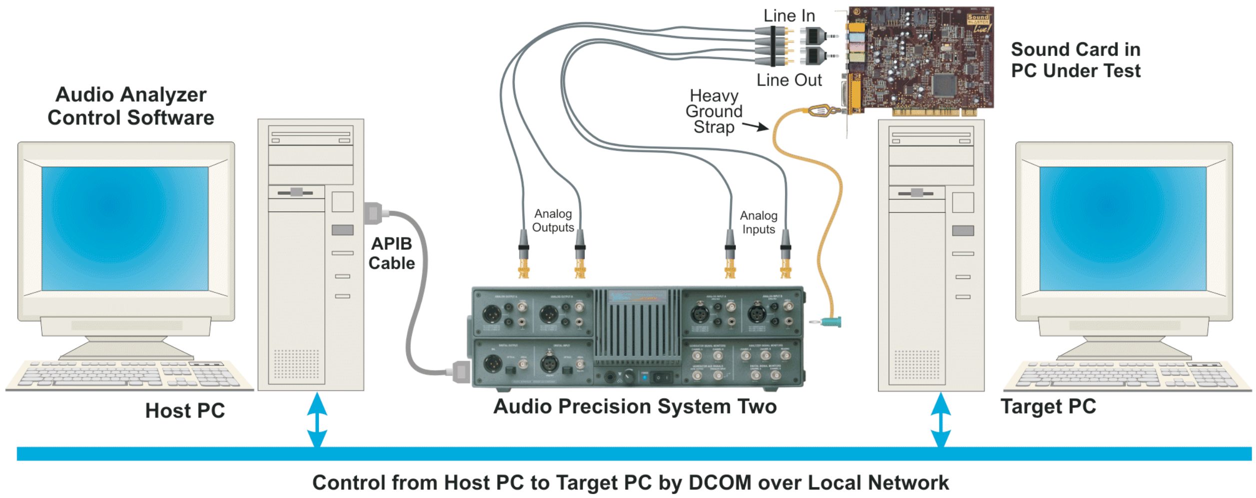

Figure 3 Test Setup. The host computer runs the test software and controls the audio analyzer. The target computer contains the PC audio device and runs a small control application that allows it to communicate with the host computer.

reference the recommendations in these documents in our discussion of test procedures.

5 Test Equipment Setups

Figure 3 shows our typical setup to measure a PC audio device. A Dual Domain audio analyzer is necessary to generate and analyze the analog and digital test signals [8]. The audio system is divided into sections: the D-to-A or Playback section and the A-to-D or Record section.

The analog signal interface to the system is generally via 3.5 mm miniature tip-ring-sleeve phone plugs, however the digital interface is less consistent and presents challenges. The PC does not provide a convenient transmit and receive interface connector to the digital bus that would allow connection of the test instrument digital generator and digital analyzer. There are, however, a number of ways to facilitate this interface. One technique is to use the file playback and record facility of the PC. A test waveform file can be “played” through the audio system. This requires that predefined, digitally created test signals be prepared as .WAV files, available to the PC under test, and loaded as required. Similarly, the A-to-D section can be tested by sending an analog test signal to the line input of the audio system and “record” this signal on the system’s hard drive. This recorded .wav file can be subsequently analyzed by the digital analyzer to complete the measurement process. A more elegant method is to stream the acquired digital signal from the audio system under test directly to the digital audio analyzer, allowing real-time “closed loop” measurements. There may be a temptation to use the SPDIF ports becoming popular on contemporary PC audio systems but our experience shows that these do not pass the digital data intact. For that reason, we developed a method of transferring the digital data directly via the network port.

The analog path is clear in this test setup. The digital path may not be as clear. The audio analyzer is controlled by a host PC that is running the test instrument control software. The PC audio device is located in a target PC. It may be a sound card or an implementation on the motherboard. A PC audio device control program is run on the target PC. The function of this program is to allow control of the audio device, deliver a test signal file, and acquire a signal acquisition file. The host PC and target PC communicate through a local network connection using Windows DCOM (Distributed Component Object Model). This allows the test control software running on the host PC to be able to control the device under test in the target PC, send test signals to the device to be played through the D-to-A playback circuit, and acquire digital signals that have been produced by the record A-to-D path for analysis. The application running on the target PC consumes limited resources, allowing minimal interference to the target application. This target application includes a digital level meter that provides information about the signal level on the digital bus of the target PC to the host test application.

An issue related to test equipment setup that deserves special mention is ground connections. Inadequate grounding of the PC under test to the audio analyzer can seriously degrade the measurement residual. The small 3.5 mm plugs do not provide adequate signal common back to the analyzer. Experience has shown that a heavy supplementary ground wire tightly bonding the chassis of the PC close to the card to a good ground connection at the analyzer can lower noise pickup by several dB. To be effective, this supplementary ground cable should be at least 12 gauge and have fastening means appropriate to the test set up. This would typically be a large spring loaded clip at the PC end to allow attachment to the PC chassis and a banana plug at the other end to allow connection to a good ground reference at the analyzer. Additionally, it is good practice to power the PC under test from the same mains outlet as the analyzer. Different mains circuits can show significant ground difference potential that will leak into the audio signal as hum and buzz. While the strong supplementary ground cable helps this problem, it is better that it not be there in the first place. Figure 4 illustrates the effect of adding a supplemental ground connection.

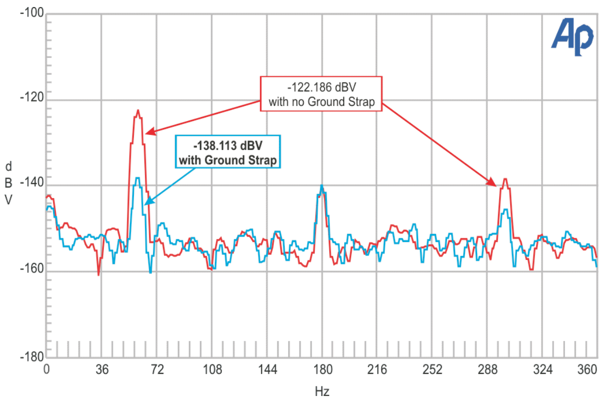

Figure 4 Spectrum analysis of noise with and without supplementary ground connection showing increase in mains hum level.

In a well-designed PC audio device, the converters should be the weakest link in the chain. If the supporting analog circuitry is designed with good engineering practice, its performance will exceed the performance of the converter. Unfortunately, this is not always the case. There are examples of PC audio devices with excellent high-resolution converters that do not deliver the full performance of the converters because of design flaws in the analog input or output circuitry.

To be able to fully characterize a high performance PC audio device, a true Dual Domain analyzer must be used. This means that the analog signal generation must be done by an analog generator, not a D-to-A based generator. Otherwise, it is hard to tell which converter is dominating the measurement—the test instrument generator D-to-A or the PC audio device A-to-D. Today's highest performance analog generators still outperform the best D-to-As by a healthy margin. Similarly, the analyzer must be a true analog design, not an A-to-D followed by a DSP analysis system. While such systems offer other advantages such as speed and lower cost, they are still limited by the performance of the A-to-D, and are below the performance of the best analog analyzers.

Of course, the same argument can be made for a self-contained measurement system. It is true that a PC audio device with proper software can be used as an inexpensive audio generator and analyzer. But again, its measurement performance and accuracy are only as good as the converters used. And if measurements show failures, it is not possible to locate the source of failure.

5.1 Surround Sound

As mentioned earlier, many of the higher performance cards now include surround sound capability with decoders for Dolby Digital (AC3) and DTS 5.1 formats. The testing of these decoders is generally managed by the developers of these codec algorithms and will not be discussed in this paper. However, the general interconnection principles, reference level setting techniques, and performance measurement methods discussed here apply.

5.2 Measurement paths

We discussed the two primary measurement paths, playback and record, above. To characterize a PC audio device, there are actually five variants of these two paths that should be tested. Note that not all paths are accessible on all PC Audio devices.

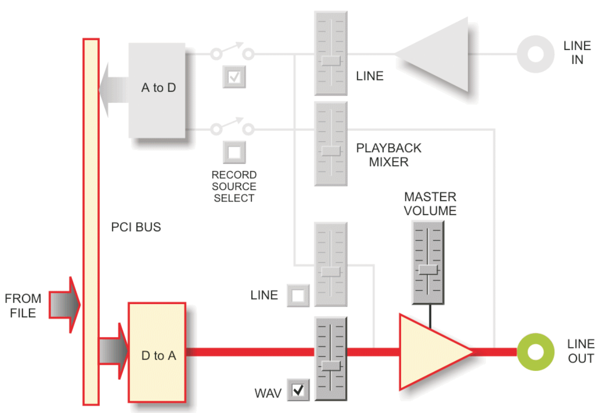

Figure 5 a PC to DA Playback Path

1) PC to DA (Playback path): This is a measurement of the D-to-A and analog output circuitry. It uses a digitally generated test signal, typically a .wav file and provides an analog output signal at line out. See Figure 5a.

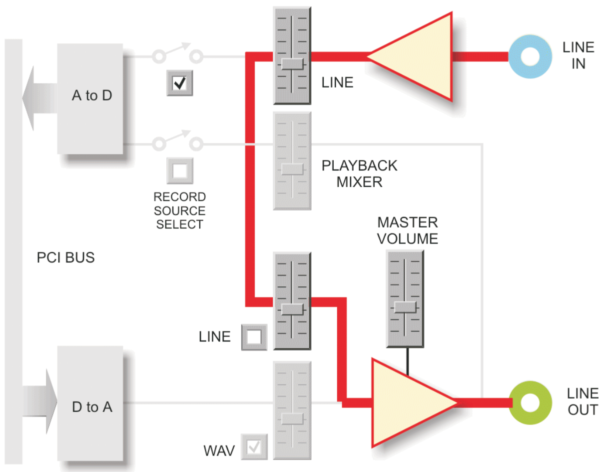

Figure 5b Analog loop-through path (mixer) - Line In to Line Out

2) A-A: Analog Mixer loop. This path measures only the analog circuitry without going through any converters. The mixer controls are set to allow line in (or mic in) to connect to line out. See Figure 5b.

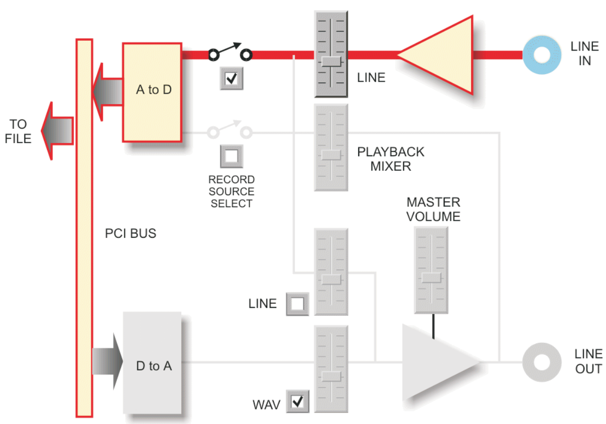

Figure 5c AD to PC (Record) Path

3) AD to PC (Record path): This is a measurement of the A-to-D and analog input circuitry. It uses an analog signal and generates a digital test signal, typically a .wav file. See Figure 5c.

