# AES-6id-2000

Version 3 draft

# AES information document for digital audio — Personal computer audio quality measurements

**Abstract**

This document focuses on the measurement of audio quality specifications in a PC environment. Each specification listed has a definition and an example measurement technique. Also included is a detailed description of example test setups to measure each specification.

**Table of Contents**

## Foreword

[This foreword is not a part of AES information document for digital audio — Personal computer audio quality measurements, AES-6id-2000.]

This document was prepared by a writing group of the SC-02-01 Working Group on Digital Measurement Techniques of the SC-02 Subcommittee on Digital Audio. Steven Harris headed the writing group.

The text of the Crystal Audio Division of Cirrus Logic, Inc. version 1.0 paper on measurement of PC audio was used to create the first proposed working group draft of this document. The working group felt that the wide-spread use of the Cirrus paper in the computer industry warranted the preparation an AES document based on it.. AES documents are subject to due process based on AES procedures and receive a public review. Their format is based on the IEC-ISO Directives, Part 3, which may be downloaded from www.iec.ch.

Initial discussions revealed that a full consensus on all provisions of the first proposed draft would not be possible. The group chose, therefore, to prepare an information document rather than a standard. An AES information document, while formatted in the same style as a standard, does not require full consensus for publication. Instead, public comments that cannot be resolved may be published in an informative annex to the document. In this manner, the information becomes quickly available for use in the industry while questions regarding particular provisions are made available for consideration by users. The comments are compiled in annex D at the end of this document.

The document remains under continuous review under project AES-6-id-R assigned to the SC-02-01 working group. It may be amended by following AESSC due-process procedures which require the same public review before publication as a new standard. Participation in this process is open to all directly and materially affected individuals who join the working group in the manner described at www.aes.org/standards/.

The document does not suggest performance limits. Such limits for some of these measurements are suggested in, for example, Microsoft PC-2001, Intel's AC '97, and in the MPC3 specification.

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2000-02-20

# AES-6id-2000

# AES information document for digital audio — Personal computer audio quality measurements

## 1 Scope

This document focuses on the measurement of audio quality specifications for devices used in or connected to a personal computer (PC). Each specification listed has a definition and an example measurement technique. Also included is a detailed description of example test setups to measure each specification. Information on signal paths and weighting filters is given in annexes.

## 2 Normative references

The following standard contains provisions which, through referenced in this text, constitute provisions of this document. At the time of publication, the edition indicated was valid. All standards are subject to revision, and parties to agreements based on this document are encouraged to investigate the possibility of applying the most recent editions of the indicated standards.

AES17-1997, *AES standard method for digital audio engineering — Measurement of digital audio equipment*. NY: Audio Engineering Society, Inc., (1997).

## 3 Definitions

### 3.0 Personal Computer (PC)

A computer designed for use by an individual in the form of a desktop or notebook. In this context, a “PC audio device” is what is commonly called a “sound card”, an externally-connected device, or a mother-board audio implementation.

### 3.1 Signal Paths

A PC Audio Device has several signal paths that should be characterized independently and in various combinations. The following definitions describe the common paths present in most PC audio devices. Also see Annex B for a more detailed description of each path and block diagrams.

#### 3.1.1 Playback (D-A) Path

The path from the internal PC bus routed through the digital to analog converter, the Playback Mixer, and out through the analog output amplifier. Digital signals may be present on the PC bus by playing or streaming a .wav file from hard disk or PC memory or by routing a signal from the digital input to the PC bus.

#### 3.1.2 Analog Loop (A-A) Path

The path from the analog input amplifier, routed through the Playback Mixer, and back out through the analog output amplifier. Several analog inputs may exist such as: Line Input, Microphone Input, Auxiliary Input, CD Drive Input, Telephone Modem (TDA) Audio, and others.

### 3.1.3 Record (A-D) Path

The path from the analog input amplifier, routed through the Record Mixer, and through the analog to digital converter to the PC bus. The digital signal on the PC bus may reside in memory or be streamed to hard disk. Alternatively, the digital signal on the PC bus may be routed out to the digital output. Several analog inputs may exist such as: Line Input, Microphone Input, Auxiliary Input, CD Drive Input, Telephone Modem (TDA) Audio, and others.

### 3.1.4 Record-Play Loop (A-D-PC-D-A) Path

This is a combination of the Record and Play paths through the PC bus. The signal path is from the analog input amplifier, through the Record Mixer, through the analog to digital converter to the PC bus, through the digital to analog converter, the Playback Mixer, and back out through the analog output amplifier.

### 3.1.5 Digital Loop (D-D) Path

The path from a digital input, through the PC bus, and returning out via the digital output. These inputs and outputs are as specified in IEC 60958-3 and also sometimes called S/PDIF. Note that AES-3id specifies an unbalanced digital interface, also with a 75 Ω impedance, but with professional status bits and 1.0 Vpp signal level. Most consumer sound cards use the 0.5 Vpp signal level specified by IEC 60958-3.

### 3.2.1 Record Mixer

The input or record user interface part of the driver associated with a PC audio device. It is mostly associated with the analog paths although occasionally includes some controls for digital paths. This mixer will typically have several faders, one for each analog, and sometimes digital, input. These faders are used to set level for the particular input. Faders are not generally calibrated and it is not usually possible to set specific gain conditions, including unity gain. Typically, associated with each input is an enable or mute check box that will turn the input on or off. In most consumer PC audio devices, these check boxes are mutually exclusive thereby making this “mixer” actually a one-of-several multiplexer.

### 3.2.2 Playback Mixer

The output or playback user interface part of the driver associated with a PC audio device. Like the record mixer, it is mostly associated with the analog paths although occasionally includes some controls for digital paths. This mixer will typically have several faders, one for each analog, and sometimes digital, output. These faders are used to set level for the particular source. Faders are not generally calibrated and it is not usually possible to set specific gain conditions including unity gain. Typically, associated with each input is an enable or mute check box that will turn the input on or off. It is generally possible to have more than one signal source active at a time with relative levels of the active sources set by the respective faders.

### 3.3 Wave file (.wav)

A wave file is a file format for transferring digital audio data between systems of different type and manufacture. It may reside in PC memory or on hard disk that contains a digital representation of an audio signal. The audio signal may be program material or test signals. The header of this file identifies the characteristics of the file including sample rate, stereo or mono, bit depth, etc. This type of file has the extension “.wav”.

### 3.4 Full-scale amplitude (digital)

Root-mean-squared (rms) amplitude of a 997 Hz sine wave in the digital domain whose positive peak value reaches the positive digital full scale, leaving the negative maximum code unused.

### 3.4.1 Decibels, full-scale (dB FS)

Digital signal rms amplitude expressed as a level in decibels relative to full-scale amplitude (20 times the common logarithm of the amplitude over the full-scale amplitude). Note that dB FS expresses a signal level of a digital signal and should not be used to express the signal level of an analog signal.

### 3.4.2 Percent, full-scale (% FS)

Digital signal rms amplitude expressed as a percentage of full-scale amplitude (100 times the amplitude over the full-scale amplitude)

### 3.4.3 Maximum signal level (analog)

0.5 dB lower than the analog signal rms amplitude of the highest undistorted signal level that is possible in an analog signal path. Undistorted in this context shall mean not to exceed –40 dB THD+N (1% THD+N). See also Annex B for additional discussion.

### 3.4.4 Full-scale input voltage (FSIV)

Rms voltage level of the *maximum analog signal level* at the input of an analog-to-digital converter.

### 3.4.5 Full-scale output voltage (FSOV)

Rms voltage level of the analog output signal from a digital-to-analog converter with a digital input consisting of a 997 Hz sine wave input whose positive peak value reaches the positive digital full scale. (Note: some devices may not able to handle a signal with an amplitude this high. In this case, 0.5 dB below the maximum signal level that just reaches –40 dB THD+N will be used. See the explanation in Annex B)

### 3.4.6 Reference signal level

Signal amplitude used for ratio-metric computations such as dynamic range and signal-to-noise-ratio. A Reference Signal Level of 0 dBr will be the Maximum signal level or, if it can be achieved, Full-scale amplitude.

### 3.4.7 dBr

A signal level referred to a reference signal level. Used for measurements involving ratio such as dynamic range. 0 dBr for dynamic range measurements is determined as specified in the maximum signal level definition.

### 3.5 Frequency response

Variation in output rms signal level, relative to the level at 997 Hz, as the frequency is varied

> Note: Amplitude limits or amplitude corners are required to define frequency response and are usually either ±1 dB or ±3 dB.

### 3.6 Total Harmonic Distortion plus Noise (THD+N)

20 times the logarithm of the ratio of the rms amplitude of all signal harmonics plus noise within the measurement bandwidth to the rms amplitude of the test signal expressed in dB.

> Note 1: It is important to include all non test-signal frequencies, not just multiples of the test signal frequency, since converters may generate aliased components anywhere in the measurement frequency band. The THD+N measurement is easier to perform than THD, since the only requirement is to filter out the test frequency and then perform a broadband measurement of the residual, rather than perform a spectral analysis (see annex D.4).

> Note 2: THD+N measurements are very dependent on test signal amplitude; therefore, the test signal amplitude relative to Reference Level should be listed with THD+N measurement results.

### 3.7 Intermodulation Distortion (IMD)

A measurement of the distortion produced by the interaction of two or more test signals. The most common technique is to use a test signal composed of two sine waves and measure the sum and difference products produced. The two sine waves are typically near the upper band edge, commonly 18 kHz and 20 kHz (for analog circuits) or 17,987Hz and 19,997 Hz (for digital circuits). The difference product at 2 kHz (2.02 kHz digital) is measured and its amplitude as a ratio of the average value of the two high frequencies expressed as the magnitude of IMD. This technique, unlike THD+N, permits valid distortion measurements near the upper band edge of a device.

### 3.8 Dynamic range (DR)

20 times the logarithm of the ratio of the Reference Level to the rms noise floor within the measurement bandwidth, expressed in dB. The measurement must be made in the presence of signal.

> NOTE: This specification is sometimes referred to as signal-to-noise ratio (SNR) in the presence of a signal. The label SNR should not be used because of industry confusion over the exact definition. SNR is

often used to indicate signal-to-noise ratio, with the noise level being measured with no signal. This can often give an optimistic result because of muting circuits, which mute the noise when no signal is present.

### 3.8.1 A-weighted dynamic range (DRA)

20 times the logarithm of the ratio of the signal Reference Level to the A-weighted rms noise floor in the presence of signal, expressed in dB.

### 3.8.2 ITU-R BS.468-4-weighted dynamic range (DR-ITU)

20 times the logarithm of the ratio of the signal Reference Level to the ITU-R BS.468-4-weighted rms noise floor in the presence of signal, expressed in dB (see annex D.3)

### 3.9 Continuous power

Average power into a specified load using a 997-Hz sine wave played at the full-scale output voltage

NOTE: This specification is only required when the line output jack is used to drive headphones and loudspeakers directly.

### 3.10 Cross-talk between signal channels

Leakage of information from one channel to another, expressed as 20 times the logarithm of the ratio of the amplitude of the test signal applied to one channel versus the amplitude of the leakage of the test signal into an undriven channel with its input terminated. The measurement should be a narrow-band level measurement to exclude broadband noise. A one-third octave filter tuned to the test signal stimulus is an example of a narrow band measurement.

### 3.11 Interchannel Phase Difference

Interchannel phase response is the phase difference between a designated reference channel and all other channels. The phase differences between every other channel and the reference channel shall be reported in degrees as a function of frequency.

### 3.12 Passband Ripple

A measurement of the amplitude variation caused by anti-aliasing and reconstruction filters associated with analog to digital and digital to analog converters. See Annex E for additional details.

### 3.13 Sampling Frequency Accuracy

Comparison of the actual sampling frequency to the theoretical sampling frequency expressed in Hz or percent. For example, if the theoretical sampling frequency is 48,000 Hz but the measured sampling frequency is 48,001.45 Hz, the error is 1.45 Hz or 0.00302%.

### 3.14.1 Interface jitter

Deviation in timing of interface data transitions (zero crossings) expressed in Unit Intervals (UI) when measured with respect to an ideal clock. See AES3 and IEC 60958-1 for a definition of Unit Interval.

### 3.14.2 Intrinsic jitter

Output interface jitter of a device that is either free running or is synchronized to a jitter-free reference. Expressed in UI. See AES3 and IEC 60958-1 for a definition of Unit Interval.

## 4 Measurement techniques

This section is a general description of measurement techniques for all of the measurements defined in this document. Included are recommendations for obtaining most accurate results, discussions of possible sources of errors, and suggestions for how to avoid these errors.

For specific information on how to perform tests on PC audio devices including a step-by-step procedure, see Section 6.

> NOTE: These measurement techniques are based on the normative reference AES17 and the informative references *Audio Precision Audio Measurement Handbook* and the EIAJ CP-307 CD measurement standard (see annex A).

All digital test signals should be properly dithered. The dither signal should be a random or pseudorandom sequence having triangular probability density and a peak-to-peak amplitude of 2 least significant bits (which is ±1 LSB) of the digital audio word length. The amplitude of the noise should be flat, that is constant per unit bandwidth (white) to at least the upper band-edge frequency.

### 4.1 Electrical specifications

To completely characterize a PC audio device, the following measurements shall be made

a) Determination of reference levels,

b) Frequency response (FR),

c) Total harmonic distortion plus noise (THD+N),

d) Dynamic range (DR),

e) (optional) Intermodulation distortion (IMD),

f) Interchannel crosstalk,

g) Interchannel phase difference,

h) Continuous power (if a speaker or headphone output is provided),

i) Noise level during system activity;

j) Sampling frequency accuracy,

k) Passband ripple,

l) Microphone bias voltage and current (if a microphone input is provided),

If a digital input, digital out, or both are present:

m) Digital carrier level,

n) Digital output source impedance,

o) Digital input impedance,

p) Verification of status bits,

q) Tests of digital data bit transfer accuracy,

r) Jitter.

Additional optional specialized tests

s) Test for “glitches”,

t) MIDI latency.

## 4.2 Analog Input and Output Conditions

### 4.2.1 Signal Levels

Each analog input and output should have an optimal operating signal level that achieves the best compromise between distortion (at high levels) and noise (at low levels). This level is sometimes informally called maximum operating level or full scale level. To avoid overload or clipping, average program signal level is typically some 10 to 20 dB below this level.

Legacy consumer audio equipment has peak line levels of 2 Volts rms. That is, the maximum signal output from a tuner, VCR, cassette player, or CD player is typically 2 V rms. PC Audio devices on the other hand often operate at lower power supply voltages and thus are often unable to deliver or accept signals levels as high as 2 V rms, since this level represents a peak to peak amplitude of over 5 V. This is particularly true for laptop computers. This has led to variances in common practices. Many older standards and recommendations having their origins with earlier consumer audio equipment specify a line input or line output maximum signal level of 2 V while some newer PC Audio recommendations or common practices suggest a 1 V maximum level. For the purposes of characterizing the performance of a PC Audio device, either the 2 V or 1 V level can be used but the level should be specified.

Microphone input circuits have a wider level range of operation as they must accommodate a wide range of microphone sensitivities. A typical PC microphone may produce a signal level of approximately 10 mV with a normal speaking voice close to the microphone. However, the microphone input circuit should be able to accept a maximum input signal level of 100 mV rms to handle high level microphones. Many circuits employ a “mic boost” feature that will allow an additional 20 dB of gain to be inserted. This is a convenient way of being able to accommodate the wide range of signal level present on such inputs.