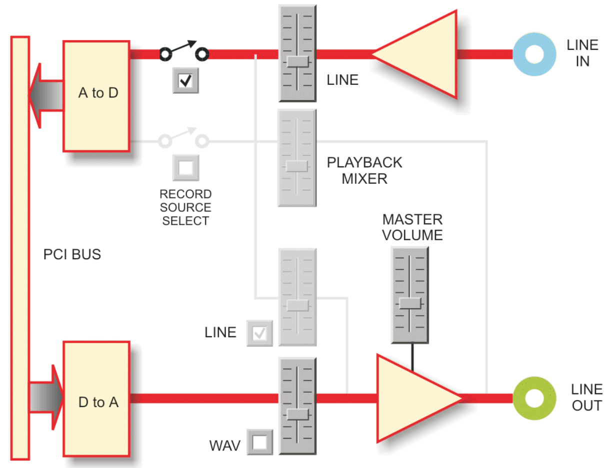

Figure 5d AD to PC to DA (Record-Play Converter loop) Path

4) AD to PC to DA: Analog measurement of Record/Play loop. This is a digital loop through connection that includes all of the elements. The analog test signal is sent to line in, flows through the A-to-D to the PCI bus, is recorded on the hard drive, and then played back through the D-to-A and out the analog line out. Alternatively, in full duplex systems, the digital output of the A to D is directly "looped through" to the digital input of the D to A. See Figure 5d.

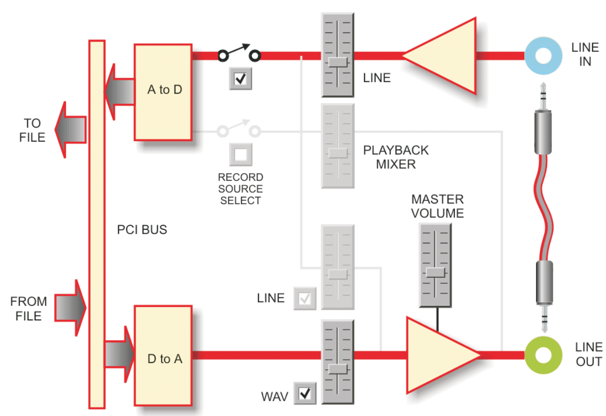

Figure 5e PC to DA to AD to PC (Record/Play loop) Path

5) PC to DA to AD to PC: Digital measurement of Record/Play loop. This is a loop through but measured on the digital side. A digital test signal, typically a wav file, is played through the D-to-A, the line out is connected to the line in, and the

signal is digitized again by the A-to-D and sent to the PCI bus where it is analyzed. This requires a system able to handle simultaneous record/play such as possible on a full duplex system. See Figure 5e.

All five of these signal path permutations are defined in the PC2001 requirements and are required to fully characterize a PC audio device. The loop through configurations do not allow separation of the record and play paths. Some problems may be masked by reciprocal behavior. It is difficult to assess parameters such as dynamic range and THD+N as the composite path can add 3 dB to the results if each individual path has equal noise. For example, a dynamic range measurement of 80 dB for the D to A path and separate 80 dB measurement for the A to D path will yield a 77 dB composite measurement for the loop through paths.

5.3 Analog Inputs and Outputs

It is useful to identify some specific characteristics of the various analog inputs and outputs. There are no definitive standards defining the characteristics of these ports but there are various guidelines in the references cited.

Microphone input. This input is usually (single channel) monophonic, feeding both stereo outputs equally. It is designed to operate with a typical “PC microphone”. Although the connector is a tip-ring-sleeve, the signal is only on the tip. The ring terminal is used to provide a nominal 5.5 V bias voltage to operate an electret microphone. Mic input circuits should be designed to provide a minimal load on a low impedance microphone. Typical input impedances vary from 2k to 4k ohms. PC2001 recommends a full-scale input sensitivity of 100 mV but adjustable gain should be able to accommodate microphones with signals down to 10 mV. The input should be AC coupled.

Line inputs. Most systems (with the exception of some notebook computers) will include at least one line input connector. This is generally stereo, with the left channel on the tip and right channel on the ring of a miniature phone jack. There is a wide variation of nominal operating signal levels for line inputs; typical levels may be within the range of a low of –30 dBV to a high of +6 dBV. Typical home stereo equipment, such as audio cassette and CD players, have “line” levels in the –10 dBV or 300 mV range. The input impedance of the line input should match typical home stereo line impedances and should not load a medium impedance source. This would suggest an ideal load impedance in the 40kΩ or more range but typical systems will show an impedance as low as 10k Ω.

Additional inputs. PC audio devices, particularly plug in sound cards, may provide one to several additional line level inputs. There will almost always be a CD input designed to handle the signal from a PC CD-ROM drive. Some systems will add a second CD input for a DVD drive. There may be an AUX input, which is simply a second line input. All of these additional inputs will typically have the same characteristics as the default line in. Additional line level inputs that differ from the audio line inputs include a TAD input to accept signal from a voice modem used for a PC based telephone answering system and a PC speaker input to accept the event beeps and sounds fed to the internal speaker. These are mono inputs split to both channels.

Line Output: Like the line input, a line output is almost always included. This is designed to drive headphones or externally amplified speakers, or to connect to a stereo entertainment system. The output source impedance should be low (< 100 ohms) so it is unaffected by typical medium impedance loads or the capacitance of long cables. Like the line input, the line output levels should deliver approximately –10 dBV (300 mV) with a full scale digital signal to be compatible with home stereo equipment.

Additional outputs: If the PC audio device includes support for Surround Sound, additional line outputs will be included. There will typically be a stereo center channel output and a single channel sub-woofer output.

Speaker outputs: It has been popular in earlier sound cards to provide a stereo low power speaker output to drive small external non-powered speakers, although this is less common with contemporary PC audio devices. But if a speaker-level output is provided, additional power amplifier tests should be performed such as power drive capability, power bandwidth, and distortion at rated power and into rated load. The typical output and load impedance for a speaker is 8 Ω. Headphones have impedances in the 16 to 92 Ω range with 32Ω being typical.

6. Conventional Audio Performance Tests

Any comprehensive quality assessment should include, as a minimum, the common suite of conventional audio measurements: frequency response, noise, and distortion. Noise measurements are often misunderstood or improperly reported. Various practices have been followed involving noise bandwidth, weighting, and detection characteristics. We examine eash of these as they apply to the architecture of the typical PC audio device. Distortion measurements also suffer from the same variability and misuse. The problems of characterizing distortion in a band-limited system are addressed. Comparisons of various single-tone, dual-tone and multi-tone [14] distortion measurement techniques are explored. Interchannel phase and crosstalk complete the suite of conventional system measurements. We will also review additional tests [15] and their benefits.

6.1 Typical tests

To completely characterize a PC audio device, a suite of tests should be performed under standardized test conditions. Many of these tests are similar to those done for conventional audio components but perhaps with additional techniques to overcome the limitations of the PC audio device architecture. Additionally, there are some specialized tests that should be added in order to characterize the unique circumstances presented in a PC audio device. Of course, all tests must be performed for both channels of a stereo pair and for all measurement paths.

6.2 Frequency Response

Perhaps the most common audio test, this measurement verifies that the complete audio band is processed equally. This measurement is sometimes called “flatness”, although that term is more commonly used to describe a generator performance.

Frequency response is typically measured at 20 dB below full scale to insure that any compression or overload issues do not affect the response.

A PC audio device is typically specified over the 20 Hz to 20 kHz audio band so this is the minimum test range.

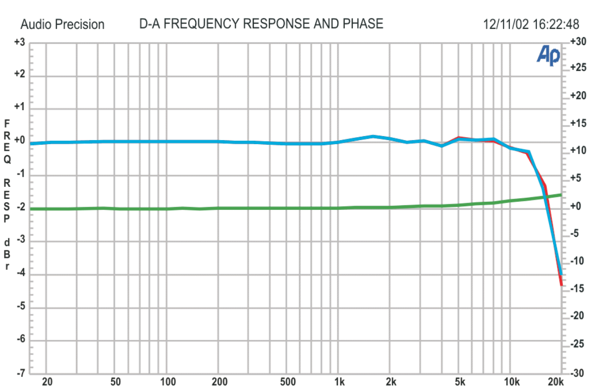

Figure 6 Typical Frequency and Phase Response

The system will likely exhibit a steep high frequency roll-off beyond 20 kHz caused by the reconstruction filters following the D-to-A. For completeness, it may be desirable to extend the frequency response measurement above 20 kHz to evaluate the roll-off characteristics of these filters.

Low frequency response is often dictated by the size of coupling capacitors used. If these are too low a value, the response will not be flat down to 20 Hz.

The quantity of test frequencies used will define the resolution of the frequency response. This is often a trade off against a test time budget. To completely characterize an audio system, a minimum of a spot frequency at octave intervals should be used in addition to test frequencies at the band edge of 20 Hz and 20 kHz. The set of test frequencies should also include 1 kHz (or 997 Hz) to establish a reference. Thus, a minimum practical quantity of test frequencies is approximately 12, although this will usually produce insufficient resolution at the high and low frequencies to completely characterize the response. A better resolution would be third-octave intervals, suggesting 30 or more points or additional points at the high and low ends.

A frequency response graph is a well-understood, familiar presentation. See Figure 6. An alternative way to present the result is a simple reported number of ± X dB relative to 1 kHz, from 20 Hz to 20 kHz. Another useful graph is gain difference between the two channels of a stereo pair. This information can be derived from the individual channel frequency response data.

PC2001 recommends a performance limit for frequency response of line inputs and outputs as no greater than 3 dB down (relative to 997 Hz) from 20 Hz to 17.6 kHz with a 44.1 kHz sample rate and 20 Hz to 19.2 kHz with a 48 kHz sample rate. Accurate assessment of this requirement will likely require more than a 30-point sweep.

6.3 Noise

There are a variety of ways to measure and specify noise in a system, and several conditions of measurement that will have a significant effect on the result. Historically, it has been popular to use the term Signal-to-Noise-Ratio or “SNR” when characterizing conventional analog home stereo components, particularly tape and cassette recorders. As its name implies, this measurement is a ratio of the nominal signal level to the noise floor. There are standards or common practices for specifying the reference signal level. For example, in tape and cassette recorders, it is the signal level that produces 3% third harmonic. For professional mixing and processing equipment, the reference level is that which produces "0 VU" on the internal level meter. This level will typically correspond to +4 or +8 dBu output. Such equipment will also have a defined maximum or clipping level that would typically be 10 to 20 dB above this nominal level, this difference being defined as the "headroom". While these noise measurement techniques are familiar and commonplace to audio engineers, they are not appropriate for noise characterization of a PC audio device that uses A-to-D and D-to-A converters. For this reason, the AES6id document specifies that "Dynamic Range" be measured and specified to characterize noise. Dynamic range is a measure of the difference between "full scale" and the noise floor.

The measurement bandwidth and response characteristics of the measurement instrument will also substantially affect the reported result. While noise signal levels are being measured, it is not appropriate to simply use a wide band level meter to determine noise levels. The measurement bandwidth must be limited to the audio band and may also need to have frequency weighting if it is to conform to a particular measurement standard or recommendation. An analog circuit would typically exhibit a flat noise spectrum from sub-audio frequencies to well above the audio band. Thus, the noise power measured would usually be proportional to measurement bandwidth. In this case, the only way to compare noise performance of different devices is to measure their noise over the same bandwidth.

6.3.1 Noise Weighting

There is another factor to consider when measuring noise. Generally, noise will be at a level far below the level of program material. The frequency response of the ear at low level signals is very poor at high and low frequencies, that is we don't perceive frequencies beyond the mid range of the audio band very well at low sound pressure levels [13]. This argues for the application of frequency weighting to the noise measurement so that the reading will better correlate with the perceived annoyance of the noise. For this reason, most noise measurements are "weighted" using a standardized frequency-weighting curve. Many years back, ANSI specified three weighting curves, A, B, and C, to be used when characterizing audible noise. The A-weighting curve, with the highest roll off of the three, was adopted by the audio industry for measuring noise in electronic equipment. It was particularly popular to characterize noise in tape and cassette recorders.

Later studies in the UK and Europe developed a different weighting curve proposed by its developers to be more appropriate for measuring electronic noise. This weighting is specified in standard CCIR-468 [12], later adopted by ITU-R-468. The CCIR-468 noise measurement standard used a filter that has a 12 dB peak at 6.3 kHz and calls for a quasi-peak detector. A subsequent recommendation by Dolby Laboratories [9] suggested that the 12 dB peak could cause problems with crest factors of some noise signals. To overcome this problem, the gain of the filter is reduced approximately 6 dB by specifying the unity gain point at 2 kHz rather than 1 kHz. Additionally, the detector has been changed to be RMS, a more commonly available choice. This has been adopted as common practice by the industry and designated CCIR-RMS. Figure 7 shows the frequency response of both weighting curves. Although they have quite different response shapes, they both attenuate high and low audio frequencies.

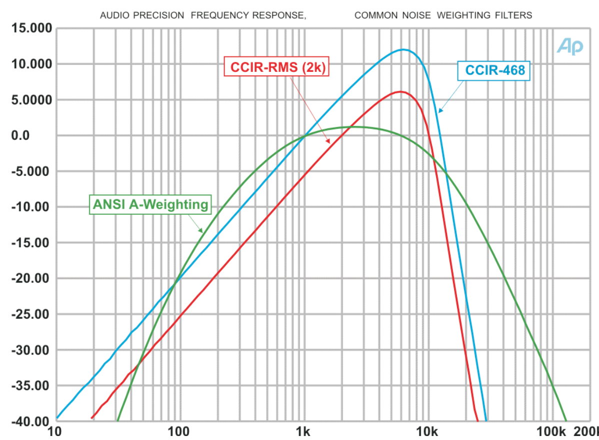

Figure 7 Common Noise Weighting Curves: ANSI/IEC "A-Weighting, CCIR-468, and CCIR-RMS.

6.3.2 Reference level

In an ideal system, the "full scale" reference level would be specified simply as 0 dBFS or digital full scale, that is the maximum value possible of the digital word. In practice, it is nearly always impossible to achieve this value in typical audio systems. Because signal overload beyond digital full scale produces such disastrous audible results, systems limit the ability of the signal to approach this value. For this reason, AES17 and AES6id specify "full scale" as the lower of either digital full scale, or if not achievable, 0.5 dB below the level that produces 1% (-40 dB) THD on the digital side.

6.3.3 Noise floor measurement.

With conventional analog circuits, noise floor is measured by simply terminating the input of the device and measuring the output. Digital and cross-domain systems require additional effort. If the analog input is terminated, automatic muting circuits will usually set the digital output to digital zero giving an excellent but false reading. The circuit needs some signal to keep the channel open. The recommended way to do this is to stimulate the circuit with a low level signal and measure the THD+N at the output. With this low a fundamental signal, the distortion components will be below the noise and the notch filter employed in the THD+N process will virtually eliminate the fundamental component. Thus the "N" in THD+N will dominate. The recommended signal level is 60 dB below the established reference level. Thus, the noise floor is then the THD+N reading plus –60 dB, assuming a conventional ratiometric distortion reading. Some analyzers can present the THD+N as an absolute reading relative to full scale and thus indicate the noise directly.

To summarize, "noise" is specified as Dynamic Range. Dynamic Range is measured by first establishing the reference full scale, the lower of digital full-scale or 0.5 dB below the level that produces –40 dB THD. Then a THD+N reading is taken with the fundamental test signal 60 dB below this reference and an A-Weighting or CCIR-RMS filter selected. If the THD+N reading obtained at this time is for example –30 dB, then the Dynamic Range is –90 dB FS A-weighted.

PC 2001 recommends performance limits for dynamic range for various paths in PC audio devices. The playback path (PC to DA) should be no worse than 80 dB, the record path (AD to PC) no worse than 70 dB. These are typically specified as –80 dB FS and –70 dB FS respectively.

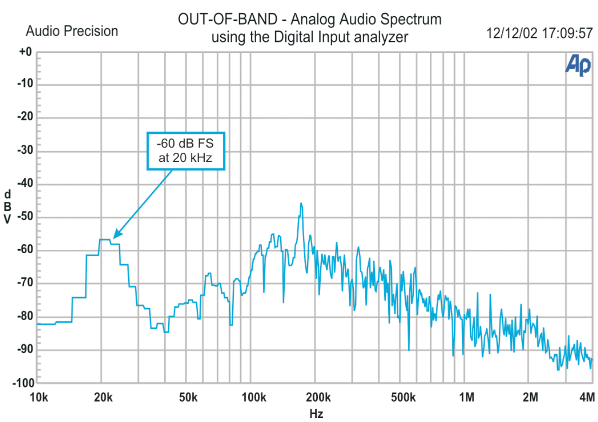

Figure 8 Spectrum analysis of the noise output of an over-sampling D to A converter showing large noise signals above 20 kHz and well out to 4 MHz.

6.3.4 Additional Noise Measurements.

We have described the recommended way of measuring dynamic range according to the standards. However, for analytical reasons, it may be desirable to make additional noise measurements. While we indicated that a weighted noise measurement was more relevant to perceived noise and fulfilled the requirements of the standards, un-weighted band-limited and wide band measurements can provide additional information. Figure 8 shows a very wide band spectrum analysis illustrating out-of-band noise. There are two common sources of noise that can be masked by using frequency weighting. Low frequency hum and buzz caused by mains conduction or induction or by poor power supply design can escape detection if a weighting filter is used because of the approximately 30 dB attenuation the filter introduces at the mains frequency. Most audio analyzers include a switchable 400 Hz high pass filter. If a flat noise measurement is made, this filter can be turned on and off to see the difference. In a circuit with no low frequency hum or buzz, there will be very little change (< 0.5 dB) in the noise reading with and without this high pass filter. If you see a significant change, then suspect an interconnection problem, insufficient grounding, poor shielding, or poor power supply design.

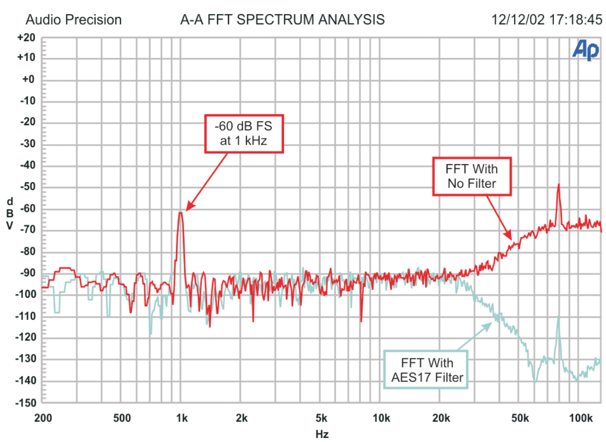

Figure 9 Spectrum analysis of an over-sampling D to A converter with and without the use of an AES17 filter.

At the high end of the spectrum, out-of-band noise produced by the noise shaping characteristics of over-sampling "sigma-delta" converters will also be masked by weighting filters. Most contemporary converter designs use this over-sampling topology. In a development environment, it is desirable to know the characteristics and magnitude of the out-of-band noise. Too much of it can cause problems with subsequent circuits. AES17 specifies a sharp low pass filter with defined characteristics be used when measuring digital circuits for this reason. Figure 9 illustrates the effect of adding an AES17 filter.

So although it is popular to express dynamic range or noise as a single number, for even greater information, consider a spectrum analysis of the noise floor. This will show exactly where the noise contributions are. Mains frequency and its harmonics will be clearly visible. Out-of-band noise, spurious components, and in-band crosstalk will all show up clearly with a spectrum analysis.

6.3.5 Noise stimulated by events

In a conventional audio component, the noise performance is generally stable for a given setup. But in a PC, the audio resources are shared with many other data needs. One cannot assume that the noise performance of the system will remain constant while other events utilize shared resources. A thorough analysis of noise performance should include monitoring the noise floor while exercising various other devices in the system such as the hard drive, network activity, and CPU intensive activity. Such activity may provoke transient spurious noise artifacts.

6.4 Phase

The relative phase response of the two stereo audio channels will have a significant impact on stereo localization and imaging. Stereo recordings are made with the assumption that there will be little phase difference between the left and right channel of the complete record/play path. Since the primary localization cue of a sound source to the ear is the relative arrival times, any phase shift with its consequent time shift will cause this perception to change, thus degrading the stereo image. Anti-aliasing filters are present before all A-to-D converters and reconstruction filters after all D-to-A converters. Since their turnover frequency is close to the upper band edge of the audio band, analog reconstruction filters will introduce phase shift at the upper frequencies. The absolute phase shift caused by these filters is not the problem, but the relative difference between the left and right filter is a problem. For minimal phase difference between the channels, precision components must be used in the design of these filters.

In addition to the phase contribution of analog filters, there are digital effects that can cause relative shifts. The serial digital data steam interleaves the data of the left and right channels in a single data stream. The information of the separate channels is reconstructed during digital to analog conversion. With proper buffering and timing, the resulting analog output will have the same time synchronization as the original stereo signal. Some sound cards have been noted to have a single sample offset in one of the channels. This of course will have a fixed time, variable phase shift versus frequency, with serious stereo imaging effects. Figure 10 shows relative phase versus frequency for these cards. Note the approximately 140-degree phase error at 20 kHz.

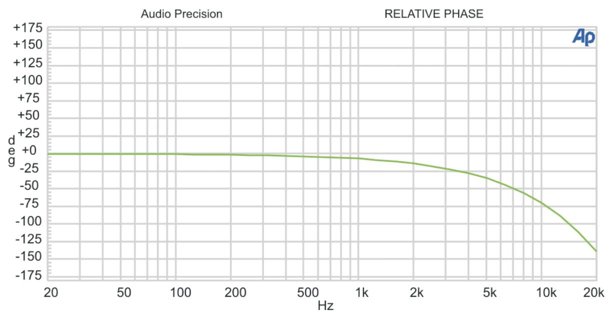

Figure 10 Phase response of a audio device with a one-sample offset of the second channel.