### 4.2.2 Impedances

Analog input circuits have characteristic input impedances and are designed for optimal operation from low source impedances. Analog outputs typically have a low source impedance and will operate properly with a limited range of load impedances from several times the output source impedance to infinity. Better performance is achieved with lower source impedances and higher load or input impedances. This will reduce loading effects and minimize the effect that cable capacitance will have on frequency response. The following table indicates typical and recommended impedances for inputs and outputs of PC Audio devices and the external devices connected to these inputs and outputs.

| | |
|---|---|
| Line Input Impedance | > 40 kΩ |
| Line Output impedance | < 100 Ω |
| Microphone Input impedance | ≥ 2 kΩ |
| Line Input Source impedance | < 2 kΩ |
| Line Output Load impedance | > 20 kΩ |
| Microphone Source impedance | ≤ 600 Ω |
| Headphone Load impedance | ≥ 32 Ω |
| Speaker Load impedance | 8 Ω |

## 4.3 Sample Rates

For digital circuits, the preferred internal sample rate is 48 kHz, 96 kHz, or 192 kHz. External devices can have other sample rates that must be accommodated. These are typically handled by using sample rate converters. The common sample rates in use are: 8 kHz, 11,025 Hz, 16 kHz, 22,050 Hz, 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, and 192 kHz.

## 4.4 Test conditions

The general conditions for measurement shall be:

a) Measurement bandwidth 20 Hz to 20 kHz (for which the cutoff of the frequency response below 20 Hz and above 20 kHz may be specified according to IEC60268-1 and ITU-R BS.468-4 for unweighted, band-limited measurements);

b) Standard test signal frequency shall be 997 Hz;

c) Preferred system sampling frequency shall be 48 kHz, 96 kHz or above (44.1 kHz may also be used);

d) Mixer settings shall be set as determined during the establishment of Reference Level process, with only the test channel unmuted.

Electrical specification measurements shall be made according to the methods in this document.

**4.5 Establishing reference levels**

The first step in any measurement activity is the establishment of *reference levels*. This provides a point of reference to set test signal levels and is used in calculating ratio-metric measurements such as dynamic range. A specified method of establishing reference levels allows consistency and repeatability of measurements.

Historically, a common reference point has been full scale level, the maximum level that a circuit can accommodate or the clipping point. In many PC audio devices, it is difficult or even impossible to accurately determine a reference point based on these criteria. See Annex C for additional discussion and example illustrations. In summary, the analog to digital transfer curve near digital full scale may be non-linear and affected by limiting circuitry built into the PC audio device.

To avoid the ambiguities that might otherwise occur, a precise process is defined to be able to determine a repeatable reference level for each of the paths in a PC audio device. This generally involves determining a maximum level according to specified methods.

The reference level established for a particular analog path is then defined as “0 dBr”. For a pure digital path, that is, one with no analog components, the reference level should be the maximum positive code. This is defined as 0 dB FS. Subsequent measurements can be made relative to this established reference. For example, frequency response is measured with a test signal level of –20 dBr or –20 dB FS. Dynamic range is a measurement of the ratio of the absolute value of noise compared to 0 dBr or 0 dB FS.

**4.6 Frequency response (FR)**

Frequency response shall be tested over the audio bandwidth of 20 Hz to 20 kHz. Spot measurements may be used at octave or smaller intervals. The relative magnitude shall be set to –20 dB FS (for D to A paths) or –20 dBr (for analog paths). The level shall be measured at 997 Hz and set to 0 dB on the response plot in the manner shown in Figure 1. All other frequency measurements shall be made relative to the 997 Hz measured amplitude. The FR shall be listed in the form: FR = + *x* – *y* dB from *n* Hz to *m* kHz. For example, in Figure 2, the FR is + 0,5 – dB from 40 Hz to 19 kHz. ($F_{cl}$ is the lower cutoff and $F_{ch}$ is the upper cutoff.)

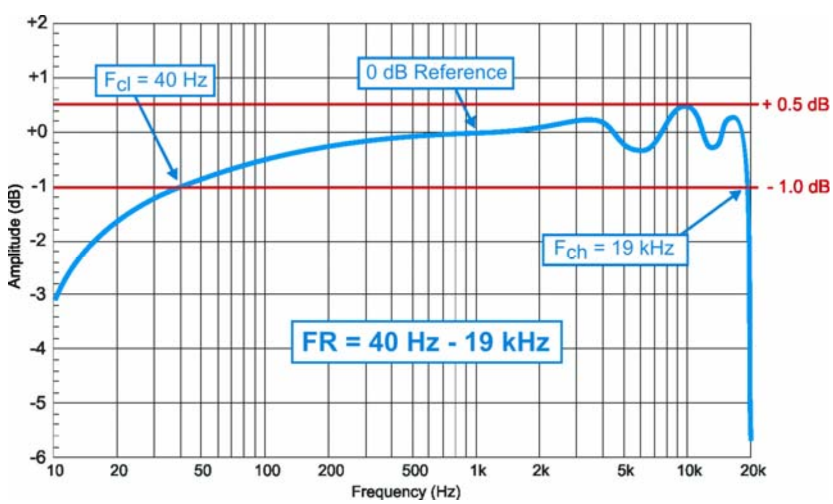



**Figure 1 — Frequency response plot**

NOTE The level of –20 dB FS is chosen to minimize distortion components, which could affect the FR measurement. Other test signal amplitudes may be used, provided that the signal level is such that no significant distortion components are generated.

### 4.7 THD+N

Total Harmonic Distortion plus Noise is a ratio metric measurement that expresses the rms sum of all harmonics and noise within the measurement bandwidth relative to the rms amplitude of the test signal. The test signal level will have an effect on the measured THD+N. If the test signal level is too high, the device may be too close to clipping or saturation and produce high readings. If the test signal level is too low, the noise floor of the device will begin to dominate the reading. AES17 specifies that THD+N measurements should be made with a test signal level of -1 dBr while PC-2001 specifies a test signal level of -3 dBr. Either level may be used but the value used should be specified. Of course, the step of determining 0 dBr must be followed carefully.

In a system with low distortion, the absolute THD+N value tends to be similar to the DR value. In a system that exhibits significant distortion, the absolute THD+N value can be worse than the DR value. This is an indication of the presence of distortion.

Measurement bandwidth is an important part of a THD+N measurement. Since the measurement includes a summation of all the energy within the measurement bandwidth, the width of this band will determine the amplitude of the noise and upper harmonic contribution to the measurement. This is particularly true if the system includes a large amount of out-of-band noise, a common situation with systems employing delta sigma converters. Many audio distortion analyzers have a total unconstrained measurement bandwidth of 200 kHz to 500 kHz or more. They also generally include low pass filters that can be engaged to constrain the measurement bandwidth to a much smaller range, typically 20 kHz, 22 kHz, 30 kHz or 80 kHz. For accurate THD+N readings, the measurement bandwidth should be constrained to 20 or 22 kHz or (30 kHz if a 20 or 22 kHz filter is not available). Note that noise-weighting filters should not be used for THD+N measurements.

The effective bandwidth of the device under test and the measurement instrument set to an audio-band-constrained measurement bandwidth will limit the practical range of test frequencies that can be used for a THD+N measurement. All PC audio devices include anti-aliasing and reconstruction low-pass filters that typically roll off above 20 kHz. To obtain a useful THD+N measurement, at least the second harmonic and preferably also the third harmonic should be measured. For a 20 kHz band-limited system, this would infer that the highest test frequency (fundamental frequency) that should be measured would be approximately 6.7 kHz. If a test frequency greater than this is used, the harmonics will be rolled off by the anti-aliasing or reconstruction filters in the device rendering the THD+N measurement only a measurement of noise. Thus the THD+N technique cannot be used to characterize distortion performance at higher frequencies, and attempts to make measurements beyond about 6 kHz will be meaningless. Intermodulation distortion measurement techniques are more appropriate for characterizing the distortion performance at higher frequencies. See IMD, below.

THD+N should be reported in dB. Alternatively, the reading may be reported in percent. It is the ratio of the sum of all harmonics and noise in the measurement bandwidth relative to the test signal level. The amplitude of the test signal level should also be reported.

EXAMPLE A measurement result using the line-in record path should be listed in the form

THD+N (997 Hz, –3 dBr) = – 85 dB

This example indicates that the test signal is an analog 997 Hz sine wave, at –3 dB relative to the full-scale input amplitude at the line input. The THD+N result is –85 dB relative to the measured digital signal amplitude. Since the standard test signal frequency of 997 Hz is used, it can be omitted from the listed result.

### 4.8 Dynamic range (DR)

Dynamic range is a measurement of the noise in a device. It is similar to the traditional signal-to-noise measurement but with some key differences appropriate to PC audio devices. A signal to noise measurement supplies no signal to the input of the device, terminates the input and measures the output noise. In a PC audio device, this technique may result in an artificially low noise figure since certain internal circuits are automatically shut down in the absence of signal. To overcome this problem, a low level pilot signal is fed to the device and this pilot signal is removed at the output prior to measuring noise. The level of the pilot signal is sufficiently low that any distortion products it may introduce are well below the noise and will not contribute to the noise reading.

A simple way to make a dynamic range measurement with a pilot signal is to make a THD+N measurement using a low level stimulus signal. This test signal amplitude is small enough to be unaffected by any large signal non-linearity, but large enough to ensure that the system under test is being exercised. A classical distortion analyzer will notch out the fundamental test signal using a band reject filter and measure everything remaining. The magnitude of this residual is expressed relative to the amplitude of the test signal. Other test signal amplitudes may be used, provided that the signal level is such that no significant distortion components are generated. For systems with a bit resolution of greater than 10 bits, the test signal level should be -60 dBr. For systems with a bit resolution less than 10 bits, a test level of -30 dBr can be used.

Dynamic range is expressed as the ratio of the rms amplitude of the noise within the measured bandwidth relative to the reference level, 0 dBr. A THD+N meter will typically compute this ration relative to its test signal level, which is –60 dBr (for high-resolution converters). Thus, dynamic range should be expressed as the ratio of the THD+N reading relative to 0 dBr. For example, if the THD+N reading at 997 Hz and –60 dBr is –32 dB, then the dynamic range is 92 dB.

If the THD+N measurement instrument is able to provide absolute measurements relative to 0 dBr, then this absolute reading can be used directly.

### 4.8.1 Types and sources of noise signals

Noise measured by the dynamic range measurement technique described above can come from a variety of sources. Many of these are steady state and are caused by the circuits themselves. These may be measured with a relatively short test interval. PC Audio devices however introduce additional transient noise sources that are more difficult to characterize. These may be caused by other devices in the PC such as hard disk drives or even nearby monitors. To evaluate the contribution of these devices, the dynamic range test should be conducted for a longer time and other components should be exercised during this time. Some of these noise components may appear as short duration transients. The measurement instrument should have the ability to make a sustained measurement over a user-defined time interval and record the duration and amplitude of all short term noise transients.

### 4.8.2 Weighted dynamic range measurements.

It is common to provide frequency weighting to noise and dynamic range measurements. Human hearing is less sensitive to low and high frequencies at low sound pressure levels. Since noise is generally at a low sound pressure level in a typical listening situation, weighting the spectrum of noise measurement, giving low and high frequencies less influence on the final measured value will better approximate the perceived noise.

There are two noise weighting filters in common use: A-weighting is popular in North America and ITU-R468 (formerly called CCIR-468) is popular elsewhere. See Annex D for a complete description of both filters.

When a weighting filter is used, the report of the dynamic range measurement should specify this information. For example:

DR = 85 dB, A-weighted

Or

DR = 85 dB, ITU-R BS.468-4 weighted

Note that the reading is a positive or unsigned number with dB units.

### 4.8.3 Signal-to-Noise Ratio (SNR)

Signal to noise ratio is a traditional measurement of the noise in an analog circuit. The result is specified as a ratio, typically in dB. The “signal” part of the ratio is the reference level defined for the particular measurement. Depending on the measurement situation, it may be just below the clipping point or a specific operating level or, in the case of a power amplifier, a specific output power level such as 1 Watt. The measurement is usually made by shorting the input or terminating the input with a low and defined impedance.

Most circuits employing analog to digital converters including muting circuits that will shut down part of the circuit with no input signal. For this reason, traditional SNR measurement techniques should not be used for PC audio devices when converters are part of the path. Simply terminating the input will result in an incorrectly low and erroneous noise reading. Dynamic Range is the correct method to measure noise in PC audio devices.

The only exception to this is the analog-to-analog (AA) path that usually has no muting circuits and behaves like a typical analog-only circuit. For comparison purposes, SNR should be measured for this path only.

### 4.9 Intermodulation Distortion (IMD)

Intermodulation distortion is measured using two (or more) test signals. For audio purposes, two test signals is most common. Historical techniques used a low frequency, typically between 40 and 60 Hz, and a high frequency, typically between 6 kHz and 8 kHz mixed in a four to one amplitude ratio. That is, the low frequency is 12 dB higher than the high frequency signal. While this technique was suitable for the devices being measured at that time such as optical film sound tracks and later analog tape recorders, it is not very useful for contemporary equipment. A more recent technique, that is also a more indicative indicator of audible performance, is to use two equal amplitude, closely spaced, high frequency tones. Distortion components will be produced as sum and difference products of the two test signals and sum and difference products of the harmonics of the two test signals. The difference components will fall at frequencies below the test signals and thus in the audio band. This IMD measurement technique has been called "Difference Frequency Distortion" and also "CCIF" IMD after the standards organization that recommended this technique. The following table illustrates how the second, third, fourth, and fifth order difference components fall within the audio band. Although only these difference components fall within the audio band, the sum components that occur at frequencies above the audio band can also have an effect. These can alias with other signals and produce artifacts that then fold down into the audio band. Therefore, to make a complete IMD measurement, the complete audio spectrum, excluding only the two nominal 18 kHz and 20 kHz test signals, should be measured.

| | | |
|---|---|---|
| F1 test signal | | 20 kHz |
| F2 Test signal | | 18 kHz |
| F1 – F2 | 2nd order | 2 kHz |
| 2F2-F1 | 3rd order | 16 kHz |
| 2F1-2F2 | 4th order | 4 kHz |
| 3F2-2F1 | 5th order | 14 kHz |

**Typical IMD test signals and the resulting distortion products**

The IMD measurement technique has an advantage over the THD+N measurement technique in band-limited systems as it is able to characterize distortion at high frequencies right up to the upper band edge of the device. Conversely, THD+N is only valid for measurements at frequencies below approximately one third of the bandwidth of the device under test. Since most PC Audio devices have a bandwidth that rolls of sharply beyond 20 kHz, the IMD measurement technique is the only way to characterize distortion in the range from approximately 6 kHz to 20 kHz.

### 4.10 Continuous power

The measurement of continuous power is only relevant if the PC audio device includes power amplifiers designed to drive external loudspeakers or headphones. Most PC audio devices manufactured since the early 1990's do not include such power circuitry and provide only line-level outputs designed to drive external power amplifiers. Line outputs do not need to be tested for continuous power.

Testing of power output stages requires that an appropriate load be connected. The load shall be specified with the measurement data.

NOTE The load is typically 8 Ω for loudspeakers and 32 Ω for headphones.

If the full-scale output under headphone loading conditions (32 Ω) is 1.2 V, the continuous power would be:

$$\text{Continuous power} = \frac{1.2^2}{32} = 45\ \text{mW}$$

This result should be listed in the form:

Headphone continuous power (32 Ω) = 45 mW per channel

For loads other than line level, the load shall be specified with the test results. Outputs designed for more than one function should be tested with each appropriate load.