A special case of interchannel phase is Polarity. Inadvertent polarity inversion of one channel with respect to another will have disastrous results with stereo imaging and monaural compatibility. A phase response test will show a polarity inversion by a 180-degree shift at mid-band frequencies.

6.5 Crosstalk

Crosstalk is the undesired leakage of one or more channels into other channels. The most commonly measured parameter is the crosstalk of the left channel into the right and the right channel into the left. This is also sometimes called “separation”. A poor crosstalk figure will also degrade the stereo image. Crosstalk can also happen between the input and output or output and

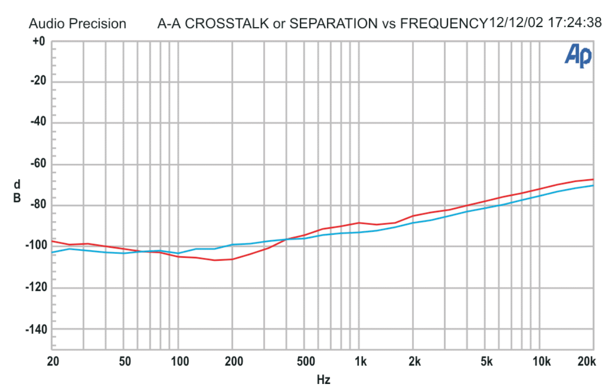

Figure 11 Interchannel crosstalk: left into right, right into left.

input. In this case, there may be unrelated signals present, and poor crosstalk performance will show up as the unwanted signal being audible in the channel being listened to.

Crosstalk is measured by stimulating the driven channel with a test signal and measuring the amount of this signal present in the un-driven channels. The measurement must be narrow band to avoid broadband noise from obscuring the reading. It is common to use a one third octave band pass filter on the measuring meter tracking the frequency of the generator providing the stimulus.

Crosstalk is typically caused by capacitive coupling between adjacent channels. Careful attention to circuit board layout and the addition of a ground plane or shielding can reduce this capacitive effect. Poor grounding practice can cause low frequency crosstalk. Circuit commons returned to a reference ground by a long or shared path can introduce this effect. Use heavy traces, ground plane, and individual paths to a reference ground in circuit layout for optimal low frequency crosstalk performance. See Figure 11 for a typical crosstalk versus frequency graph. The gradual rise at higher frequencies is normal and due to capacitive coupling between the two channels.

PC 2001 recommends a performance limit for crosstalk as equal or greater than 60 dB at 10 kHz.

6.6 Distortion

Distortion is the creation of new signal components by the circuit that were not present in the input signal. Harmonic distortion is the creation of components harmonically related to the input signal. Intermodulation distortion is the creation of new components caused by interaction between two or more input signals. Typically, two signals will "mix" (in the RF sense of the word) producing sum and difference components.

Distortion can be caused by a variety of problems including poor circuit design, slew rate limitations of amplifiers, and non-linear behavior of passive and active components. Different types and degrees of distortion will have varying degrees of "annoyance" value. For example, even order harmonics will be significantly less objectionable than high-order odd harmonics or intermodulation distortion (IMD).

Distortion is measured in PC audio devices with a test signal level of –3 dB FS. Again, this is relative to the determined effective full-scale value as defined.

6.6.1 Band-limited devices

Digital audio systems by their very nature are band-limited. Their upper cutoff frequency, determined by the sample rate in use, is generally very sharp at the upper edge of the audio band—20 kHz or slightly more. Such systems make distortion measurement at high frequencies difficult. Harmonic distortion measurements beyond one half of the upper band edge will be meaningless since all of the harmonics are rolled off by the reconstruction filters. In fact, one might argue that measurements with fundamentals higher than perhaps 20 to 30% of the cutoff are of little value. For a PC audio device this means that harmonic distortion (THD+N) measurements above 5 kHz have diminishing value. But this of course does not mean that the circuits are immune to high-frequency distortion. To characterize high-frequency distortion performance, use intermodulation distortion measurement techniques. These can be reliably made right up to the upper band edge. A common type of IMD measurement uses a twin-tone signal composed of two closely spaced high frequency signals. The first-order difference frequency component will fall in-band, as will higher-order sum and differences of harmonics of the two test signals. Typically the two frequencies have an offset of 1 or 2 kHz. The higher of the two test signals will be usually at the top of the available band. Common test frequencies are 19 kHz and 20 kHz.

6.6.2 THD Measurement Techniques

The most common method for measuring THD+N and the technique employed by virtually all commercial analog-based audio analyzers is the band-reject or notch filter technique. The device under test is stimulated by an ultra low distortion sinewave test signal. Contemporary analog generators are able to produce sine waves with harmonics lower than –120 dB, relative to the fundamental. The resulting signal from the output of the device under test is first normalized to a convenient operating level and then fed to a notch filter centered at the fundamental frequency of the test signal. Notch filters in contemporary audio analyzers are able to attenuate the fundamental by over 130 dB. The residual signal is band limited, measured and its level compared to the level of the fundamental. The ratio of the two is expressed as a dB or percentage figure. This residual should contain the harmonics and any noise components in the measured band and is therefore given the designation THD+N.

An alternative method of measuring THD is to use a spectrum analyzer, typically an FFT analyzer. Used unaided, this can give inaccurate readings when measuring analog domain signals. Firstly, the dynamic range of a typical FFT analyzer is limited by it's A-to-D converters and is typically in the 80 to 90 dB range, perhaps 30 to 40 dB or more worse than can be achieved by a good quality analog audio analyzer. Second, to make a THD reading, it is necessary to calculate a root sum square of all the harmonics. Some FFT analyzers may include a utility to make this computation but if not, it is a tedious exercise. Thirdly, a spectrum analyzer will usually compute THD, not THD+N. While separating the noise contribution from the harmonics may provide useful diagnostic insight, it will not allow correlation with other results expressed as THD+N. FFT analysis of digital domain signals of course does not suffer a converter signal degradation and can be reliably used for distortion measurements provided the FFT is synchronous. (A synchronous FFT does not require windowing and provides a highly selective measurement.)

PC 2001 recommends THD+N performance limits for PC audio devices. The playback path THD+N (relative to –3 dB FS) should be no worse than –65 dB FS A-weighted, the record path no worse than –60 dB FS A-weighted.

6.6.3 Multitone Total distortion

We have discussed single tone THD+N and dual tone IMD methods for measuring distortion. A newer technique is multitone distortion [8]. This process stimulates the device under test with a composite of several pure sine waves carefully chosen to span the measurement band of interest and so that their harmonics do not overlap. The analysis at the output of the device under test is done with a synchronous FFT that will examine the entire frequency band except where fundamentals exist. The root mean square sum of these components is called *Total Distortion* since it sums all of the harmonic and intermodulation products. The key to successful operation is the synchronous nature of the FFT, which does not require the usual windowing and ensures that there will be no overlap of adjacent frequency bins. Thus, the highly selective FFT can accurately measure energy as close as one bin away from a fundamental. In the digital domain, this selectivity can exceed 140 dB providing an accurate indication of very low distortion residuals.

The multitone distortion measurement technique provides at least two significant advantages over conventional single-tone measurements. It is much faster than the single tone approach. The whole measurement band is measured in parallel compared to the single tone approach that measures each frequency one at a time. Secondly, the spectrally dense multitone signal is similar to program material and stresses the device under test more realistically than a single tone. For some systems, the multitone signal provides a third advantage by stimulating the complete spectrum of interest at once. Systems with codecs and some other signal processing facilities may behave differently with a single tone stimulus than with spectrally dense program material. With these systems, the broadband multitone test signal exercises the dynamic behavior of the device the same way program material would, and yields a more realistic test result.

We mentioned that the multiple sine wave components of the multitone must be carefully chosen. The frequencies should be chosen to provide an even span over the measurement band of interest and also so that the individual harmonics do not fall on top of each other or on top of other fundamentals. With moderately dense multitones, perhaps 30 to 50 individual tones, this is relatively simple to do.

In addition to choosing the frequency of each tone, the relative phase should be chosen to minimize the crest factor of the composite waveform. A poorly chosen phase distribution can lead to signal build-up resulting in a test signal with a high crest factor. This will require that the RMS amplitude of the test signal be kept low to avoid overload of the device. Randomly distributed phase angles will produce a multitone test signal with a crest factor in the range of 3.22, similar to that of typical program material. When all the components are set to ninety degrees, a rather high crest factor of 7.84 results. Our experiments show that choosing a random distribution of phase angles produces a range of crest factors that never falls below the value produced by a zero degree setting and is frequently well above it.

6.6.4 Digital Audio and Converter Measurements

The discussions so far have focused on conventional audio measurements including frequency and phase response, dynamic range, and distortion. There are another class of measurements related to digital audio techniques that will not be addressed here. These measurements are more commonly of interest to manufacturers of components used in PC audio devices such as analog/digital converters and include parameters such as jitter. See *Measurement Techniques for Digital Audio* by Julian Dunn [7] for a thorough discussion of this subject. These additional measurements are not commonly characterized during routine PC audio device measurements, such as those required in PC2001.

6.7 Power Amplifier measurements

If the PC audio device includes a power amplifier, additional tests will be required to test the performance of this circuit. The EIA/CEA-490 [10] standard defines a series of tests for audio power amplifiers. Many of these tests are the same as those we have described here for line level outputs but power amplifiers add a few additional tests.

The standard defines *Power Output Rating* as the sinewave power output in watts that can be delivered to an its specified load at 1 kHz at no more than 1% THD+N. The load impedance (typically 8Ω for speakers) must be specified with the power rating.

The standard also describes *Dynamic Headroom*, a ratio of a pulsed sine wave at the clipping point of the amplifier to the 1% continuous sinewave power output. This measurement is more appropriate to high power amplifiers and not commonly measured on low power amplifiers as found in PC audio devices.

6.8 Codec measurements

Many high quality PC audio devices now include surround sound playback capability. These may be described as Dolby Digital 5.1 or DTS. The assessment of codec behavior will not be a subject of this paper.

When looking at codec quality there are two classes of assessment: 1- the objective assessment of the perceptual quality of the codec algorithm and 2- the determination of conformance of the specific codec implementation to that intended by the developers of the codec. Perceptual assessment is an evolving science. The work done with PEAQ (Perceptual Evaluation of Audio Quality) looks very promising and is gathering followers [11] [16]. A license and certification program managed by the codec developer generally handles measurement of conformance to a developer's standard.

6.9 Acoustic measurements

The discussion so far has focused on the electronic elements of a PC audio device. Of course, rounding out the system would typically be speakers and perhaps a microphone. Acoustic measurements involve a number of specialized techniques that are outside the intended area of this paper. However, we will give a brief overview of the main considerations.

Measurements of transducers such as speakers and microphones add an additional dimension to the testing process. The testing environment will have a significant impact on the measurements. The location used for the testing will most certainly be bounded by walls, floor, and ceiling, all of which present a reflective surface. A speaker is tested by positioning a measurement microphone in front of it. This microphone will pick up the incident sound from the speaker but also the reflected versions of this signal. The incident and reflected signals will add and subtract at various parts of the spectrum based on the delays between incident and reflected signals. If a typical swept sine wave test signal is used to measure frequency response, the results will be erroneous. For this reason, acoustic transducers are often characterized in an anechoic chamber. A room such as this has special low reflectivity treatment to wall, floor, and ceiling surfaces. It is difficult to create surfaces that are non-reflective at low frequencies, but anechoic rooms will provide accurate frequency response measurements over most of the audio band.

An anechoic chamber is expensive and not readily available to most individuals wanting to test speakers. Over the years there have been several electronic techniques developed to allow speakers to be measured in a non-anechoic environment—that is, a typical room. All of the techniques attempt to separate the incident and reflective signals by relying on the assumption that the reflected signal arrives later than the incident signal. Time Domain Spectrometry (TDS) and time gating are two techniques that provide narrow time or frequency windows to reject reflected signals from the measurement.

A more recent technique uses a special broad-spectrum noise signal called a Maximum Length Sequence (MLS)[17] and applies cross correlation on the analysis side to extract the impulse response. From this response, time windowing can be applied to detect the incident and reject the reflected signal. While there are other techniques to derive the impulse response, the MLS technique offers a significant signal-to-noise ratio improvement over the single impulse technique, in the range of 45 dB. With the time-windowed impulse response, FFT techniques can be used to extract an accurate frequency and phase response characterization comparable to what would be obtained in a true anechoic environment. The MLS technique applies well to measurement of PC audio speakers in a typical lab or listening room test environment and is part of the measurement capability of some commercial audio analyzers.

Complete characterization of a speaker system is an involved process and would extend beyond the capabilities of MLS and the functions of the typical audio analyzer. But the frequency response measurement provided by the simple MLS technique is very useful and will be a relatively good indication of speaker quality.

The MLS measurement should be made using a measurement-grade microphone on axis with the speaker, that is pointed directly at the center of the speaker and approximately one-third meter (one foot) from the speaker for the near-field response. Reflections should be minimized by attempting to keep the speaker and microphone as far away as possible from all surfaces. An ideal, but difficult positioning, is to suspend the microphone and speaker near the center of the room, thus typically providing more than a meter (3-foot) distance to any reflective surface. Avoid sitting the speaker on a table that will present a strong close-in reflective surface. Close reflective surfaces make the time difference between incident and reflected signals very small, with consequent difficulty rejecting the reflected signals.

7 Time related measurements

Propagation delay, latency, and interchannel time differences are of little concern on conventional analog audio systems as they are virtually non-existent. However, the multi-tasking environment of the PC presents new challenges in this area. The analog to digital and digital to analog conversion process itself can introduce absolute delays and more problematical differential delays. Some of these errors are a function of the PC handling other tasks and, in some cases, the characteristics of the operating system as it assigns and shares resources.

In addition to the audio facilities provided on PC audio devices, other capabilities frequently included on such systems are a synthesizer generation and MIDI processing facility. Here as well, the multi-tasking nature of the PC environment can introduce unwanted delays and latencies. One of the most annoying of these is MIDI latency, a measure of synthesizer responsiveness. This is the delay between the sending of a MIDI command to the MIDI input of the audio subsystem (from an external keyboard, for example) to the start of the audio output of the selected note by the synthesizer. An additional path that should be characterized for latency is the audio streaming versus MIDI data output. That is, if audio is being streamed from .wav files while MIDI data is being outputted via the MIDI port, how is the synchronism being maintained?

We developed a simple test for MIDI latency. Using a MIDI transmit device that is able to generate a MIDI “note” and an audio analyzer wuth an programmable acquisition buffer, we initiated a MIDI event and then recorded the time to produce an audio output representative of the requested event. We characterized the inherent propogation delay of the measurement system and backed this out of the measurement. This setup was able to meassure the time in milliseconds from the initial edge of a MIDI note request until the beginning of the audio result.

7.1 Frequency Accuracy and Pitch

Pitch is an extremely important parameter to a musician or music listener. When a music file is played back through the audio system, the sample rate must match the sample rate used during recording for accurate pitch to be maintained. In a PC audio system, this would typically be 44,100 Hz or 48,000 Hz.

The test recommended by the standards measures the frequency of a test signal of 997 Hz while being played back at a standard sample rate of 44.1 kHz. The AES6id document recommends that this sample rate be accurate to ±25 μHz, a frequency accuracy that is unlikely to be achieved. The predessor to this document [3] used a tolerance of ±25 ppm. AES6id also suggests that the original test signal of 997 Hz should be accurate to within ±25 μHz and that the frequency measurement equipment used to make this determination be accurate to ±10 μHz. Again, a more realistic tolerance would be 997 Hz ±25 ppm and a frequency meter with an accuracy of ±10 ppm.

8 Resource usage

A consistent delivery of program material is taken for granted in conventional analog audio systems. But in a PC environment, the rendering and delivery of audio program material must compete with other processes, possibly including delivery of accompanying video program material. Recognizing the unacceptability of disruptions to the audio stream, various error correction systems are present to attempt to deliver a steady flow of program material. Occasionally, these systems may produce a “glitch”, a momentary signal dropout, a click or pop or a repeat of a portion of a signal. These are generally caused by system resources being overtaxed and unable to then service the needs of the audio subsystem. A test has been developed to monitor an audio stream, detect such glitches, and record the time and quantity of these glitches. The test system makes these assessments while the host computer is exercised through a routine of activity designed to stress various resources to provoke the occurrence of such glitches.

The test uses a pure single sine wave as the audio signal and measures the THD+N of this signal using a conventional notch-filter audio analysis technique. To achieve the required rejection of the test frequency, such notch filters must use sophisticated automatic tuning. These automatic systems continuously monitor and correct the fine-tuning of the filter effectively adapting to any gain or phase changes as they happen. Notch attenuations of over 120 dB are easily achievable with good designs.

Although this tuning system is extremely sensitive to any amplitude, frequency, or phase changes in the test signal, it has a time constant while it attempts to quickly correct for changes. During the correction interval, the notch depth is compromised. If the level of fundamental at the output of the notch filter is monitored, any significant change in the level of this signal will be reflective of even a slight change in the test signal. Tests we conducted showed easy detection of a change of even a single cycle of the test waveform. Any discontinuity of the waveform, such as would happen with a repeat, will exhibit an instantaneous phase change with resulting servo correction and a momentary large increase in the output from the notch filter. A dropout or level shift will also produce a large notch filter output level change.

To date, this tool has been used only for investigative purposes to assess the impact of activity in the PC not related to the audio streaming. If the PC is playing a music file, we would like that program material to continue as intended while other PC activity takes place. Perhaps another file is being downloaded to the hard drive or the PC is being used to send an email. Even large bursts of CPU and hard drive activity should not disrupt the integrity of the audio playback. This notch filter "glitch" detection tool allows convenient, objective, and predictable determination of any disruption of the audio playback while various PC activities are exercised. Differences in how effectively various operating systems and audio drivers handle the multitasking resource assignments can be assessed. The tool includes a logging facility allowing long unattended operation.

9 Test Reports

Test reports can take several forms depending on their application. Reports for certification to a standard need only express the relevant numerical results for comparison. A development environment may appreciate graphical presentation for easy analysis for results.

The most important issue with test reports is to be sure they completely specify the conditions of measurement for every reported value. For example, it is essential that the weighting filter used for dynamic range be specified (A-weighting, CCIR-RMS or 20 Hz to 20 kHz unweighted). Similarly, for distortion, what type of distortion is being measured, what weighting, if any, is applied?

Because of the familiarity of certain notations used commercially to present audio test results for home stereo equipment, there has been a slow adoption of correct results presentation of PC audio device performance results. Commercial sound cards still have retail packaging with statements like "96 dB SNR" rather than "96 dB dynamic range CCIR-RMS weighted". Distortion results are specified without a frequency reference or weighting information.

Appendix A summarizes the results of measuring several representative commercially available PC audio devices (sound cards and mother board implementations) to illustrate a range of results.

Appendix B shows some examples of test reports produced by the application software used for the test results gathered during the preparation of this paper.

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Appendix A - Examples of Measured Results

The following pages present the results of measurements on several commercial sound cards or integrated mother board audio devices to illustrate a range of performance levels. Five devices were tested on two different computers. The measurements were made using PC2001 recommendations.

All measurements were made using a 44.1 kHz sample rate and 16-bit word width. All THD+N measurements and dynamic range measurements were A-weighted. Here is a summary of the devices that were tested:

- Device A: High end sound card
- Device B: Pro-audio sound card
- Device C: On-board audio device
- Device D: On-board audio device
- Device E: Older model sound card
- Machine A: year 2002 vintage PC
- Machine B: year 2000 vintage PC

PC DA (Playback) tests

		Device A Machine A	Device B Machine A	Device C Machine A	Device A Machine B	Device D Machine B	Device E Machine B
THD+N A-weighted in dB	Ch A	-87.10	-84.633	-85.379	-87.592	-38.087	-81.583
	Ch B	-87.57	-84.908	-85.456	-87.592	-37.985	-81.729
Multitone Total Distortion in dB	Ch A	-86.05	-83.99	-76.003	-86.28	-48.887	-29.044
	Ch B	-85.72	-84.342	-80.257	-86.239	-45.346	-78.617
Dynamic Range, A-weighted in dB	Ch A	-96.92	-100.398	-89.415	-99.324	-51.447	-81.583
	Ch B	-96.92	-100.267	-89.415	-100.461	-49.013	-81.729
Dynamic Range, multitone in dB	Ch A	-93.08	-97.809	-78.318	-96.437	-49.414	-29.044
	Ch B	-93.01	-97.865	-83.21	-96.433	-46.863	-78.617
Frequency response ± dB rel to 1 kHz							
20 Hz to 1 kHz	Ch A	0.01/-0.068	0.008/-0.016	0.021/-0.178	0.01/-0.07	0.023/-0.912	-0.081/-0.38
20 Hz to 1 kHz	Ch B	0.01/-0.069	0.008/-0.013	0.021/-0.156	0.01/-0.071	0.12-2.906	-0.084/-0.627
1 kHz to 20 kHz	Ch A	0/-1.78	0.041/-0.091	0.097/-0.697	0/-1.763	0.025/-0.911	0.933/-4.424
1 kHz to 20 kHz	Ch B	0/-1.77	0.025/-0.144	0.092/-0.709	0/-1.762	0.146/-2.869	0.998/-2.799
Phase response deviation in degrees 16 Hz to 20 kHz	Max	0.176	0.483	0.967	0.176	0.527	157.939
	Min	-0.088	0	0.044	-0.088	-0.396	-2.549
Line Out level in dBV for 0 dB FS wav file	Ch A	6.242	1.993	0.339	6.257	-3.332	0.543
	Ch B	6.256	2.037	0.208	6.27	-3.27	0.408
THD+N at ref level in dB	Ch A	-84.3	-81.921	-81.46	-84.319	-38.355	-76.999
	Ch B	-83.79	-82.071	-81.623	-83.805	-37.821	-76.423
Sample rate error ± %		< 0.001	-0.001%	-0.006	-0.001	0.016	0.008

The playback tests (above) show some interesting differences among the tested devices.