NOTE Output full scale is similarly defined to input full scale with one exception — the load can be important. Outputs are designed to drive three types of loads: line level, headphones, and loudspeakers. Outputs designed for line level usually have minimal drive capability and the load should be set above 10 kΩ. For outputs designed to drive headphones and loudspeakers, the load used for making measurements is very important and should reflect what the end user would experience. Headphone impedances vary from

16 Ω to 92 Ω nominal, with 32 Ω being the most common. Loudspeaker loading is set at 8 Ω . As shown in Figure 2, some analyzers allow the output signal to be swept in amplitude, which shows the clipping point very well. Figure 2 depicts the THD+N of an output designed for both headphone and line-level drive under proper loading conditions. When the output is loaded, the full-scale output changes, illustrating the importance of proper loading when making measurements with analog outputs.

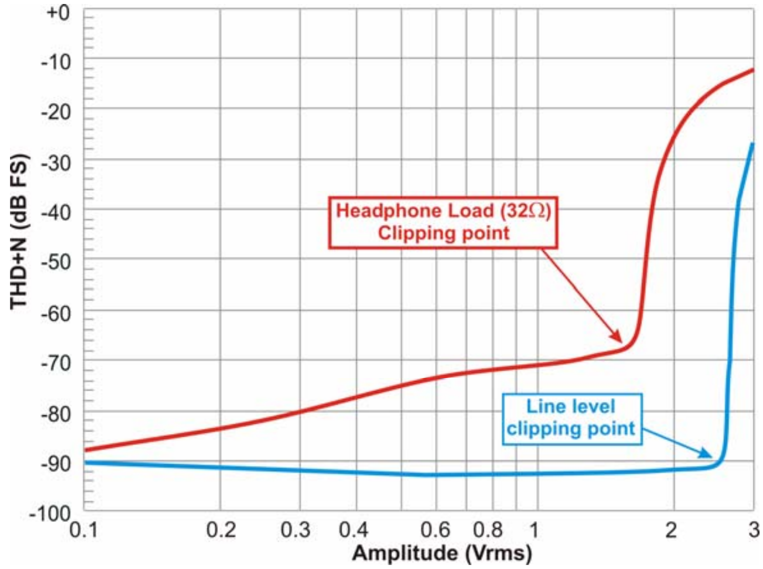



**Figure 2 — Headphone clipping point**

### 4.11 Cross-talk between signal channels

Cross-talk is leakage between signal paths. Crosstalk can occur in multiple situations, for example: left to right or right to left on any stereo input pair or stereo output pair, input to output cross-talk, and output to input cross-talk. The measurement technique is the same for each of these measurements, with the appropriate signal path connections made. The test signal amplitude shall be 0 dB FS (digital) or –3 dBr (analog). The inputs to all other channels shall either be terminated to signal ground via a 50 Ω resistor (for analog inputs), or driven with digital zero (for digital inputs). The output signal from each undriven channel shall be analyzed for the presence of the test signal frequency. The amplitude of the test signal shall be measured and expressed as a dB ratio to the test signal amplitude. This may be done by using a narrow band filter (such as a third octave filter) to measure the undriven channels, or by performing an FFT analysis. The test frequency should be varied in steps of less than one octave, and the measurements repeated. The result can then be plotted as a graph of cross-talk amplitude vs. frequency. If a single spot frequency is used, it shall be 10,007 Hz.

### 4.12 Interchannel phase response

Phase difference between channels can affect stereo imaging and cause comb filtered response effects. Extreme phase differences or polarity inversion can cause signal cancellation in monaural outputs or the “hole in the middle” effect in stereo. Phase difference between stereo channels and between surround channels should be measured at several frequencies, particularly at high frequencies, that is, above 5 kHz. Measurements should be made at octave intervals or better up to the upper cutoff frequency of the device. If phase difference is measured at only one frequency, it should be measured at 10,007 Hz.

### 4.13 Noise level during system activity

The purpose of this test is to measure any degradation in the noise floor of the audio playback or record path during the activity of the mouse, keyboard, display, disk drive, or CD-ROM drive. One way to perform the measurement is to repeat the DR measurements of interest, while forcing activity of the peripheral device in question. If any significant degradation in DR is noted, it can be verified by listening tests. See also Glitch detection.

### 4.14 Frequency accuracy

When a PC is used by musicians for recording and playback of music, the accuracy of the sampling frequency becomes important. If the sampling frequency is not quite correct, then pieces recorded on the PC become pitch shifted when played back on a different system. Similarly, playback of pre-recorded material can be pitch shifted.

### 4.14.1 Sampling frequency accuracy on playback (PC-D-A)

A test data file containing a 997 Hz ±25 ppm sine wave shall be recorded with a sampling frequency of 48 kHz ±25 ppm. This data file shall be played back using the PC under test. The frequency of the 997-Hz sine wave shall be measured using a frequency meter with an accuracy of better than ±10 ppm. The deviation from 997 Hz shall be expressed as a percentage.

### 4.14.2 Sampling frequency accuracy on record (A-D-PC)

A 10-second period of 997 Hz ±25 ppm sine wave shall be recorded with a sampling frequency of 48 kHz. This file shall be played back using a system whose sampling frequency is known to be 48 kHz ±25 ppm. The frequency of the 997-Hz sine wave shall be measured using a frequency meter with an accuracy of better than ± 10 ppm. The deviation from 997 Hz shall be expressed as a percentage.

## 4.15 Passband ripple

PC audio devices include converters and thus anti-aliasing and reconstruction low-pass filters. High order filters can exhibit ripples in the frequency response near the cutoff frequency. Passband ripple expresses the magnitude of this ripple. Early converter designs used analog filters with cutoff frequencies near the upper audio band, that is approximately 20 kHz. Newer converter designs, in particular over sampling converters, inherently achieve lower ripple, in most cases below other response variations. For this reason, there is less need to characterize ripple on new designs. However, the technique is explained here since popular practice often requires this measurement.

A measurement of passband ripple starts with a frequency response measurement of the high frequency band near the cutoff region. This response measurement will include the roll off characteristics of the filter and the device in addition to the ripple. The ripple portion should be isolated so only the ripple can be reported.

Make a frequency response measurement with sufficient resolution to capture the ripple. A linear frequency axis work best with a resolution of at least 1 kHz steps. If the resulting response data is smoothed, the ripple will be removed leaving only the gradual roll off. This roll off can be removed from the original response to leave only the ripple. Find the mean value of the ripple excursions and establish this as 0 dB. The ripple should be expressed as ±x dB relative to this 0 dB mean value. See Annex E for a more complete discussion of this subject including examples and a description of the measurement process.

## 4.16 Microphone bias voltage and current

*(if a microphone input is provided)*

Microphone inputs on PC audio devices must provide power for electret microphones. The connector is typically a tip-ring-sleeve 3.5 mm connector. The microphone input is typically a single channel (mono) with signal between tip and sleeve. The bias voltage is supplied between ring and sleeve. It should have a maximum value of 5.5 V dc unloaded and be able to supply a current of at least 0.8 mA at a minimum of 2 V dc when loaded with a 2.5 kΩ load.

## 4.17 Digital interface measurements

If a digital output, digital input or both are provided, certain characteristics should be measured. The digital input or output should conform to the IEC 60958-3 specification which states that the impedance should be 75 ohms and signal level should be 0.5 Vpp. (Note that AES 3id also specifies a similar unbalanced interface with a 75 ohm impedance but a signal level of 1.0 Vpp). The following characteristics of this interface should be measured.

### 4.17.1 Digital carrier level

The peak to peak value of a digital output should be measured. The output should be terminated with 75 Ω for this measurement.

For digital inputs, the signal supplied from the test generator should be 0.5 Vpp sourced from a 75 Ω source impedance.

### 4.17.2 Digital output impedance

The digital output source impedance should be 75 ohms. Its actual impedance can be inferred by measuring the output signal level open circuit and with a 75-ohm load. The source impedance can be determined using the following formula:

$$Z_s = 75 \frac{V_o}{V_t} - 75$$

Where $Z_S$ = the source impedance of the digital output

$V_O$ = Peak to peak output voltage open circuit

$V_t$ = Peak to peak output voltage terminated in 75 ohms

### 4.17.3 Digital input impedance

The digital input impedance should be 75 ohms. The actual input impedance can be inferred by noting the level difference between the open circuit (unterminated) level of the test generator compared to the test generator level when connected to the digital input. The test signal generator should have a source impedance of 75 ohms. The device digital input impedance may be determined by the following formula:

$$Z_i = \frac{75V_t}{V_o - V_t}$$

Where $Z_i$ = the input impedance of the digital input

$V_O$ = Peak to peak output voltage open circuit

$V_t$ = Peak to peak output voltage terminated in 75 ohms

### 4.17.4 Status bit verification

A digital signal conforming to AES 3 or IEC 60958-3 must include status bits with particular values. For example, a consumer PC audio device should indicate “consumer” in the status bits. For a digital output, each of the status bits should be examined for correct value. For a digital input, the test generator should generate the correct status bit values and observe correct behavior of the device. For a digital input that uses a Sample Rate Convertor (SRC), also referred to as a Sample Frequency Convertor, the digital output status bits should correctly contain the code indicating an SRC is used.

### 4.18 Glitch detection

A PC audio system shares resources with other PC hardware and programs. Occasionally the total demands of all programs may exceed the hardware’s ability to process requests resulting in delays, hesitations, dropouts, signal repeats, etc. PC audio systems include error correction facilities to minimize the perception of such unwanted artifacts in an audio delivery.

While the consequences of overtaxed resources to non-audio programs may be tolerated as just sluggish behavior or delayed reaction, the consequences to an audio program delivery may be intolerable. These signal dropouts, repeats, or stutters have been called “glitches”.

One method to characterize a glitch of this nature is to use a distortion analyzer as a “magnifier” of signal disturbance. A typical audio distortion analyzer includes an automatically tuned notch filter with two sophisticated servo systems constantly monitoring the level and phase of the signal. They are very sensitive to changes in either of these parameters and even very slight, sometimes audibly imperceptible changes will cause significant disturbances to the servo loops and be very obvious at the residual output of the analyzer.

By playing a steady state wave file of a sine wave at 10,007 Hz and monitoring the signal level of the “distortion” residual output, very subtle disturbances in delivery can be detected. This can be used to test the system’s ability to share resources and the effect that non-audio demands have on audio delivery. With this test running, other programs can be run to provoke disturbances.

### 4.19 Latency

While not strictly an audio measurement issue, latency is included here because of its importance. This is a measure of the time delay through the device. It can be signal path delay such as may be experienced through a converter or control delay such as may be the case in a MIDI control.

4.19.1 MIDI LatencyThis is a measure of the time delay between a MIDI command sent to the PC audio device and the generation of the requested sound. Most PC audio devices include a synthesizer able to generate musical sounds in response to MIDI commands sent to a MIDI interface included on the device. When a command is generated externally, for example by a musician playing a MIDI keyboard controller, the expectation is that the requested note will be audible virtually instantly. Any perceived delay will be not tolerated by a musician. This should be tested by generating defined MIDI commands, starting a timer at the time of command initiation, detecting the onset of the requested MIDI note, and stopping the timer. This time, typically tens of milliseconds, should be reported.

### 4.19.2 Converter Latency

This is a measure of the signal delay through an analog to digital, digital to analog converter, or both converters cascaded for analog to digital and returning through digital to analog. Digital signal processing in the path may introduce delays. When the device is part of a system, differential delays between systems or between audio and video systems can cause unacceptable results.

## 5 Test equipment

This section describes the test equipment and interface of the test equipment to the device under test. It includes requirements of the test setup to assure accurate and repeatable results.

### 5.1 Validating the test system

To have confidence that measured results are accurate, the test equipment and test setup need to have the facilities and accuracy to perform the tests described in this document. Equally important, the test setup and interconnections between the test instruments and the device under test need to be high quality. This includes cable quality, grounding, and reduction of interference. See the following sections for additional discussions on these subjects.

The audio equipment under test (EUT) can only measure as well as the test system. Therefore, the limitations of the test hardware shall be known before testing the EUT. If external test equipment is used, it should be looped back on itself using good, shielded cables, with the cables positioned near the EUT. Putting the cables near the EUT can show any degradation caused by poor shielding on the cables. If the test system is not significantly better than the expected EUT results, then the EUT measurements cannot be accurate. For example, a test system measuring 10 dB better than the EUT produces a measurement of the EUT 0.5 dB worse than it actually is. If the test system has the same performance as the EUT, then the EUT can measure 3 dB worse than it actually is. Once the performance of the test system is known, the ground of the test system shall be connected to the ground of the EUT.

### 5.2 Proper grounding

Ground loops cause many problems in audio testing. If the performance gets worse when the grounds are connected, grounding problems exist and shall be resolved before EUT testing can begin. The spectrum analysis of noise shown in Figure 3 illustrates the results of an EUT and external test equipment with the grounds between the systems connected two different ways. The upper set of data contains harmonics of the power line frequency (60 Hz) which are artifacts of improperly configured grounds. The bottom set of data has the grounds connected properly, which improved the performance by 16 dB in this example. Good grounding involves minimizing ground loops. If external test equipment is used, it should be plugged into one power strip. That equipment power strip should be plugged into the same outlet or power strip as the system under test. The bad grounding example in Figure 3 was generated by using two widely separated wall sockets, one for the test equipment and one for the EUT.

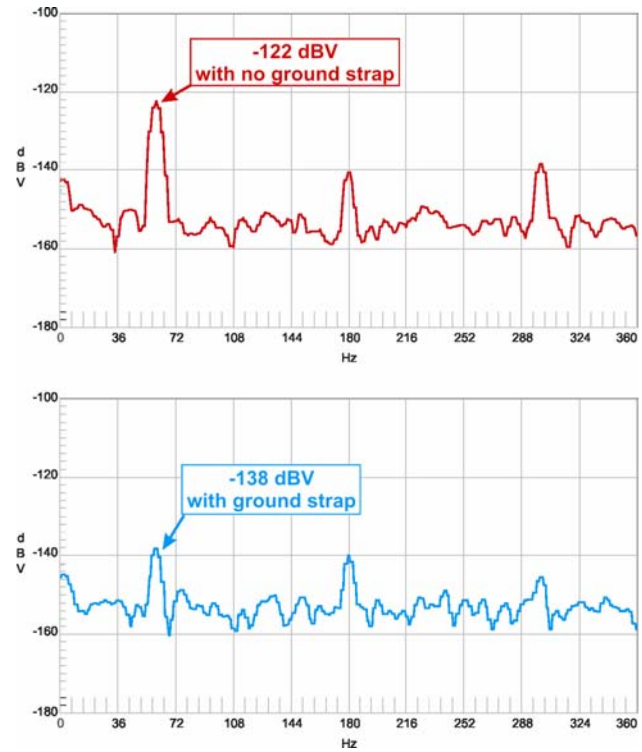



**Figure 3 Test system grounding example**

A robust, low impedance ground connection between the EUT and the test equipment is essential to minimizing power line interference as shown in the upper graph of Figure 3. The typical audio cables and their 3.5 mm connectors used to provide signal connections to most PC audio devices provide insufficient ground integrity. To minimize the flow of equalizing ground current through signal cable shields, a supplemental ground cable is often necessary. This should be a heavy gauge wire, number 12 AWG or better, stranded or braided wire with a means to connect to a good ground lug on the test equipment at one end and the metal chassis of the PC audio device at the other end. A large spring-loaded fastening clip (alligator clip) for the PC end usually works well.