Device D clearly does not perform very well, even if the architecture of the machine B does not show significant differences for Device A. The output level calibration could not reach the 1% distortion for this device.

Device E shows some bad interchannel phase errors and interesting ripples in the frequency response.

Device C clearly outperforms device D, which shows an improvement for the on-board chipsets.

Devices A has the best sample rate error as well as the best THD+N, while device B shows the best dynamic range performance.

The multitone tests show a large difference between the distortion of two channels for the device E which can be partially understood by looking at the f>1 kHz frequency response error. A frequency response plot for this device shows some irregular ripples above 2 kHz and also that the -3dB point is bellow 20 KHz.

A to A (Analog Loop) tests							
		Device A Machine A	**Device B Machine A**	**Device C Machine A**	**Device A Machine B**	**Device D Machine B**	**Device E Machine B**
THD+N A-weighted in dB	Ch A	-81.85	-82.241	-84.36	-86.376	-28.993	
	Ch B	-79.33	-82.163	-84.858	-86.692	-38.502	
Multitone Total Distortion in dB	Ch A	-85.50	-82.358	-84.175	-85.853	-29.272	
	Ch B	-86.37	-82.345	-85.677	-86.23	-44.651	
Dynamic Range, A-weighted in dB	Ch A	-100.52	-98.753	-94.118	-97.818	-60.213	
	Ch B	-100.52	-99.296	-94.046	-98.417	-39.81	
Dynamic Range, multitone in dB	Ch A	-97.10	-98.626	-94.261	-96.88	-60.663	
	Ch B	-97.05	-99.169	-94.2	-96.652	-40.792	
Frequency response ± dB rel to 1 kHz							
20 Hz to 1 kHz	Ch A	0.01/-0.226	0.008/-0.062	0.007/-3.885	0.01/-0.226	0.008/-4.11	
20 Hz to 1 kHz	Ch B	0.0/-0.226	0.008/-0.059	0.007/-4.004	0.009/-0.226	-0.033/-16.474	
1 kHz to 20 kHz	Ch A	0/-2.08	0.016/-0.329	0.008/-0.03	0/-2.075	0.012/-0.037	
1 kHz to 20 kHz	Ch B	0/-2.08	0.01/-0.381	0.008/-0.042	0/-2.074	0.085/0	
Phase response deviation in degrees 16 Hz to 20 kHz	Max	0	0.615	0.659	0	-0.044	
	Min	-0.08	-0.176	-0.439	-0.32	-46.683	
Line Out level in dBV	Ch A	+6.149	1.645	3.658	6.991	6.991	
	Ch B	+6.341	1.737	3.382	7.154	7.154	
Line In level in dBV for 0 dB FS ref	Ch A	6.218	1.975	0.313	-4.731	-4.731	
	Ch B	6.211	1.971	0.311	-4.731	-4.731	
THD+N at ref level in dB	Ch A	-81.85	-79.341	-60.87	-60.925	-39.246	
	Ch B	-79.33	-79.532	-60.899	-40.994	-38.421	

In the above analog loop results we see that the PC model influences the Device A (especially the reference levels).Also while Device B has almost the same distortion and dynamic range as for the D-A test, Device A shows a drop for the aforementioned measurements.

In the A to D (record loop) results on the next page, Device A proves again the PC influence on the audio device. Also notice the almost constant reference levels for Device B that made the calibration very easy troughtout the test.

AD to PC (Record Analog Loop) tests							
		Device A Machine A	Device B Machine A	Device C Machine A	Device A Machine B	Device D Machine B	Device E Machine B
THD+N A-weighted in dB	Ch A	-86.85	-84.598	-76.31	-89.473	-71.755	
	Ch B	-86.74	-84.458	-76.875	-89.53	-51.517	
Dynamic Range, A-weighted in dB	Ch A	-90.53	-95.098	-73.997	-88.787	-73.935	
	Ch B	-90.96	-94.61	-76.318	-88.608	-51.531	
Frequency response ± dB rel to 1 kHz							
20 Hz to 1 kHz	Ch A	0.11/-0.02	0.007/-0.031	0.019/-1.872	0.117/-0.002	0.019/-2.172	
20 Hz to 1 kHz	Ch B	0.11/-0.02	0.007/-0.031	0.018/-1.799	0.117/-0.024	-0.12/-13.165	
1 kHz to 20 kHz	Ch A	0.12/-2.06	0.002/-0.243	0.046/-6.147	0.12/-2.063	0.062/-3.075	
1 kHz to 20 kHz	Ch B	0.12/-2.06	0.002/-0.244	0.045/-6.148	0.12/-2.063	0.096/-3.041	
Phase response deviation in degrees 16 Hz to 20 kHz	Max	+0.308	0.176	0.791	0.264	0	
	Min	-0.044	-0.176	0.044	0	-43.418	
Line In level in dBV for 0 dB FS wav file	Ch A	-4.75	1.973	0.313	-2.554	0	
	Ch B	-4.75	1.978	0.32	-3	0	
Digital Output level in dB FS	Ch A	-0.405	-0.318	-0.306	-5.619	-16.573	
	Ch B	-0.559	-0.366	-0.204	-5.611	-16.567	
THD+N at ref level in dB	Ch A	-42.73	-81.886	-57.278	-87.874	-41.954	
	Ch B	-73.38	-82.045	-43.973	-87.59	-51.547	

AD to PC to DA tests							
		Device A Machine A	Device B Machine A	Device C Machine A	Device A Machine B	Device D Machine B	Device E Machine B
THD+N A-weighted in dB	Ch A	-80.64	-82.424	-80.6	-88.42	-37.979	
	Ch B	-81.58	-82.81	-83.395	-88.64	-38.171	
Multitone Total Distortion in dB	Ch A	-25.972	-79.34	-25.972	-13.363	-18.012	
	Ch B	-75.759	-81.744	-75.759	-86.293	-43.253	
Dynamic Range, A-weighted in dB	Ch A	-91.5	-96.76	-83.861	-91.834	-52.291	
	Ch B	-90.9	-97.024	-83.396	-92.503	-49.467	
Dynamic Range, multitone in dB	Ch A	-89.09	-94.41	-73.282	-71.824	-50.095	
	Ch B	-88.79	-94.631	-74.398	-89.182	-43.591	
Frequency response ± dB rel to 1 kHz							
20 Hz to 1 kHz	Ch A	0.108/-0.102	0.008/-0.062	0.033/-1.992	0.107/-0.104	0.039/-2.007	
20 Hz to 1 kHz	Ch B	0.107/-0.101	0.008/-0.057	0.031/-2.052	0.108/-0.098	0.046/-10.205	
1 kHz to 20 kHz	Ch A	0.113/-3.836	0.016/-0.329	0.143/-6.832	0.113/-3.837	0.183/-5.966	
1 kHz to 20 kHz	Ch B	0.113/-3.835	0.01/-0.381	0.139/-6.842	0.113/-3.836	0.23/-5.924	
Phase response deviation in degrees 16 Hz to 20 kHz	Max	0	0.615	0.571	0	0.22	
	Min	-0.132	-0.176	-0.396	-0.088	-41.968	
Line Out level in dBV for 0 dB FS wav file	Ch A	+6.242	1.993	0.339	6.257	-3.332	
	Ch B	+6.263	2.043	0.214	6.276	-3.898	

The complete loop through path of A to D to PC to D to A is shown in the final table above. The fact that this test is not just a union of results is clearly proven by the multitone and frequency error for the Device A, the behavior being consistent across the two tested machines. Again Device B shows the same consistency in the measurements.

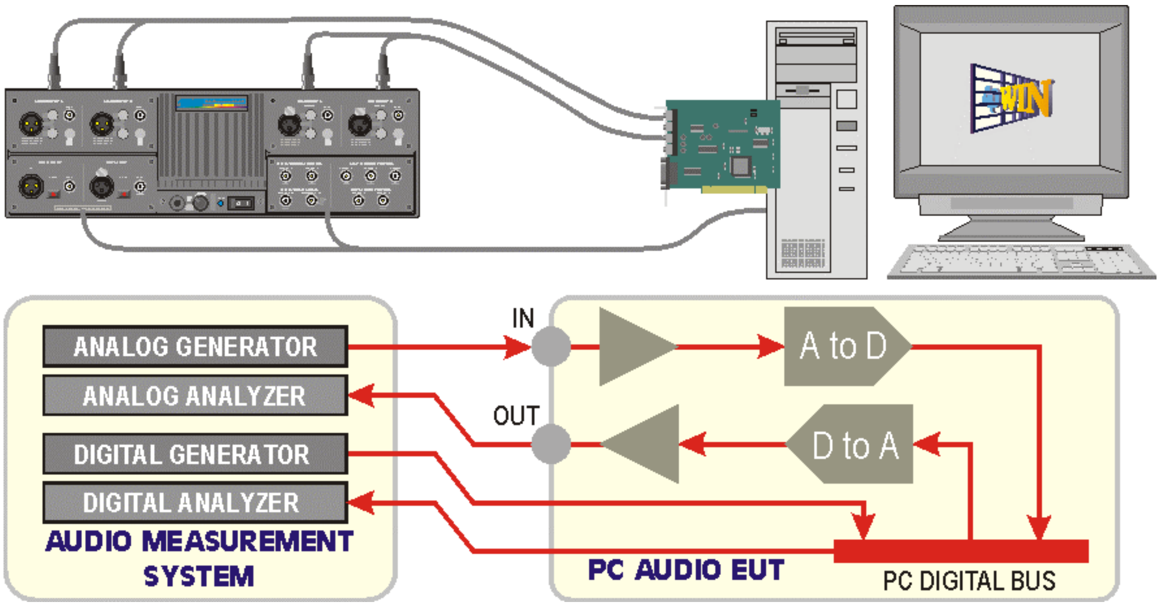
Appendix B Sample Test Reports

The following pages illustrate some examples of test reports produced by the testing application used during this project. The application is able to generate reports in Microsoft Word format or HTML format.

Pages 23 through 31 are an example of a test report generated as a Word document. Page 23 is a description of the card, serial number etc. Page 24 is the test results of the D to A (playback) section. Page 25 is a frequency and phase response graph of the same data. Pages 26 and 27 shows data for the analog loop through path. Pages 28 and 29 shows data for the A to D (record) path. Finally, pages 30 and 31 illustrate data for the A to D, PC loop, and back through the D to A path.

PC Audio Device Performance Tests

Test Results

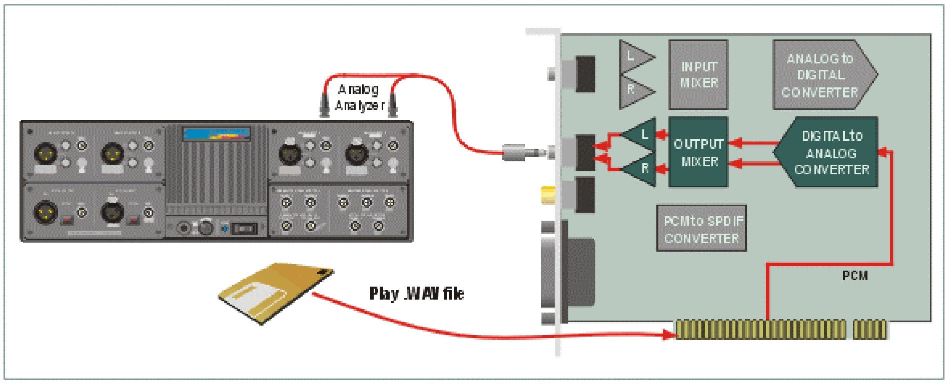


MODEL –	**Generic Sound Card**	SERIAL # –	**0001**
Done for	**Audio Precision**	by	**Mr. Sound Engineer**
Tested on	**May/3/2000**	and started at	**15:34**

GENERAL TEST COMMENTS: Example Report

DUT SAMPLE RATE **44100** Hz
SAMPLE RATE ERROR **-0.003 %**

D to A TEST GROUP Generic Sound Card



Signal source is a WAV file

ANALOG OUTPUT REFERENCE LEVEL from 0dBFS 997Hz WAV file

0dBFS digital level = 0dBr at these Output Amplitudes.

ChA (Left)	**0.883 dBV**	ChB (Right)	**0.845 dBV**

THD+N at this Digital Level *Maximum of -40dB (1.0%)*

ChA (Left)	**-82.578 dB**	ChB (Right)	**-82.722 dB**

DISTORTION at **997Hz** and **-3 dBFS** amplitude.

FASTTEST with ***22kHz low-pass and NO weighting***

ChA (Left)	**-84.478 dBr**	ChB (Right)	**-84.875 dBr**

Analog Analyzer with 22kHz low-pass and NO weighting

ChA (Left)	**-84.928 dBr**	ChB (Right)	**-85.117 dBr**

DYNAMIC RANGE (Noise with Low-level signal) at **997Hz** and **-60dBFS**.

FASTTEST with ***22kHz low-pass and NO weighting***

ChA (Left)	**-86.795 dBr**	ChB (Right)	**-86.812 dBr**

Analog Analyzer with ***MSWFilters***

ChA (Left)	**-86.816 dBr**	ChB (Right)	**-86.901 dBr**

FREQUENCY RESPONSE ERROR

FASTTEST ISO31 multi-tone signal, 1/3oct. spacing from 16Hz-20kHz.
Error is in dB relative to 1kHz = 0 dBr reference.

Low Frequency (16Hz-1kHz)

ChA MAX	**0.005 dBr**	ChB MAX	**0.005 dBr**
ChA MIN	**0.001 dBr**	ChB MIN	**0.001 dBr**

High Frequency (1kHz-20kHz)

ChA MAX	**0.030 dBr**	ChB MAX	**0.028 dBr**
ChA MIN	**-3.876 dBr**	ChB MIN	**-3.900 dBr**

RELATIVE PHASE ERROR

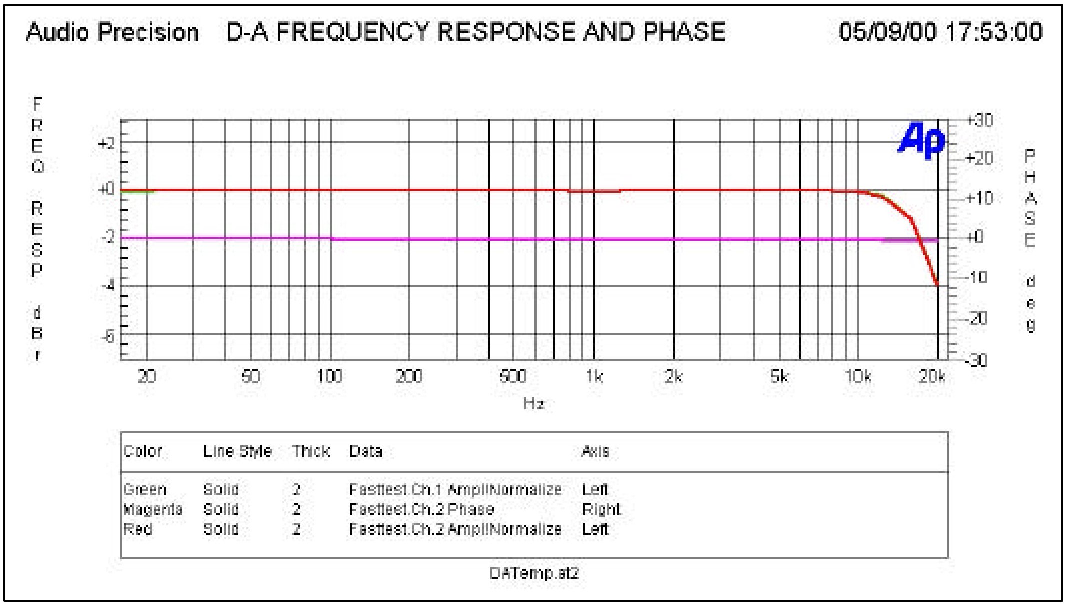
FASTTEST Phase - ISO31 multi-tone, measurement from 16Hz-20kHz.

Phase error range **0.132 deg** to **-0.439 deg**

Data file for Response and Phase is saved as: **Generic Sound CardDA.ada**

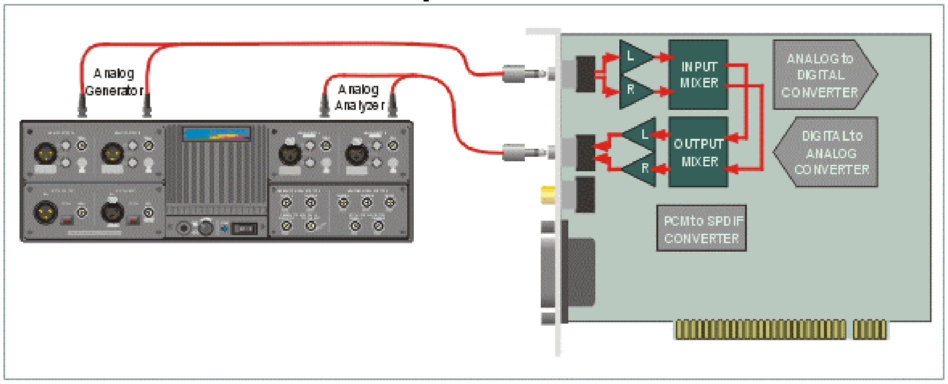
D to A TEST GROUP Generic Sound Card

Frequency and Phase Response Graph



D to A TEST COMMENTS: Test comment for Example Report.

A-to-A Test Group Generic Sound Card



Signal source is Audio Generator to Line Input

ANALOG OUTPUT REFERENCE LEVEL from Line In			
Output Levels = 0dBr at these Output Amplitudes.			
ChA (Left)	**-0.374 dBV**	ChB (Right)	**-0.368 dBV**
THD+N at Ref. Level Maximum of -40dB (1.0%)			
ChA (Left)	**-66.201 dB**	ChB (Right)	**-60.653 dB**
*DUT Line Input Levels for Gen output of **0.883 dBV***			
ChA (Left)	**0.863 dBV**	ChB (Right)	**0.865 dBV**

DISTORTION at **997Hz** and **-3 dBr** (0dBr = maximum amplitude).			
*FASTTEST with **22kHz low-pass and NO weighting***			
ChA (Left)	**-76.214 dBr**	ChB (Right)	**-76.215 dBr**
*Analog Analyzer with **22kHz low-pass and NO weighting***			
ChA (Left)	**-72.468 dBr**	ChB (Right)	**-72.019 dBr**

DYNAMIC RANGE (Noise with Low-level signal) at **997Hz** and **-60dBFS**.			
*FASTTEST with **22kHz low-pass and NO weighting***			
ChA (Left)	**-82.705 dBr**	ChB (Right)	**-83.037 dBr**
*Analog Analyzer with **22kHz low-pass and NO weighting***			
ChA (Left)	**-74.286 dBr**	ChB (Right)	**-73.699 dBr**

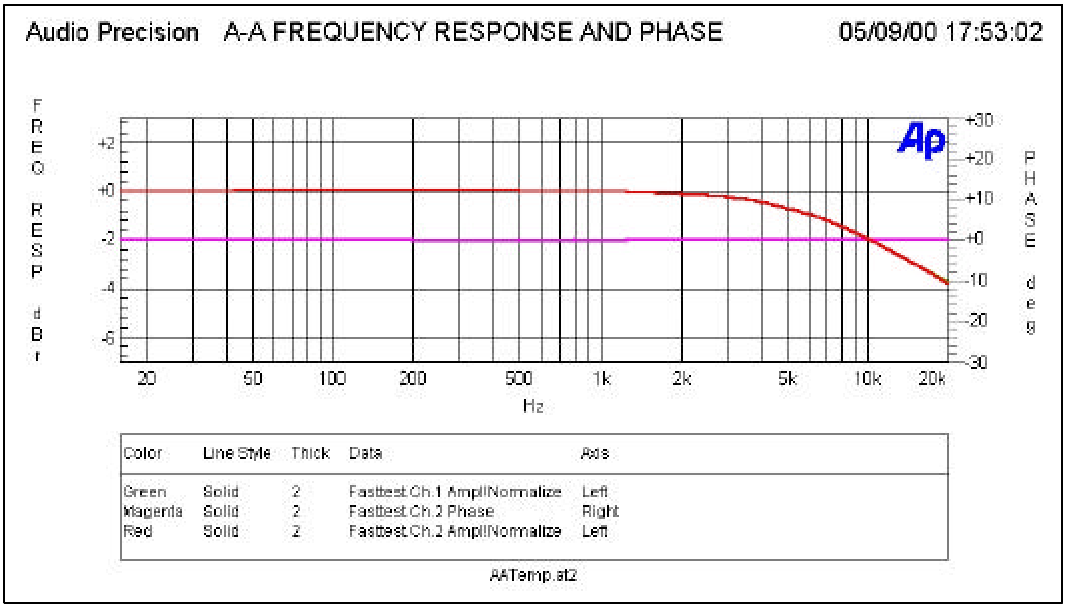
FREQUENCY RESPONSE ERROR			
FASTTEST ISO31 multi-tone signal, 1/3oct. spacing from 16Hz-20kHz. Error is in dB relative to 1kHz = 0 dBr reference.			
Low Frequency (16Hz-1kHz)			
ChA MAX	**0.040 dBr**	ChB MAX	**0.040 dBr**
ChA MIN	**0.014 dBr**	ChB MIN	**0.014 dBr**
High Frequency (1kHz-20kHz)			
ChA MAX	**0.000 dBr**	ChB MAX	**0.000 dBr**
ChA MIN	**-3.747 dBr**	ChB MIN	**-3.796 dBr**

RELATIVE PHASE ERROR
FASTTEST Phase - ISO31 multi-tone, measurement from 16Hz-20kHz.
Phase error range **0.220 deg** to **0.000 deg**

Data file for Response and Phase is saved as: **Generic Sound CardAA.ada**

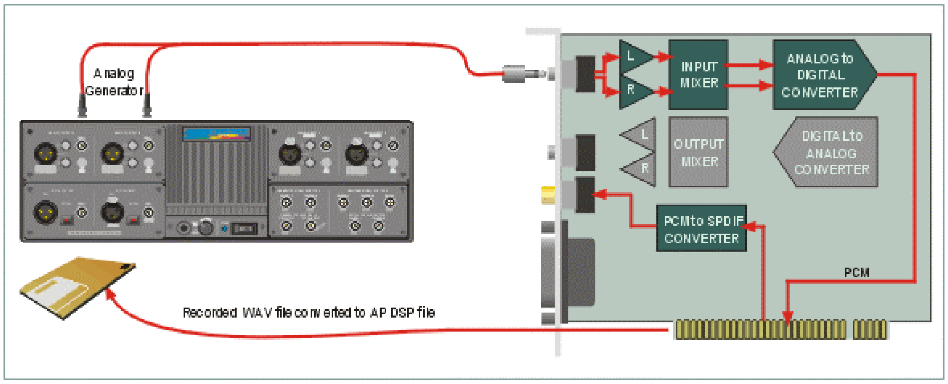
A to A TEST GROUP Generic Sound Card

Frequency and Phase Response Graph



A to A TEST COMMENTS: Test comment for Example Report.