### 5.3 Out of band noise

In PC audio devices digital-to-analog converters frequently use delta-sigma modulation and oversampling techniques to achieve high performance at a low cost. The noise inherent in this type of conversion is pushed up in frequency out of the audio band by means of high-order noise shaping. The resultant signal is low noise within the audio range but carries large amounts of out-of-band noise. Measuring the audio performance of devices with such converters can be a challenge for most contemporary wide-band audio distortion and noise analyzers. These instruments are designed to characterize classic analog audio devices, which typically exhibit a noise floor spread evenly over the full spectrum of the analyzer, usually diminishing with increased frequency. Out-of-band noise and interference are expected to be low in amplitude in relation to the signal, so band-limiting or noise-weighting filters, if needed, are inserted at the end of the measurement path, following several gain stages. When measuring classical analog audio devices, this approach yields accurate and repeatable THD+N results. The spectrum of the noise floor of a noise-shaped delta-sigma D-to-A converter, however, shows a steeply rising energy characteristic beyond the 20 kHz upper limit of the audio band. When measuring low-level signals, the energy contribution of this ultrasonic noise can be substantial. In many situations it can overload instrument gain stages or throw off ranging circuits and cause inaccurate measurements. Conventional band-limiting and noise-weighting filters cannot solve the problem because they are located too late in the measurement chain—the damage has already been done.

AES17 specifies the use of a sharp low pass filter to avoid the problems caused by out of band noise. THD+N and dynamic range measurements should be made with an AES17 low pass filter inserted between the PC audio device and the measurement circuits.

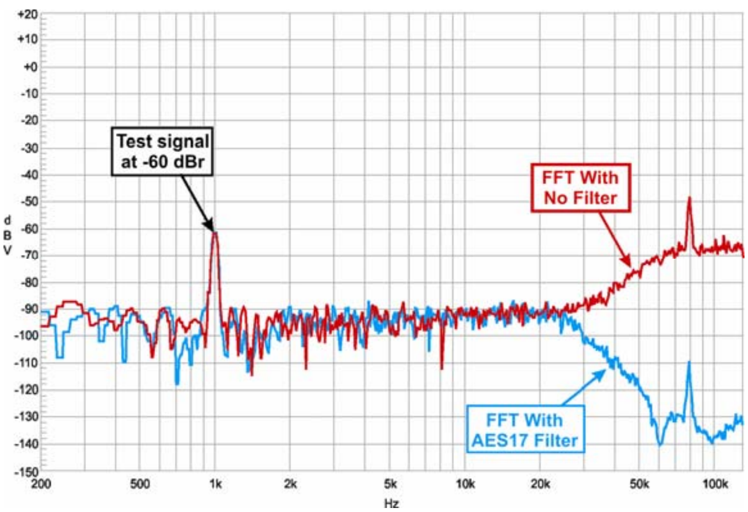



**Figure 4 Spectrum analysis showing out-of-band noise with and without AES17 low pass filter**

### 5.4 Sample rates

Most devices will have an internal sample rate and include sample rate converters to handle external signals with different sample rates. Errors introduced by sample rate converters can affect audio performance. Thus, for completeness, audio performance measurements using digital inputs or digital outputs should be made at all sample rates available on the device under test.

**6.0 Step-by-step procedure to characterize a PC Audio device.**

This section describes in detail the processes and procedures necessary to completely characterize a PC audio device using the measurement principles described in section 4 and test setups described in section 5. It provides a consistent test methodology across different audio systems with different capabilities. Properly setting up the audio subsystem is critical to getting consistent, fair, and repeatable readings.

The procedure characterizes each of the separate paths of a device, beginning with the playback or D to A path. Reference should be made to the signal path diagrams and descriptions in Annex B.

**6.1 Characterize the Play (D-A) Path**

*Note: Play tests may be conducted using a prerecorded .WAV file as the test stimulus or alternatively by feeding a digital test signal directly into the PC bus. Similarly, record tests may be conducted by recording a .WAV file or alternatively by measuring the digital signal directly on the PC bus. In playback cases the signal level on the PC bus is specified and is typically 0 dB FS. In those cases where a 0 dB FS digital signal produces an overload condition (clipping or analog distortion >-40 dB), a test level lower than 0 dB FS may be used. In this case, the highest digital signal level that just produces <-40 dB THD+N is determined. A reference signal level 0.5 dB lower than this maximum should be substituted for 0 dB FS and this alternative test signal level should be specified. The following step-by-step procedure calls for .WAV files in several instances. Direct generation and measurement of digital signals on the PC bus may replace these procedures.*

6.1.1 Connect the PC device as per Annex B.1

**6.1.2 Establish Play Reference Level**

1. Supply a pure digital sine wave at 997 Hz at full scale to WAVE slider input. This may be a WAVE (.wav) file or be directly generated and must be correctly dithered.
2. Set Play Mixer Master control to maximum. If there are additional Group controls set these to the maximum as well. Mute all unused inputs.
3. Increase Play Mixer WAVE fader until the maximum output level is achieved that just produces –40 dB THD+N or maximum WAVE slider setting is reached
4. If it is not possible to achieve less then –40 dB THD+N, lower the Master fader to half way, then try increasing the WAVE fader again to just achieve <-40 dB THD+N.
5. If it is not possible to achieve <-40 dB THD+N at any fader settings, perform the test with a WAVE test signal that is lower than 0 dB FS. Use this lower level WAVE test level as the reference amplitude in place of 0 dB FS for all future play tests.
6. If the WAVE and any group or master sliders are also at maximum and –40 dB THD+N is not reached then this output level is the **Play Reference Level 0 dBr**. If a lower slider setting is required then measure the analog output level that just produces <-40 dB THD+N and establish a reference signal level 0.5 dB lower – this is the Play Reference Level 0 dBr to be used for subsequent relative measurements. This is also the Full Scale Output Voltage (FSOV) when expressed in Vrms.
7. Note the fader position and do not change it for any future measurements.

**6.1.3 Measure THD+N**

1. Supply a pure digital sine wave at a level of -1 dB FS (as per AES17) or -3 dB FS (as per PC2001). For a single frequency measurement, use 997 Hz. For measurements of THD+N vs. Frequency across the full bandwidth, use frequencies from 20 Hz to 0.25 times the sample rate, at least 1 point per octave.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz)
3. With the WAVE fader at the same setting as determined in the Establish Play Reference Level step, measure the THD+N as a ratio to the measured total signal amplitude.
4. Express the result in dB or percent (the reading in dB will be negative).

**6.1.4 Measure IMD**

1. Supply a pure digitally generated two-tone sine wave signal with the amplitudes in a 1:1 ratio with the peak amplitude adjusted to be equal to the full-scale sine wave reference amplitude. Each signal should be 6 dB below the full-scale reference amplitude. If the signal generator output is calibrated in sine wave peak-equivalent amplitude (normally the case) then the generator amplitude should be set to the Full Scale digital signal used in the Establish Play Reference Level step. The frequency of the upper tone shall be 20 kHz for 48 kHz sample rate or 0.42 times the sample rate used. The second tone shall be 2 kHz below the upper tone frequency.
2. Set the WAVE fader at the same setting as determined in the Establish Play Reference Level step.
3. Measure the rms sum of the lower second and third order difference frequency components with an IMD or spectrum analyzer.
4. Express the IMD distortion result in dB relative to the peak amplitude of the total signal (the reading will be negative).

**6.1.5 Measure Dynamic Range**

1. Supply a pure digital sine wave or play a WAVE file with a 997 Hz -60 dB FS dithered test signal.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz). If significant out-of-band energy exists use an AES-17 filter.
3. Optionally choose a frequency weighting: A-weighting with rms detection or ITU-R BS.468-4 and quasi-peak detection.
4. With the WAVE fader at the same setting as determined in the Reference Level step, measure absolute THD+N (typically predominately noise) and express this relative to Play Reference Level (dBr) in dB. (This dB value will be a positive or unsigned number.) If weighting is engaged, dynamic range should be specified as: XX dB A (A weighted) or XX dB ITU-R BS.468-4 (quasi-peak detection).

**6.1.6 Measure Frequency Response**

1. Supply a single tone sine wave or play a WAVE file with test frequencies covering the complete range of the PC device under test, typically 20 Hz to 20 kHz (or 0.4 times the sample rate) and a signal level of -20 dB FS. Frequency spacing should approximate one-third octave spacing across the frequency range of interest. A multitone signal may also be used for this provided that the peak amplitude of the signal is at least 1 dB below the Play Reference Level.

1. The WAVE fader should be at the same setting as determined in the Play Reference Level step. Measure the amplitude at each test frequency.
2. Express the amplitude at every frequency as ±dB relative to the level at 997 Hz.

**6.1.7 Measure Interchannel Phase Response**

*(may be measured at the same time as the Frequency Response)*

1. Supply a single tone sine wave or play a WAVE file with test frequencies covering the complete range of the PC device under test, typically 20 Hz to 20 kHz (or 0.4 times the sample rate) and a signal level of -20 dB FS. Frequency spacing should be at least one per octave. A multitone signal may also be used for this provided that the peak amplitude of the signal is at least 1 dB below the Play Reference Level.
2. The WAVE fader should be at the same setting as determined in the Play Reference Level step.
3. Measure the phase difference between left and right channels. Express the result as ±degrees relative to the left channel.

**6.1.8 Measure Interchannel Crosstalk.**

1. Supply a single tone sine wave or play a WAVE file test signal that has a signal on one channel only. Measure at least a single frequency of 9997 Hz or preferably several frequencies from 5 kHz to one half the sample rate. The level of the signal should be -20 dB FS (as per AES17) or -3 dB FS (as per PC2001). (A

high stimulus level is recommended.)

2. With the signal on one channel only, measure the level of the signal on the second channel using a band pass filter tuned to the stimulation signal frequency. (The band pass filter should be approximately one-third octave.) This may also be accomplished using spectrum analysis.
3. Express the level as a ratio of the crosstalk signal in dB relative to the signal amplitude of the driven channel (normally a negative number).
4. Repeat for the other channel.

**6.1.9 Measure Passband Ripple.**

*(see Annex D for additional clarification)*

1. Supply single tone sine wave or play a WAVE file with test covering the upper frequency range of the PC device under test, typically 1 kHz to 0.2 times the sample rate and a level of -20 dB FS. Choose frequencies with linear spacing and a sufficient quantity to assess ripple, typically at least every 1 kHz from 1 kHz to 0.2 times the sample rate. (PC2001 specifies 40 Hz to 0.2 SR). A multitone signal may also be used for this provided that the peak amplitude of the signal is at least 1 dB below the Play Reference Level.
2. Measure frequency response. The response will typically show a gradual roll off at high frequencies with an abrupt, steep roll off at one half the sample rate. Viewed on a linear frequency scale, there may also be small response variations with an obvious pattern or period.
3. Remove the gradual roll off from the response leaving only the periodic ripple. This determines the mean level.
4. Express the maximum amplitude deviation as ±dB relative to the mean level.

**6.2 Characterize the Record Analog Loop (A-A) Path**

**6.2.1 Establish Input Reference Level:**

1. Supply a 997 Hz, -10 dBV sine wave signal to the Line Input.
2. Establish a signal connection from the Line In to the Line Out by selecting Line In on the Play Mixer. Mute all unused inputs. Master and any Group controls should be set to Maximum. See Annex B-2
3. Adjust the Line Input fader to achieve unity gain. In most cases exact unity gain will not be possible due to the discrete attenuation steps of the slider control. Set to the closest slider setting to the gain that is closest to unity gain, choosing a negative gain if unity is not available. Take note of this setting.
4. Do not change this slider setting for the remainder of the Analog Loop path tests.
5. Increase the sine wave signal input amplitude until the signal at the Line Out is at the maximum amplitude possible with <-40 dB THD+N.
6. The signal level measured at the Line Input is the **Input Reference Level 0 dBr.**
7. Note the fader position and do not change it for any future measurements.

**6.2.2 Measure THD+N**

1. Adjust the sine wave input signal level for -1 dBr (as per AES17) or -3 dBr (as per PC2001) where 0 dBr is the Analog Loop (A-A) Input Reference Level.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz)
3. With the Line In fader at the same setting as determined in the Input Reference Level step, set the sine wave frequency to 997 Hz and measure THD+N as a ratio to the measured total signal amplitude. For THD+N vs Frequency measure THD+N ratio from 20 Hz to one half sample rate with at least 1 point per octave.
4. Express the result in dB or percent (the reading in dB will be negative).

### 6.2.3 Measure IMD

1. Supply an analog two-tone sine wave signal with the amplitudes in a 1:1 ratio with the peak amplitude adjusted to be equal to the Analog Loop (A-A) Input Reference Level. If the signal generator output is calibrated in sine wave peak-equivalent amplitude (normally the case) then the generator amplitude should be set to the Analog Loop (A-A) Input Reference Level. The frequency of the upper tone shall be 20 kHz for 48 kHz sample rate or 0.42 times the sample rate used. The second tone shall be 2 kHz below the upper tone frequency.
2. Set the Line In fader at the same setting as determined in the Input Reference Level step.
3. Measure the rms sum of the lower second and third order difference frequency components with an IMD or spectrum analyzer.
4. Express the IMD distortion result in dB relative to the peak amplitude of the total signal (the reading will be negative).

### 6.2.4 Measure Dynamic Range

*Note: Although the Analog Loop (A-A) path normally does not include A-D or D-A converters the Dynamic range measurement is included for performance comparison.*

1. Adjust the sine wave input signal level for -60 dBr where 0 dBr is the Analog Loop (A-A) Input Reference Level.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz). If significant out-of-band energy exists use an AES-17 filter.
3. Optionally choose a frequency weighting: A-weighting or CCIR-RMS or ITU-R BS.468-4.
4. With the Line In fader at the same setting as determined in the Input Reference Level step, measure absolute THD+N (typically predominately noise) and express this relative to Input Reference Level (dBr) in dB. (This dB value will be a positive or unsigned number). If weighting is engaged, dynamic range should be specified as: XX dB A or XX dB ITU-R BS.468-4 (Quasi-peak detection)..

### 6.2.5 Measure Signal to Noise ratio

*Note: The Analog Loop (A-A) path is the only path where a standard analog Signal to Noise Ratio test (SNR) is appropriate.*

1. Remove the signal input and terminate the input with the source impedance of the generator used, typically 50 ohms.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz). If significant out-of-band energy exists use an AES-17 filter.
3. Optionally choose a frequency weighting: A-weighting or ITU-R BS.468-4.
4. With the Line In fader at the same setting as determined in the Input Reference Level step, measure the absolute amplitude of the noise and express this relative to Input Reference Level (dBr) in dB. If weighting is engaged, dynamic range should be specified as: XX dB A (A weighted) or XX dB ITU-R BS.468-4 (Quasi-peak detection).

### 6.2.6 Measure Frequency Response

1. Set the test generator to test frequencies covering the complete range of the PC device under test, typically 20 Hz to 20 kHz (or 0.4 times the sample rate) and a test generator signal level of -20 dBr. Frequency spacing should approximate one-third octave spacing across the frequency range of interest. A multitone signal may also be used for this test provided that the peak value of the signal is at least 1 dB below the Reference Level.
2. Measure the amplitude at each test frequency at Line Out.
3. Express the amplitude at every frequency as ±dB relative to the level at 997 Hz.

### 6.2.7 Measure Interchannel Phase Response

*(may be measured at the same time as the frequency response)*

1. Set the test generator to test frequencies covering the complete range of the PC device under test, typically 20 Hz to 20kHz sample rate and a signal level of -20 dBr. Frequency spacing should be at least one per octave. A multitone signal may also be used for this provided that the peak amplitude of the signal is at least 1 dB below the Play Reference Level.
2. The Line In fader should be at the same setting as determined in the Play Reference Level step.
3. Measure the phase difference between left and right channels. Express the result as ±degrees relative to the left channel.