A to D TEST GROUP Generic Sound Card



Signal source is Audio Generator to Line Input

DIGITAL REFERENCE LEVEL from recorded Line In

Digital levels should be about 0dBFS.

ChA (Left)	**-0.717 dBFS**	ChB (Right)	**-0.650 dBFS**

THD+N at this Digital Level *Maximum of -40dB (1.0%)*

ChA (Left)	**-80.781 dBFS**	ChB (Right)	**-80.418 dBFS**

*DUT Line Input Levels for Gen output of **5.039 dBV***

ChA (Left)	**5.026 dBV**	ChB (Right)	**5.025 dBV**

DISTORTION at **997Hz** and **-3 dBr** (0dBr = maximum amplitude).

FASTTEST with 22k bandwidth and NO weighting.

ChA (Left)	**-82.700 dBFS**	ChB (Right)	**-82.464 dBFS**

DYNAMIC RANGE (Noise with Low-level signal) at **997Hz** and **-60dBFS**.

FASTTEST with 22k bandwidth and NO weighting. 0dBr = 0dBFS

ChA (Left)	**-87.748 dBFS**	ChB (Right)	**-87.701 dBFS**

FREQUENCY RESPONSE ERROR

FASTTEST ISO31 multi-tone signal, 1/3oct. spacing from 16Hz-20kHz.
Error is in dB relative to 1kHz = 0 dBr reference.

Low Frequency (16Hz-1kHz)

ChA MAX	**0.031 dBr**	ChB MAX	**0.031 dBr**
ChA MIN	**-0.047 dBr**	ChB MIN	**-0.047 dBr**

High Frequency (1kHz-20kHz)

ChA MAX	**0.175 dBr**	ChB MAX	**0.176 dBr**
ChA MIN	**-0.605 dBr**	ChB MIN	**-0.582 dBr**

RELATIVE PHASE ERROR

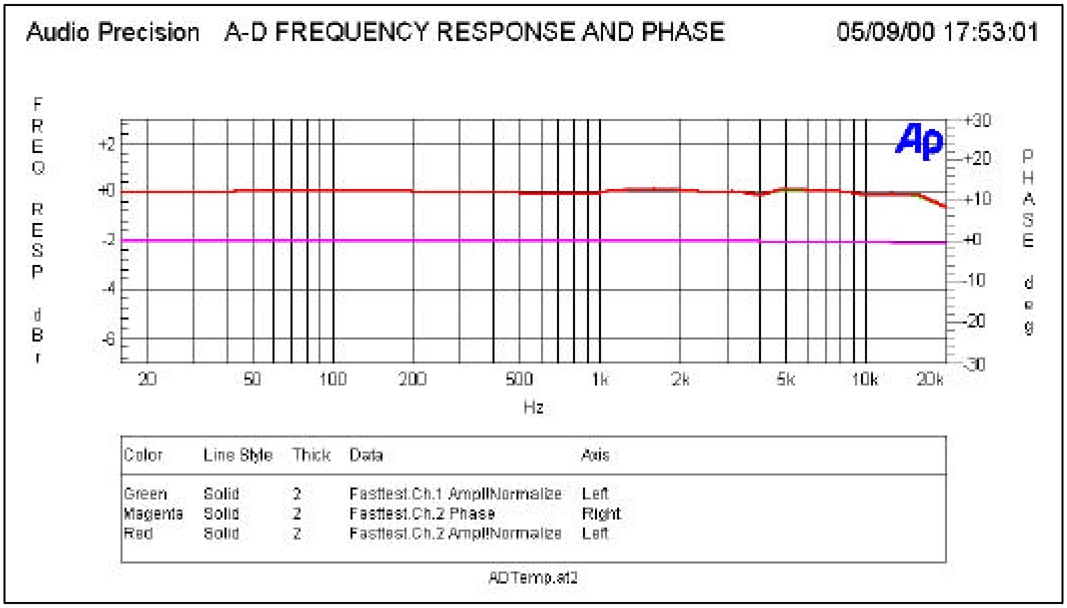
FASTTEST Phase - ISO31 multi-tone, measurement from 16Hz-20kHz.

Phase error range **0.000 deg** to **-0.703 deg**

Data file for Response and Phase is saved as: **Generic Sound CardAD.ada**

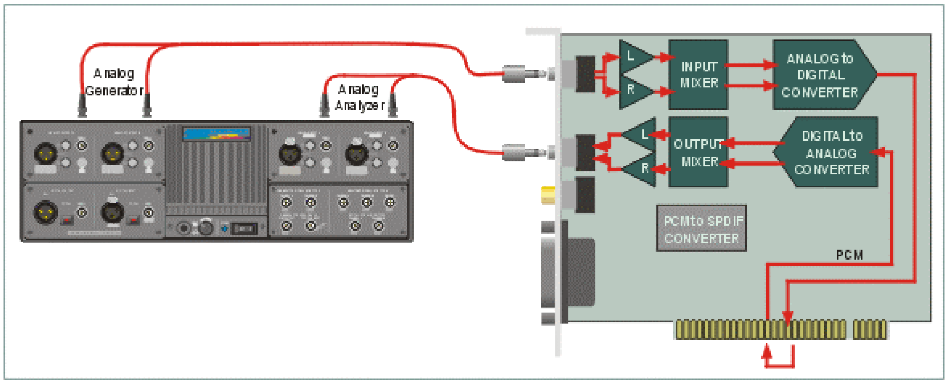
A to D TEST GROUP Generic Sound Card

Frequency and Phase Response Graph



D to A TEST COMMENTS: Test comment for Example Report.

A to D to PC to D to A TEST GROUP Generic Sound Card



Signal source is Audio Generator to Line Input

ANALOG OUTPUT REFERENCE LEVEL from Line In

EUT Output Levels: 0dBr = 0dBFS at these Output Amplitudes.

ChA (Left)	**0.883**	ChB (Right)	**0.846**

Generator Output Amplitude at Line Inputs of EUT.

ChA (Left)	**5.026 dBV**	ChB (Right)	**5.025 dBV**

DISTORTION at 997Hz and **-3 dBr** (0dBr = maximum amplitude).

*FASTTEST with **22kHz low-pass and NO weighting***

ChA (Left)	**-81.678 dBr**	ChB (Right)	**-81.790 dBr**

*Analog Analyzer with **22kHz low-pass and NO weighting***

ChA (Left)	**-81.543 dBr**	ChB (Right)	**-81.800 dBr**

DYNAMIC RANGE (Noise with Low-level signal) at **997Hz** and **-60dBr**.

*FASTTEST with **22kHz low-pass and NO weighting***

ChA (Left)	**-85.101 dBr**	ChB (Right)	**-85.008 dBr**

*Analog Analyzer with **22kHz low-pass and NO weighting***

ChA (Left)	**-85.204 dBr**	ChB (Right)	**-85.310 dBr**

FREQUENCY RESPONSE ERROR

FASTTEST ISO31 multi-tone signal, 1/3oct. spacing from 16Hz-20kHz.
Error is in dB relative to 1kHz = 0 dBr reference.

Low Frequency (16Hz-1kHz)

ChA MAX	**0.034 dBr**	ChB MAX	**0.034 dBr**
ChA MIN	**-0.046 dBr**	ChB MIN	**-0.045 dBr**

High Frequency (1kHz-20kHz)

ChA MAX	**0.178 dBr**	ChB MAX	**0.179 dBr**
ChA MIN	**-4.481 dBr**	ChB MIN	**-4.483 dBr**

RELATIVE PHASE ERROR

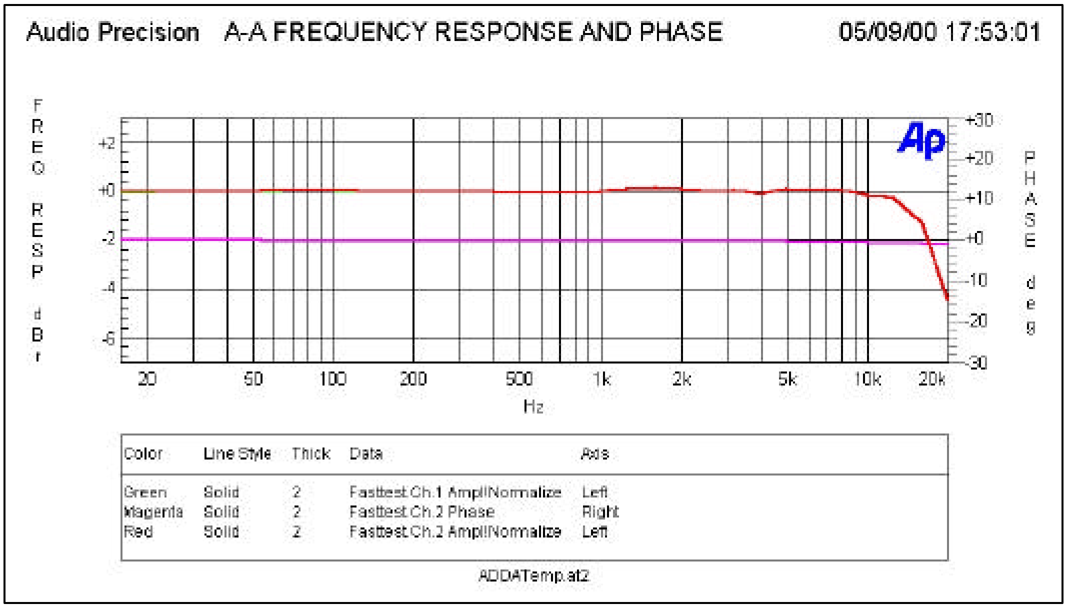
FASTTEST Phase - ISO31 multi-tone, measurement from 16Hz-20kHz.

Phase error range **0.132 deg** to **-1.230 deg**

Data file for Response and Phase is saved as: **Generic Sound CardADDA.ada**

A to D to PC to D to A TEST GROUP Generic Sound Card

Frequency and Phase Response Graph



A to D to PC to D to A TEST COMMENTS: Test comment for Example Report.