### 6.2.8 Measure Interchannel Crosstalk.

1. Set the test generator so there is a signal on one channel only. Terminate the channel without signal in the source impedance of the test generator (back-termination) or 50 ohms. Measure at least a single frequency of 9997 Hz or several frequencies from 5 kHz to 20 kHz. The level of the signal should be -20dBr (as per AES17) or -3 dBr (as per PC2001). (A high stimulus level is recommended.)
2. With a signal on one channel only, measure the level of this signal on the second channel using a band pass filter tuned to the stimulation signal. (The band pass filter should be approximately one-third octave.) This may also be accomplished using spectrum analysis.
3. Express the level as a ratio of the crosstalk signal in dB relative to the signal amplitude of the driven channel (normally a negative number).
4. Repeat for the other channel.

## 6.3 Characterize the Microphone Analog (A-A) Path (optional, only if available on device)

### 6.3.1 Establish Mic Input Reference Level:

1. Supply a 997 Hz, -60 dBV (1 mV) sine wave signal to the Mic Input.
2. Measure the signal at the Line Out connector.
3. Establish a signal connection from the Mic In to the Line Out by selecting Mic In on the Play Mixer. Mute all unused inputs. Master and any Group controls should be set to Maximum. See Annex B-2
4. Increase the Mic Input fader until the signal at the Line Out is <-40 dB THD+N. Take note of the signal level at Line Out.
5. Decrease the Mic Input fader to lower the level at Line Out by one level step. Increase the input level to achieve about –40 dB THD+N. Repeat this until reducing the Mic Input control no longer reduces the distortion to <-40 dB THD+N. Next increase the Mic Input fader by one level step and reduce the input level for a maximum amplitude with < -40 dB THD+N.
6. The signal level measured at the Mic Input is the **Mic Input Reference Level 0 dBr**.
7. Note the fader position and do not change it for any future measurements.

### 6.3.2 Measure THD+N

1. Adjust the sine wave input signal level for -1 dBr (as per AES17) or -3 dBr (as per PC2001) where 0 dBr is the Analog Loop (A-A) Mic Input Reference Level.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz)
3. With the Mic In fader at the same setting as determined in the Mic Input Reference Level step, set the sine wave frequency to 997 Hz and measure THD+N as a ratio to the measured total signal amplitude. For THD+N vs Frequency measure THD+N ratio from 20 Hz to one half sample rate with at least one point per octave.
4. Express the result in dB or percent (the reading in dB will be negative).

#### 6.3.3 Measure IMD

1. Supply an analog two-tone sine wave signal with the amplitudes in a 1:1 ratio with the peak amplitude adjusted to be equal to the Mic Input Reference Level. If the signal generator output is calibrated in sine wave peak-equivalent amplitude (normally the case) then the generator amplitude should be set to the Mic Input Reference Level. The frequency of the upper tone shall be 20 kHz for 48 kHz sample rate or 0.42 times the sample rate used. The second tone shall be 2 kHz below the upper tone frequency.
2. Set the Mic In fader at the same setting as determined in the Mic Input Reference Level step.
3. Measure the rms sum of the lower second and third order difference frequency components with an IMD or spectrum analyzer.
4. Express the IMD distortion result in dB relative to the peak amplitude of the total signal (the reading will be negative).

#### 6.3.4 Measure Dynamic Range

*Although the Microphone Input Analog Loop (A-A) path normally does not include A-D or D-A converters the Dynamic range measurement is included for performance comparison.*

1. Adjust the sine wave input signal level for -60 dBr where 0 dBr is the Analog Loop (A-A) Mic Input Reference Level.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz). If significant out-of-band energy exists use an AES-17 filter.
3. Optionally choose a frequency weighting: A-weighting or ITU-R BS.468-4.
4. With the Mic In fader at the same setting as determined in the Input Reference Level step, measure absolute THD+N (typically predominately noise) and express this relative to the Mic Input Reference Level (dBr) in dB. (This dB value will be a positive or unsigned number). If weighting is engaged, dynamic range should be specified as: XX dB A (A weighted) or XX dB ITU-R BS.468-4 (Quasi-peak detection).

#### 6.3.5 Measure Signal to Noise ratio

*The Analog Loop (A-A) path is the only path where a standard analog Signal to Noise Ratio test (SNR) is appropriate.*

1. Remove the signal input and terminate the input with the source impedance of the generator used, typically 50 ohms.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz). If significant out-of-band energy exists use an AES-17 filter.
3. Optionally choose a frequency weighting: A-weighting or ITU-R BS.468-4.
4. With the Line In fader at the same setting as determined in the Input Reference Level step, measure the absolute amplitude of the noise and express this relative to Input Reference Level (dBr) in dB. If weighting is engaged, dynamic range should be specified as: XX dB A (A weighted) or XX dB ITU-R BS.468-4 (Quasi-peak detection).

#### 6.3.6 Measure Frequency Response

1. Set the test generator to test frequencies covering the complete range of the PC device under test, typically 20 Hz to 20 kHz and a test generator signal level of -20 dBr. Frequency spacing should approximate one-third octave spacing across the frequency range of interest. A multitone signal may also be used for this test provided that the peak value of the signal is at least 1 dB below the Reference Level.
2. Measure the amplitude at each test frequency at Line Out.
3. Express the amplitude at every frequency, except 997 Hz, as ±dB relative to the level at 997 Hz.

#### 6.3.7 Measure Mic Bias Voltage and current (optional – PC2001)

Measure the voltage between the ring and the sleeve of the mic jack with no load.

The voltage should be a maximum of 5.0 V dc . The minimum voltage of 2.0 V dc must be able to sustain a current of 0.8 mA.

### 6.4 Characterize the Record (A-D) Path

*Note: Record tests require measurements of the digital signal at the output of the analog to digital converter. This can be done either by direct interface to the PC digital bus or, if the bus is not accessible, by recording a signal on the hard drive as a WAVE file and then analyzing this WAVE file with suitable software or measurement instrument. A digital output (IEC60958 SPDIF or AES3) may be used if it can be set to unity gain. This means the bit steam will be output without change relative to that of a WAVE file recorded internally.*

#### 6.4.1 Establish Record Reference Level

1. Supply a 997 Hz, -10 dBV sine wave signal to the Line Input.
2. The measurement bandwidth of the digital signal should be set to be to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz). If this is not possible half the sample rate frequency will be assumed to be the upper bandwidth limit.
3. Increase the Record Line Input control until the digital signal level reaches 0 dB FS. If THD+N is below -40 dB use method (a). If 0 dB FS cannot be reached or THD+N >-40 dB, use the second method (b).
   a) Reduce the Record Line Input control one level step (below 0 dB FS) and increase the input sine wave signal level until the signal just reaches or slightly exceeds 0 dB FS with THD+N <-40 dB. Repeat this until reducing the Record Line input control no longer reduces the distortion below –40 dB. Increase the Record Line Input control one level step and reduce the generator sine wave signal amplitude to reach 0 dB FS. THD+N should be <-40 dB.
   b) Reduce the Record Line input control one level step and increase the input level until THD+N is just over –40 dB THD+N. Repeat this until reducing the Record Line input fader level no longer reduces the distortion to <-40 dB THD+N. Next increase the Record Line input fader by one level step and reduce the sine wave input level for maximum amplitude with THD+N < -40 dB. Note this level and reduce the sine wave input signal 0.5 dB.
4. Measure the THD+N of the signal on the PC digital bus (or by analyzing the recorded WAVE file).
5. This signal level measured at Line Input is **Record Reference Level 0 dBr**. This is also the Full Scale Input Voltage (FSIV) when expressed in Vrms.
6. Note the fader position and do not change it for any future measurements.

#### 6.4.2 Measure THD+N

1. Adjust the sine wave input level to produce a level at Line In of -1 dBr (as per AES17) or -3 dBr (as per PC2001) relative to the Record Reference Level.
2. Set the measurement bandwidth to 20 Hz (22.4 Hz) to 20 kHz (22.4 kHz) or half the sample rate.
3. With the Record Line In control at the same setting as determined for the Record Reference Level step, set the sine wave frequency to 997 Hz and measure THD+N as a ratio to the measured total signal amplitude on the PC bus or from the recorded WAVE file. For THD+N vs. Frequency measure THD+N from 20 Hz to 20k Hz (or one half the sample rate) with at least one point per octave.
4. Express the result in dB or percent (the reading in dB will be negative).

#### 6.4.3 Measure IMD

1. Supply an analog two-tone sine wave signal with the amplitudes in a 1:1 ratio with the peak amplitude adjusted to be equal to the Record Reference Level. If the signal generator output is calibrated in sine wave peak-equivalent amplitude (normally the case) then the generator amplitude should be set to the Record Reference Level. The frequency of the upper tone shall be 20 kHz for 48 kHz sample rate or 0.42 times the sample rate used. The second tone shall be 2 kHz below the upper tone frequency.
2. Set the Record Line In control at the same setting as determined for the Record Reference Level step.

3. Measure the rms sum of the lower second and third order difference frequency components with an IMD or spectrum analyzer.
4. Express the IMD distortion result in dB relative to the peak amplitude of the total signal (the reading will be negative).

### 6.4.4 Measure Dynamic Range

1. Adjust the sine wave level to produce a Line In level of -60 dBr relative to the Record Reference Level at a frequency of 997 Hz.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz)
3. Optionally choose a frequency weighting: A-weighting or ITU-R BS.468-4 and select whether quasi-peak or RMS detection.
4. With the Record Line In fader at the same setting as determined for the Record Reference Level step, measure absolute THD+N (typically predominately noise) on the PC bus and express this relative to Input Reference Level (dBr) in dB as Dynamic Range. (This dB value will be a positive or unsigned number). If weighting is engaged, dynamic range should be specified as: XX dB A (A weighted) or XX dB ITU-R BS.468-4 (quasi-peak detection).

### 6.4.5 Measure Frequency Response

1. Set the test generator to test frequencies covering the complete range of the PC device under test, typically 20 Hz to 20 kHz (or one half the sample rate) and a test generator signal level of -20 dBr. Frequency spacing should approximate one-third octave spacing across the frequency range of interest. A multitone signal may also be used for this test provided that the peak value of the signal is at least 1 dB below the Reference Level.
2. The Record Line In fader should be at the same setting as determined in the Record Reference Level step.
3. Measure the amplitude on the PC bus at each test frequency.
4. Express the amplitude at every frequency, except 997 Hz, as ±dB relative to the level at 997 Hz.

### 6.4.6 Measure Interchannel Phase Response

*(may be measured at the same time as the frequency response).*

1. Set the test generator to test frequencies covering the complete range of the PC device under test, typically 20 Hz to 20k Hz (or one half the sample rate) and a signal level of -20 dBr. Frequency spacing should be at least one per octave. A multitone signal may also be used for this test provided that the peak value of the signal is at least 1 dB below the Reference Level.
2. The Record Line In fader should be at the same setting as determined in the Record Reference Level step.
3. Measure the phase difference between left and right channels. Express the result as ±degrees relative to the left channel.

### 6.4.7 Measure Interchannel Crosstalk

1. Set the test generator so there is a signal on one channel only. . Terminate the channel without signal in the source impedance of the test generator (back-termination) or 50 ohms. Measure at least a single frequency of 9997 kHz or several frequencies from 5 kHz to 20 kHz (or one half the sample rate). The level of the signal should be -1 dBr (as per AES17) or -3 dBr (as per PC2001). (A high stimulus level is recommended.)
2. With a signal on one channel only, measure the level of this signal on the PC bus of the second channel using a digital band pass filter tuned to the stimulation signal. (The band pass filter should be approximately one-third octave.)
3. Express the level as a ratio of the crosstalk signal in dB relative to the signal amplitude of the driven channel (normally a negative number).
4. Repeat for the other channel.

### 6.4.8 Measure Passband Ripple

*(see Annex E for additional clarification)*

1. Set the test generator to test frequencies covering the upper frequency range of the PC device under test, typically 1 kHz to 0.2 times the sample rate and a level of -20 dBr. Choose frequencies with linear spacing and a sufficient quantity to illustrate ripple, typically at least every 1 kHz from 1 kHz to 20 kHz. A multitone signal may also be used for this provided that the peak amplitude of the signal is at least 1 dB below the Play Reference Level.
2. Measure Frequency Response on the PC bus. The response will typically show a gradual roll off at high frequencies with an abrupt, steep roll off at one half the sample rate. Viewed on a linear frequency scale, there may also be small response variations with an obvious pattern or period.
3. Remove the gradual roll off from the response leaving only the periodic ripple. This determines the mean level.
4. Express the maximum amplitude deviation at every frequency as ±dB relative to this mean level.

### 6.5 Characterize the Mic Record (A-D) Path

*(optional, only required if a microphone input is present)*

### 6.5.1 Establish Record Reference Level

1. Supply a 997 Hz, -60 dBV (1 mV) sine wave signal to the Mic Input.
2. The measurement bandwidth of the digital signal should be set to be to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz). If this is not possible half the sample rate frequency will be assumed to be the upper bandwidth limit.
3. Increase the Record Mic Input control until the digital signal level reaches 0 dB FS. If THD+N is below –40 dB use method (a) If 0 dB FS cannot be reached or THD+N >-40 dB use the second method (b).
   a. Next reduce the Record Mic Input control one level step (to below 0 dB FS) and increase the input sine wave signal level until the signal just reaches or slightly exceeds 0 dB FS with THD+N <-40 dB. Repeat this until reducing the Record Mic input fader control no longer reduces the distortion below –40 dB. Reduce the Record Mic Input control one level step and adjust the generator sine wave signal amplitude to reach 0 dB FS. THD+N should be <-40 dB.
   b. Next reduce the Record Mic input control one level step and increase the input level until THD+N is just over –40 dB THD+N. Repeat this until reducing the Record Mic input fader level no longer reduces the distortion to <-40 dB THD+N. Next reduce the Record Mic input control by one level step and increase the sine wave input level for maximum amplitude with THD+N < -40 dB. Note this level and reduce the sine wave input signal 0.5 dB.
4. Measure the THD+N of the signal on the PC digital bus (or by analyzing the recorded WAVE file).
5. This signal level measured at the Mic Input is **Mic Record Reference Level 0 dBr**. This is also the Full Scale Input Voltage (FSIV) for the microphone input when expressed in Vrms.
6. Note the fader position and do not change it for any future measurements.

### 6.5.2 Measure THD+N

1. Adjust the sine wave input level to produce a level at Mic In of -1 dBr (as per AES17) or -3 dBr (as per PC2001) relative to the Mic Record Reference Level.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz) or half the sample rate.
3. With the Mic In control at the same setting as determined for the Mic Record Reference Level step, set the sine wave frequency to 997 Hz and measure THD+N as a ratio to the measured total signal amplitude on the PC bus or from the recorded WAVE file. For THD+N vs. Frequency measure THD+N from 20 Hz to 20k Hz (or one half the sample rate) with at least 1 point per octave.
4. Express the result in dB or percent (the reading in dB will be negative).

### 6.5.3 Measure IMD

1. Supply an analog two-tone sine wave signal with the amplitudes in a 1:1 ratio with the peak amplitude adjusted to be equal to the Mic Record Reference Level. If the signal generator output is calibrated in sine wave peak-equivalent amplitude (normally the case) then the generator amplitude should be set to the Mic Record Reference Level. The frequency of the upper tone shall be 20 kHz for 48 kHz sample rate or 0.42 times the sample rate used. The second tone shall be 2 kHz below the upper tone frequency.
2. Set the Mic In fader at the same setting as determined in the Mic Input Reference Level step.
3. Measure the rms sum of the lower second and third order difference frequency components with an IMD or spectrum analyzer.
4. Express the IMD distortion result in dB relative to the peak amplitude of the total signal (the reading will be negative).

### 6.5.4 Measure Dynamic Range

1. Adjust the sine wave input level to produce a Mic In level of -60 dBr relative to the Mic Record Reference Level at a frequency of 997 Hz.
2. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz)
3. Optionally choose a frequency weighting: A-weighting or ITU-R BS.468-4 and select whether quasi-peak or RMS detection.
4. With the Mic In control at the same setting as determined for the Mic Record Reference Level step, measure absolute THD+N (typically predominately noise) on the PC bus and express this relative to Mic Input Reference Level (dBr) in dB as Dynamic Range. (This dB value will be a positive or unsigned number). If weighting is engaged, dynamic range should be specified as: XX dB A (A weighted) or XX dB CCIR-RMS (RMS detection) or XX dB ITU-R 468 (Q peak detection).

### 6.5.5 Measure Frequency Response

1. Set the test generator to test frequencies covering the complete range of the PC device under test, typically 20 Hz to 20 kHz (or one half the sample rate) and a test generator signal level of -20 dBr. Frequency spacing should approximate one-third octave spacing across the frequency range of interest. A multitone signal may also be used for this test provided that the peak value of the signal is at least 1 dB below the Reference Level.
2. The Mic In control should be at the same setting as determined in the Mic Record Reference Level step.
3. Measure the amplitude on the PC bus at each test frequency.
4. Express the amplitude at every frequency, except 997 Hz, as ±dB relative to the level at 997 Hz.

### 6.5.6 Measure Passband Ripple

*(see Annex E for additional clarification)*

1. Set the test generator to test frequencies covering the upper frequency range of the PC device under test, typically 1 kHz to 0.2 times the sample rate and a level of -20 dBr. Choose frequencies with linear spacing and a sufficient quantity to illustrate ripple, typically at least every 1 kHz from 1 kHz to 20 kHz. A multitone signal may also be used for this test provided that the peak value of the signal is at least 1 dB below the Reference Level.
2. Measure Frequency Response on the PC bus. The response will typically show a gradual roll off at high frequencies with an abrupt, steep roll off at one half the sample rate. Viewed on a linear frequency scale, there may also be small response variations with an obvious pattern or period.
3. Remove the gradual roll off from the response leaving only the periodic ripple. This determines the mean level.
4. Express the maximum amplitude deviation at every frequency as ±dB relative to this mean level.

### 6.6 Characterize the Record/Play (A-D-PC-D-A) Path

> The Reference Levels used for the Record/Play path are the **Play Reference Level 0 dBr** (6.1) and **Record Reference Level 0 dBr** (6.4) for a Line level record input or **Mic Record Reference Level 0 dBr** (6.5) when using the Mic level record input instead. Set the Wave control and Line In or Mic In record controls to the same setting used for these measurements. These controls should remain unchanged for this section's measurements.

#### 6.6.1 Measure THD+N

1. Adjust the test generator level to produce a level of -1 dBr (as per AES17) or -3 dBr (as per PC2001) relative to the appropriate 0 dBr reference. Set the sine wave frequency to 997 Hz. For THD+N vs. Frequency use at least 1 point per octave from 20 Hz to 20k Hz (or one half the sample rate).
2. Select the Line In or Mic In on the record mixer with the reference level setting as determined above.
3. Record a WAVE file of enough length to make the measurement (usually about 5 seconds). For multiple frequency measurements record each frequency with enough length for a valid measurement.
4. Select WAVE in the playback mixer with the reference level setting as determined above. Mute all other mixer sources.
5. Set the measurement bandwidth to 20 Hz (22.4 Hz) to 20 kHz (22.4 kHz) or half the sample rate.
6. Play the recorded wave file(s) and measure THD+N at the Line Output as a ratio to the measured total signal amplitude on the PC bus or from the recorded WAVE file.
7. Express the result in dB or percent (the reading in dB will be negative).

#### 6.6.2 Measure IMD

1. Supply an analog two-tone sine wave signal with the amplitudes in a 1:1 ratio with the peak amplitude adjusted to be equal to the appropriate Record Reference Level (either Record Reference Level or Mic Record Reference Level). If the signal generator output is calibrated in sine wave peak-equivalent amplitude (normally the case) then the generator amplitude should be set to the appropriate Record Reference Level. The frequency of the upper tone shall be 20 kHz for 48 kHz sample rate or 0.42 times the sample rate used. The second tone shall be 2 kHz below the upper tone frequency.
2. Select the Line In or Mic In on the record mixer with the reference level setting as determined above.
3. Record a WAVE file of enough length to make the measurement (usually about 5 seconds).
4. Select WAVE in the playback mixer with the reference level setting as determined above. Mute all other mixer sources.
5. Play the recorded wave file(s) and measure the rms sum of the lower second and third order difference frequency components with an IMD or spectrum analyzer.
6. Express the IMD distortion result in dB relative to the peak amplitude of the total signal (the reading will be negative).

#### 6.6.3 Measure Dynamic Range

1. Adjust the test generator level to produce a level of -60 dBr (as per AES17) relative to the appropriate 0 dBr reference. Set the sine wave frequency to 997 Hz.
2. Select the Line In or Mic In on the record mixer with the reference level setting as determined above.
3. Record a WAVE file of enough length to make the measurement (usually about 5 seconds).
4. Select WAVE in the playback mixer with the reference level setting as determined above. Mute all other mixer sources.
5. Set the measurement bandwidth to 20 Hz to 20 kHz (or 22.4 Hz to 22.4 kHz). If significant out-of-band energy exists use an AES-17 filter.
6. Optionally choose a frequency weighting: A-weighting or ITU-R BS.468-4.

7. Play the WAVE file and measure absolute THD+N at the Line Output and express this relative to the Record Reference Level or Mic Record Level in dB as selected. (This dB value will be a positive or unsigned number). If weighting is engaged, dynamic range should be specified as: XX dB A (A weighted) or XX dB ITU-R BS.468-4 (quasi-peak detection).

#### 6.6.4 Measure Frequency Response

1. Set the test generator to test frequencies covering the complete range of the PC device under test, typically 20 Hz to 20 kHz (or one-half the sample rate) and a test generator signal level of -20 dBr. Frequency spacing should approximate one-third octave spacing across the frequency range of interest. A multitone signal may also be used for this test provided that the peak value of the signal is at least 1 dB below the Reference Level.
2. Select the Line In or Mic In on the record mixer with the reference level setting as determined above.
3. Record a WAVE file of enough length to make the measurement (usually about 5 seconds). Record each frequency with enough length to make a valid measurement.
4. Select WAVE in the playback mixer with the reference level setting as determined above. Mute all other mixer sources.
5. Play the WAVE file(s) at the Line Output and measure the amplitude on the PC bus at each test frequency.
6. Express the amplitude at every frequency, except 997 Hz, as ±dB relative to the level at 997 Hz.

#### 6.6.5 Measure Interchannel Phase Response

*(may be measured with the frequency response).*

1. Set the test generator to test frequencies covering the complete range of the PC device under test, typically 20 Hz to 20 kHz (or one-half the sample rate) and a test generator signal level of -20 dBr. Frequency spacing should be at least one per octave across the frequency range of interest. A multitone signal may also be used for this test provided that the peak value of the signal is at least 1 dB below the Reference Level.
2. Select the Line In or Mic In on the record mixer with the reference level setting as determined above.
3. Record a WAVE file of enough length to make the measurement (usually a few seconds). Record each frequency with enough length to make a valid measurement.
4. Select WAVE in the playback mixer with the reference level setting as determined above. Mute all other mixer sources.
5. Play the WAVE file(s) and measure at the Line Output the phase difference between left and right channels. Express the result as ±degrees relative to the left channel.

#### 6.6.6 Measure Interchannel Crosstalk.

1. Set the test generator so there is a signal on one channel only. Terminate the channel without signal in the source impedance of the test generator (back-termination) or 50 ohms. Use at least a single frequency of 9997 Hz or preferably several frequencies from 5 kHz to one-half the sample rate. The level of the signal should be -20 dB FS (as per AES17) or -3 dB FS (as per PC2001). (A high stimulus level is recommended). Adjust the test generator level to produce a level of -60 dBr (as per AES17) relative to the appropriate 0 dBr reference. Set the sine wave frequency to 997 Hz.
2. Select the Line In or Mic In on the record mixer with the reference level setting as determined above.
3. Record a WAVE file of enough length to make the measurement (usually a few seconds). Record each frequency with enough length to make a valid measurement.
4. Select WAVE in the playback mixer with the reference level setting as determined above. Mute all other mixer sources.
5. Playback the WAVE file and the measure level of this signal at Line Out of the second channel using a band pass filter tuned to the stimulation signal frequency. (The band pass filter should be approximately one-third octave.) This may also be accomplished using spectrum analysis.

6. Express the level as a ratio of the crosstalk signal in dB relative to the signal amplitude of the driven channel (normally a negative number).
7. Repeat for the other channel.

### 6.7 Characterize the Digital Loop (D-D) Path

*Note: Some PC audio devices provide a digital output (commonly called a SPDIF Output), a digital input, or both. If only a digital output is provided, tests can be conducted by playing a WAVE file off the hard disk. If only a digital input is provided, tests should be conducted by supplying a digital signal to this input and measuring on the PC Bus or recording the signal as a WAVE file on the hard drive. This WAVE file will have to be subsequently analyzed using PC software or an instrument capably of directly characterizing the WAVE file.*

#### 6.7.1 Digital Output Tests

1. Establish the Digital Input Reference Level: 0 dB FS
2. Play a Full Scale 997 Hz 48 kHz sample rate dithered WAVE file (or supply a Full Scale digital signal to the PC bus).
3. Terminate the digital output with the correct impedance load as per AES 17 (75 ohms unbalanced, 110 ohms balanced)
4. Measure the AES3 carrier peak-to-peak level. Express in Volts Vpp
5. Remove the termination. Measure the new carrier level.
6. Comparing this unterminated level to the previous terminated level, calculate the digital output source impedance. Express in ohms.
7. Adjust the Digital Output fader for a digital level of 0 dB FS
8. Measure the frequency error of the 997.000 Hz signal. Express this error in percent as Sample Rate Error.
9. Observe Status Bits for correct values. Report errors.
10. Play a Full Scale 48 kHz sample rate WAVE file (or supply a Full Scale digital signal to the PC bus) with a “walking ones” and “walking zeros” bit pattern. Look for corresponding errors on the output.
11. Play a Full Scale 997 Hz 48 kHz sample rate WAVE file (or supply a Full Scale digital signal to the PC bus). Measure intrinsic jitter of the digital output.

#### 6.7.2 Digital Input Tests

1. Supply a 0 dB FS digital signal to the digital input
2. Adjust Digital Input fader for 0 dB FS on PC bus. Note fader setting for unity gain.
3. Measure the digital input loading impedance. Using a 0 dB FS digital signal source, with a precise 75 ohm source impedance, compare the signal level open circuit and with the digital input as a load to compute the input impedance.
4. With the 0 dB FS digital test signal, simulate specific amounts of jitter from ?? UI to ?? UI to evaluate the tolerance of the digital input to jitter.
5. With the digital input control set to unity, supply a digital signal with a “walking ones” and “walking zeros” bit pattern. Look for corresponding errors on the output (possibly caused by Sample Rate Converter errors).

## Annex A

(Informative)

### Informative references and bibliography

BALLOU, G. *Handbook for Sound Engineers*. Carmel, IN, US: SAMS, a Division of Macmillan Computer Publishing, 1991, p. 1398-1390 and p. 1435.

CP-307, *Methods of Measurement for CD Players*, Standards of Electronic Industries Association of Japan, 1985.

HARRIS, S. and SANCHEZ, C. *Personal Computer Audio Quality Measurements*. Cirrus Logic Inc., www.cirrus.com/, 1999.

IEC 60268-1 (1985-01) *Sound system equipment. Part 1: General.* Geneva, CH: International Electrotechnical Commission.

ISO 226 (1987), *Acoustics— Normal Equal-Loudness Level Contours*. Geneva, CH: International Organization for Standardization.

ITU-R Recommendation BS.468-4, *Measurement of Audio Frequency Noise in Broadcasting, Sound Recording Systems and on Sound Programme Circuits.* Geneva, CH: International Telecommunication Union, 1986.

METZLER, B. *Audio Measurement Handbook*. Portland , OR, US: Audio Precision, Inc., 1993.

TREMAINE, H. *Audio Cyclopedia*, Indianapolis, IN, US: Howard W. Sams & Co., Inc., 1969, p. 1663-1668.

AES17-1998 (r2004) *AES standard method for digital audio engineering---Measurement of digital audio equipment.* Audio Engineering Society

JONES, WAYNE. *Testing Challenges in Personal Computer Audio Devices.* Audio Engineering Society Convention Paper 5814, presented at the 114th Convention, 2003 March 22-25, Amsterdam, The Netherlands

PC 2001 System Design Guide Co-authored by Intel Corporation and Microsoft Corporation
http://download.microsoft.com/download/win2000pro/PCG/1.0/NT5/EN-US/pc2001v10.exe

IEC 61672-1 First edition 2002 Electroacoustics- Sound level meters – Part 1: specifications

IEC 60958-3 First edition 1999 Digital audio interface – Part 3 Consumer applications

**Annex B**

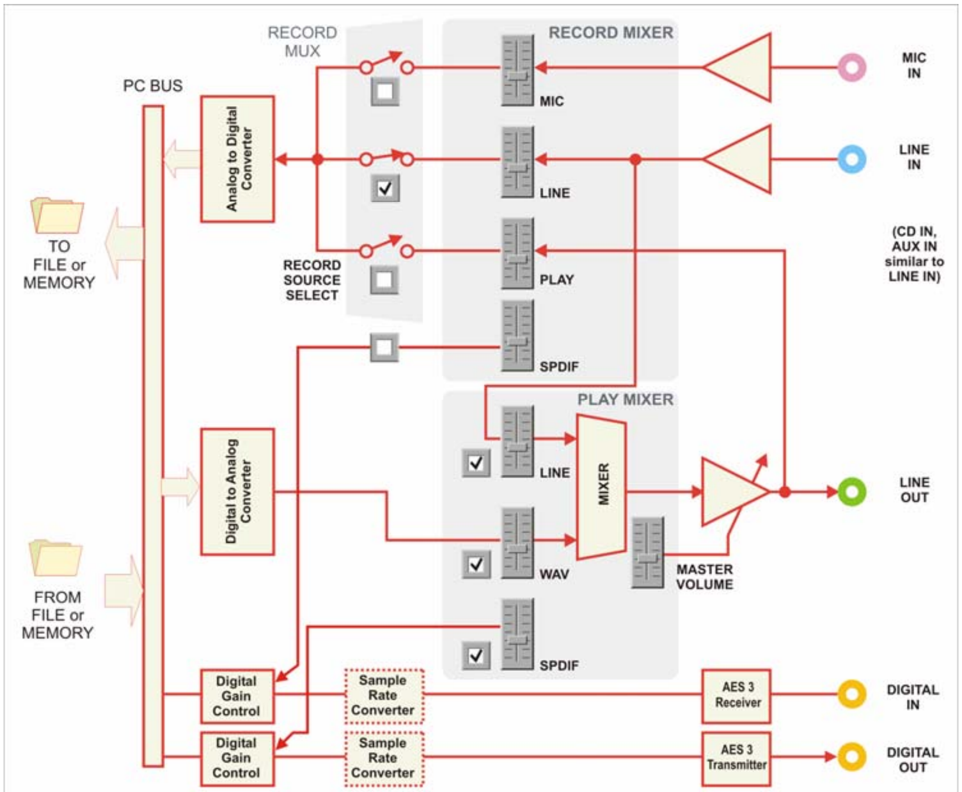
(Informative)

**Measurement Paths**

**B.1** The following diagrams show the signal paths in a typical PC audio device. Not all PC audio devices will have all of the paths shown and some devices may have additional paths. For example, notebook computers may have only a Mic Input and not have a Line Input. Many devices may also have a Auxiliary input or a CDROM input, both of which are similar to the Line Input shown.

Although these simplified diagrams show only a single channel, all analog paths are usually two-channel (stereo) except the microphone input that is typically single channel.

The Record Mixer in most consumer PC audio devices is actually a multiplexer that only allows a single input to be active at one time. The Play Mixer usually functions as a true mixer allowing any combination of inputs to be active, each at its respective level as set by its respective fader control.

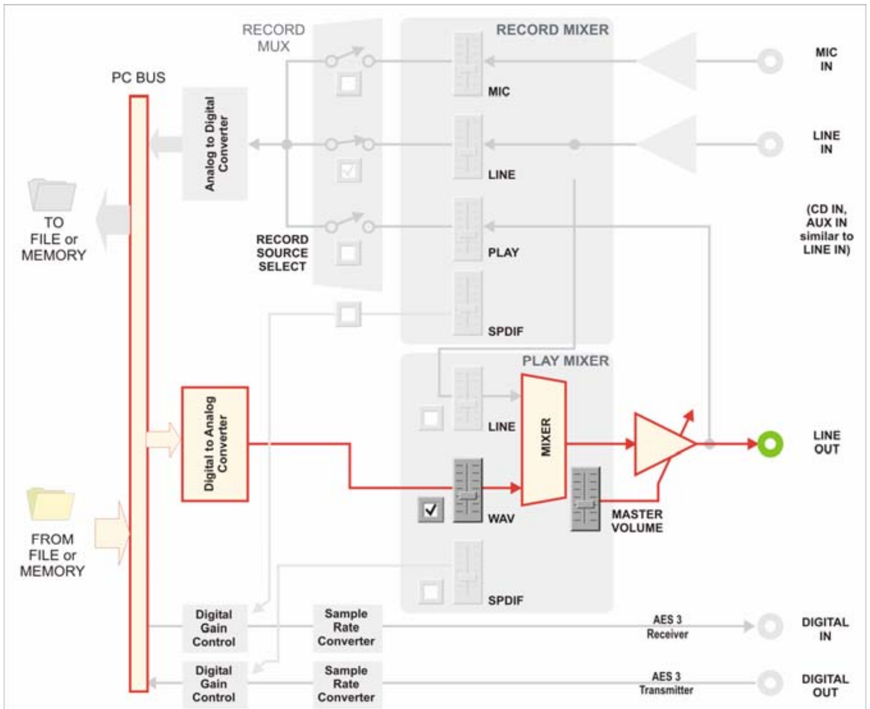



**Example of a complete PC audio device**. A single line-level input is shown labeled Line In. Some devices will have additional line-level inputs such as CD In, Aux In, and others. These will have paths similar to that shown.

In addition to the several paths shown on the succeeding pages, a “Digital to Analog to Digital” path has been used. This path requires an analog cable between Line Out and Line In. The signal path is then PC bus to D to A, through the analog output, externally looped back to the Line In, analog input circuitry, the A to D and back to the bus. While this path may have some attraction for use with a software-based, automated self-test, since there is no way to programmatically set all faders to unity gain, it is not possible to optimize signal levels on the analog side.

**B.2 Play (D-A) Path**

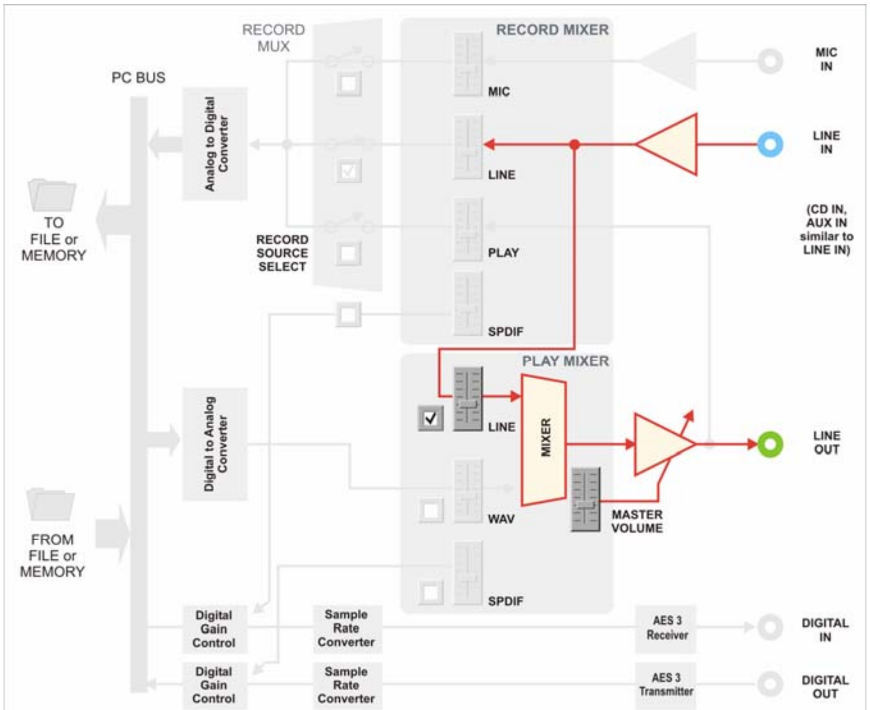
The Play or Digital to Analog Path is the starting point for any set of measurements. This path is exercised by playing a signal through the D to A converter, through the Play mixer to the Line Out. The signal may be a .wav file played from disk or from PC memory or it may be generated directly on the PC audio bus.




The Play mixer will usually have a Wave fader that controls the level in this path. Some systems may also include a Master fader that provides additional control of signal level.

**B.3 Analog Loop (A-A) Path**

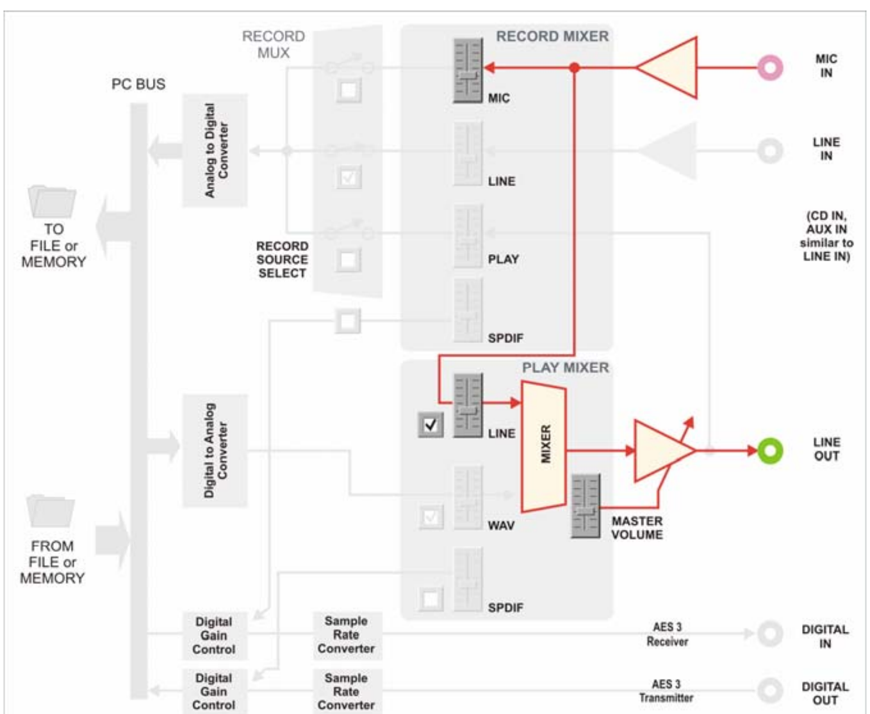
The analog loop provides a signal path through most of the analog circuitry but not through any analog to digital or digital to analog converters.




The Play mixer should have a Line In fader that will affect the signal level in this path. Some systems may also include a Master fader that provides additional control of signal level.

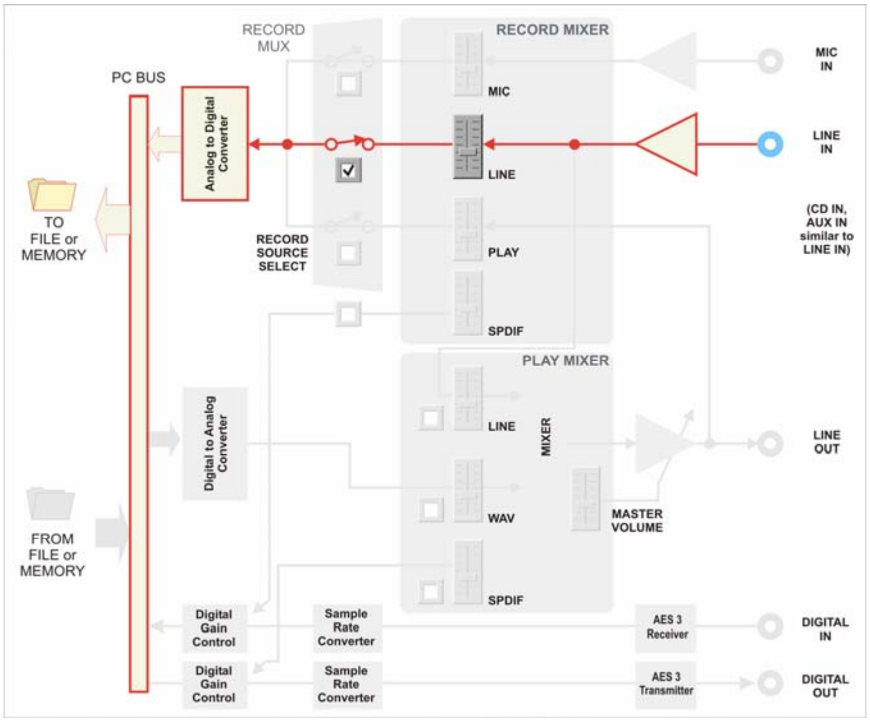
**B.4 Mic Analog Loop (A-A) Path**

The microphone analog loop is identical to the analog loop shown previously except the single-channel (mono) microphone input is used rather than the two-channel line input.



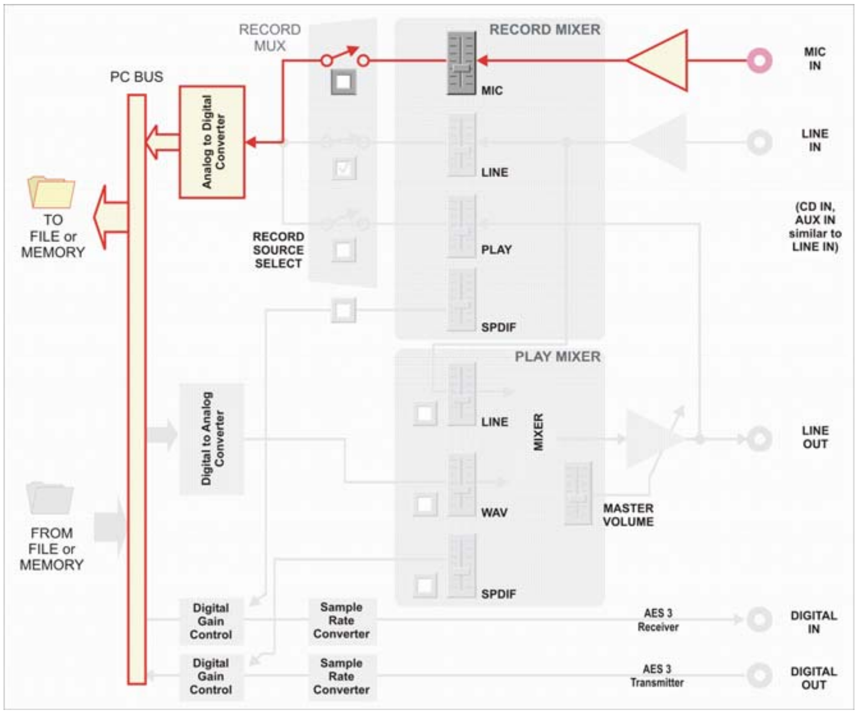
### B.5 Record (A-D) Path

The record path includes the input analog circuitry and the analog to digital converter. The digital output of the converter feeds either computer memory or may be streamed to the hard drive as a file.



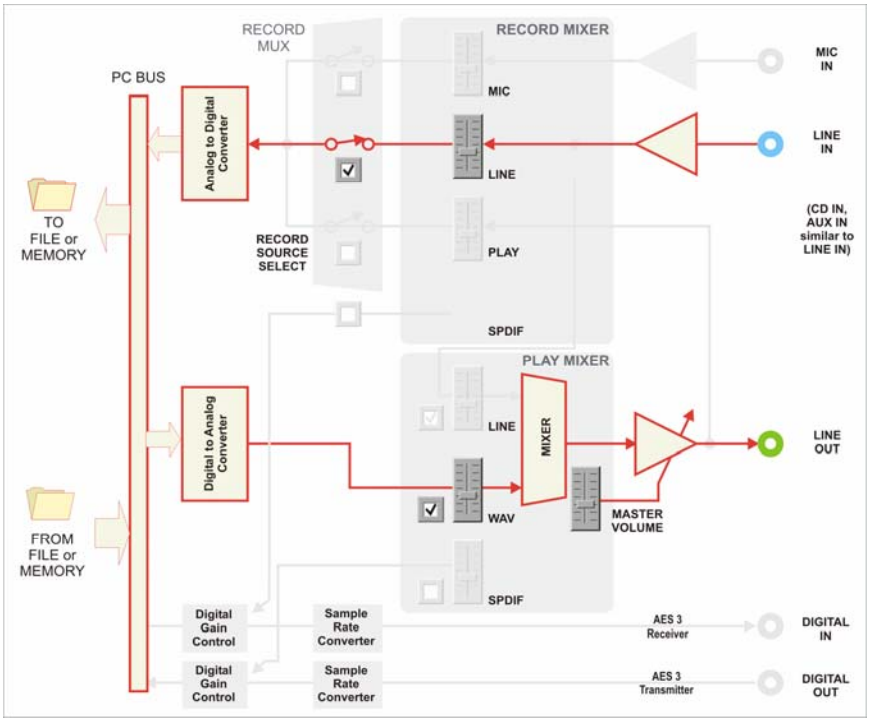

### B.6 Mic Record (A-D) Path

The microphone record path is similar to the line input record path and includes the input analog circuitry and the analog to digital converter. The digital output of the converter feeds either computer memory or may be streamed to the hard drive as a file. The microphone input is usually a single-channel (mono) input. The 3.5 mm ring-tip-sleeve microphone in connector should provide a 5 V dc bias voltage on the ring with respect to the sleeve. The tip is the microphone input, the sleeve is signal ground.



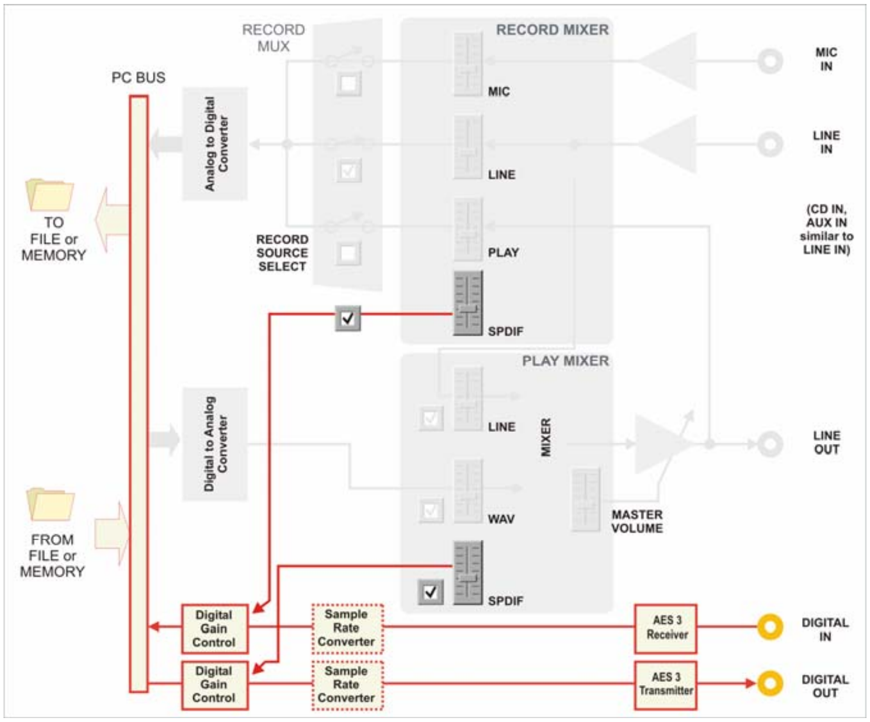
### B.7 Record/Play Loop (AD-PC-DA) Path

This path is a complete loop through the analog input circuitry, the analog to digital converter, the PC digital bus, the digital to analog converter, and finally out through the analog output circuitry. If the device is full duplex, this path may be operated in real time. Simplex devices will require a signal to be recorded and subsequently played back.




### B.8 Digital Loop (D-D) Path

Some PC audio devices may include a digital output, a digital input, or both. Playback and Record mixers may include faders labeled "S/PDIF" or "Digital". These faders will affect the digital path gain in the system. Digital paths also typically include sample rate converters to normalize various input and output sample rates to a single internal sample rate.



# Annex C

(Informative)

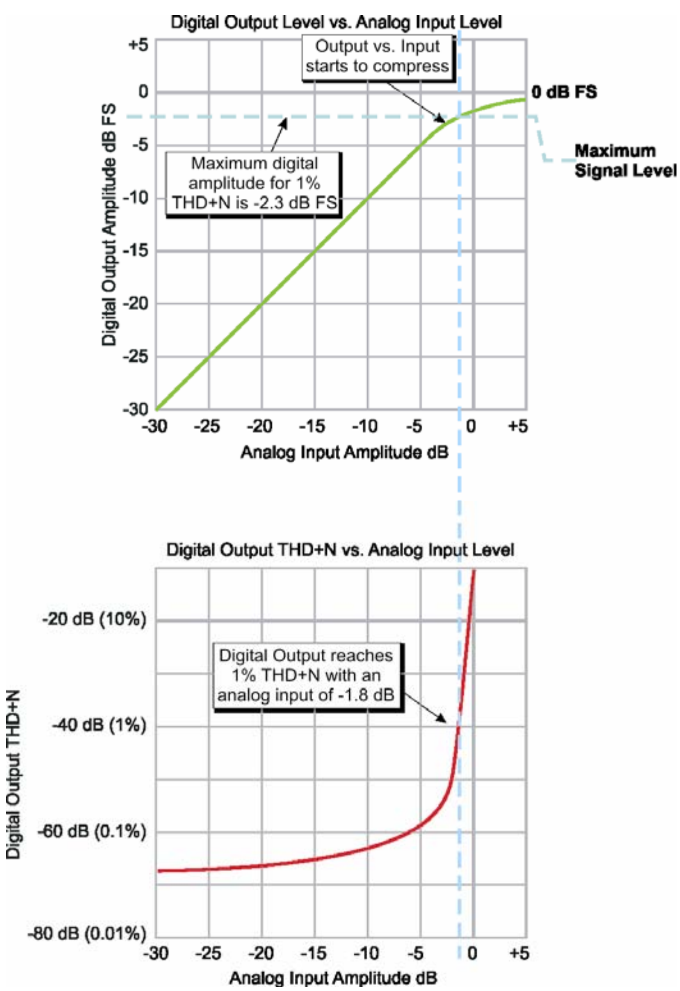
## C.1 Maximum Signal Level and Full-Scale Signal Level

Certain measurements, such as Dynamic Range and Signal to Noise Ratio, are expressed relative to a *Reference Level*. The classic reference level used in digital audio circuits is Digital Full Scale, where 0 dB FS is the signal level that produces the maximum positive digital code. In the A to D (record) path, corresponding to this digital full scale signal level, there should be an equivalent analog signal level to use as a “full scale” analog reference. However, some PC audio devices incorporate a form of compression or limiting in the A to D path that prevents the digital signal from being able to achieve the full scale level. Sometimes, the highest signal level that can be achieved may be 1 to 3 dB below the theoretical full-scale level. Nevertheless, it is important to be able to determine a repeatable reference level that can be used as a reference level for all ratio metric measurements. For optimal performance measurements, this reference level should be as close to “full scale” as possible.

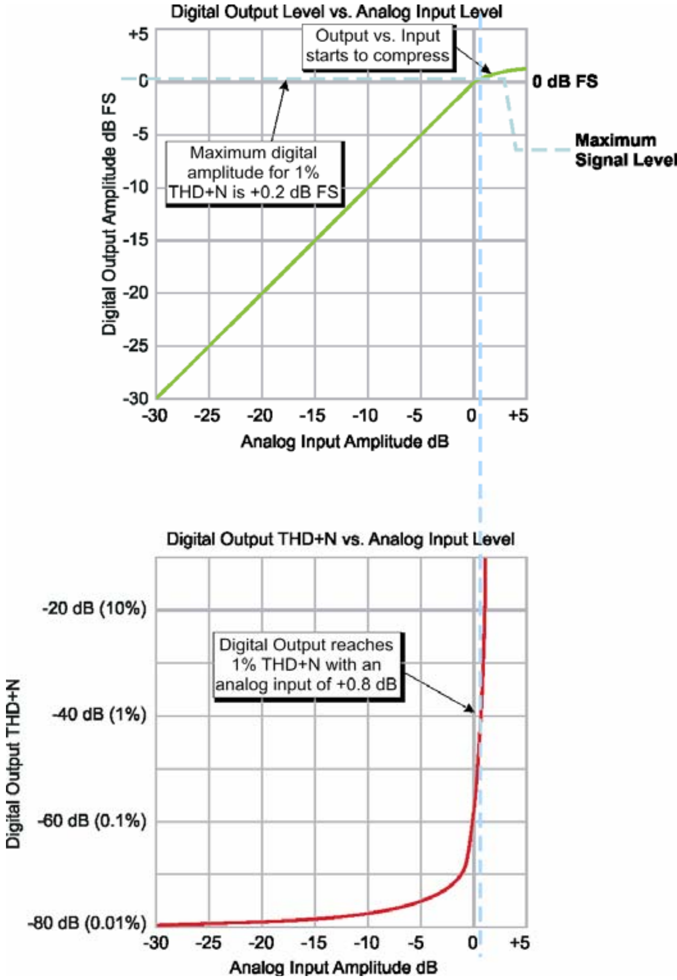
The practice to be followed is to find the highest undistorted signal level that can be achieved in the circuit and establish the Reference Level as 0.5 dB below that point. For this purpose, undistorted shall mean less than -40 dB THD+N (1%).

The graphs below illustrate the behavior of a typical consumer PC audio device and show the compression effect. The upper graph is a plot of digital signal level output from an analog to digital converter against analog input signal level. Notice how the transfer curve is linear until just below full scale, when compression begins. The lower graph is a plot of THD+N of the digital signal at the output of the converter against the same analog input signal level.

In this particular example, the maximum analog input signal that just produced –40 dB THD+N resulted in a digital level of –2.3 dB FS. Thus, the Reference Level should be established at 0.5 dB below that level, or –2.8 dB FS.



**C.2** The graph below is similar to the example on the previous page but illustrates the behavior of a professional PC audio device. In this example, the circuit is able to achieve a much higher signal level, in some cases actually slightly higher than digital full scale as shown on this graph. Using the same process as described previously, the signal level that will produce –40 dB digital THD+N is determined and that analog signal level is reduced by 0.5 dB to be used as the reference level for dynamic range measurements. In this particular example, the digital signal level that just produced -40 dB THD+N was +0.2 dB FS. The equivalent analog signal level was +0.8 dBV. Thus the Reference Level 0 dBr should be 0.5 dB below this or -0.7 dB FS digital or +0.3 dBV analog.



**Annex D**

(informative)

## D.1 Weighting filters

Human hearing is not equally sensitive at all audio frequencies. The ear is most sensitive to mid-band frequencies, that is the 1 to 5 kHz range. This mid-band dominance becomes more pronounced at lower sound pressure levels. Since noise is generally at a lower acoustic level, applying frequency weighting to noise measurements will better approximate the subjective perception of noise. Weighting filters help to provide readings that roughly correspond to the way humans respond to noise.

## D.2 Filters based on A-weighting

."A-Weighting" is a frequency weighting curve that had its origins in acoustic noise measurement. The current standard specifying this weighting filter is IEC 61672-1 First edition 2002. Figure D-1 illustrates the shape of this curve.

## D.3 Filters based on International Telecommunications Union (ITU) standards

Other filters are also used for audio measurements. The sound broadcast industry uses an ITU-R BS.468-4 weighting filter for noise measurements and uses a quasi-peak method of measurement. This filter is illustrated in Figure D1 AES17 recommends an attenuated version of this filter (CCIR-RMS) where the reference level is at 2 kHz instead of 997 Hz, with noise measured using a r.m.s. calibrated indicator.

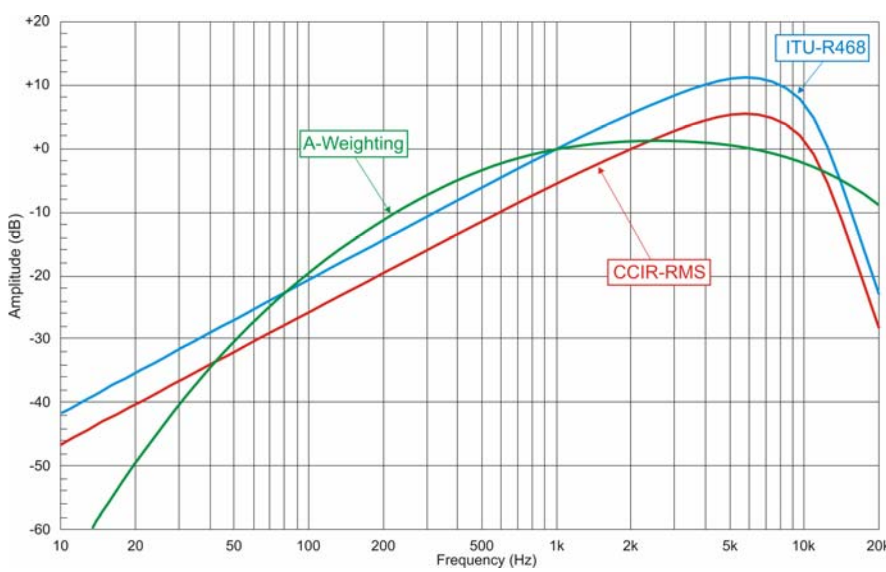



**Figure D-1 —Noise Weighting curves**

## D.4 Reporting

When making an audio measurement, the result should indicate when a weighting filter is used. For example, making a dynamic range measurement for the line-in to line-out analog mixer (A-A) may be listed as

A-weighted analog mixer DR = 90 dB A

If the dynamic range measurement is made using the ITU-R468 filter, the measurement may be listed as

Line level analog mixer DR = 90 dB CCIR-RMS

**Annex E**

(informative)

## E.1 Computation of Passband Ripple

Anti-aliasing and reconstruction filters used with converters can introduce passband ripple. To determine the amount of passband ripple, this response irregularity must be separated from other frequency response deviations. The following graphs and process description indicates how this should be done.

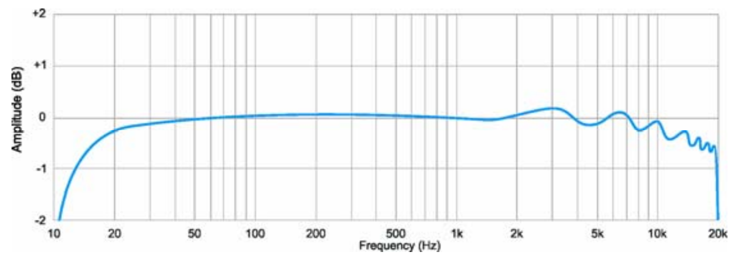



**Figure E-1**

**E.2** The above graph shows a typical frequency response graph of a converter illustrating the passband ripple at the high frequencies. (Note: The passband ripple shown here has been intentionally exaggerated for illustration purposes.) The response graph also shows the typical gradual roll off at high and low frequencies in addition to the ripple. The frequency axis is the familiar logarithmic scale. A logarithmic scale makes it harder to see the detail of the ripple at higher frequencies.

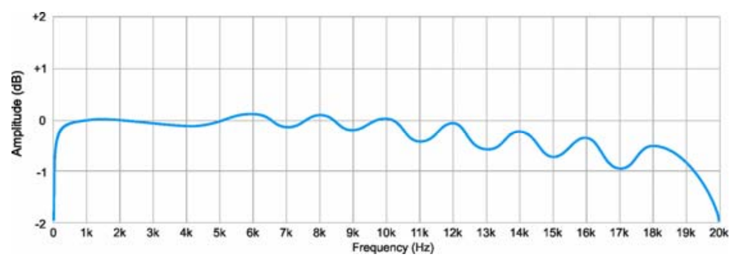



**Figure E-2**

E.3 The second graph above shows the same data with a linear frequency axis, not normally used to show audio frequency response data. However, the linear scale shows the ripple detail much better and shows a clear periodic behavior to the ripple. The high frequency roll off is still part of the data making it difficult to determine absolute ripple magnitude.

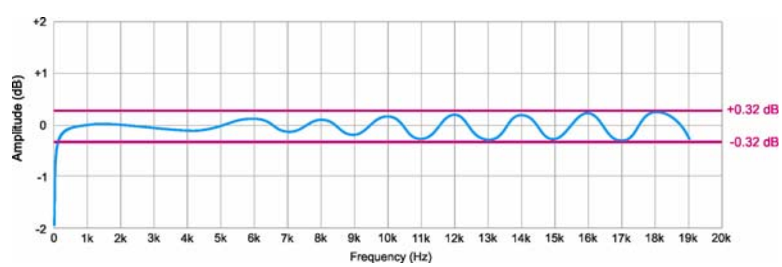



**Figure E-3**

**E.4** The third graph shown above has removed the gradual high frequency roll off now clearly showing the ripple behavior relative to the ideal 0 dB point. The passband ripple it the maximum positive and negative excursion relative to the mean value normalized to 0 dB. This is done by first smoothing the curve to remove the ripple but not the roll off and then subtracting this roll off from the ripple curve. Then the mean value between the maximum and minimum ripple is determined and the curve is normalized to set this mean value to 0 dB.