

# Personal Computer Audio Quality Measurements

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Comments on this document are very welcome.

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## **1 INTRODUCTION**

This document focuses on the measurement of audio quality in a PC environment. Each specification listed has a definition and an example measurement technique. Both electrical and acoustic specifications are discussed. Also included is a detailed description of example test setups to measure each specification. Appendices are used to define the signal paths for measurement, decibel definitions, and weighting filters. Quality levels for some of these measurements are suggested in Microsoft's PC '97, PC '98, PC '99, Intel's AC '97, and in the MPC3 specification.

This document is under review by a number of industry representatives, with a goal of establishing an industry wide consensus on PC audio measurement techniques. If you wish to contribute to this process, please contact Clif Sanchez.

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## 2 ELECTRICAL PERFORMANCE SPECIFICATIONS

The following list of audio electrical performance specifications is relevant to PC audio:

- 1 Full Scale Input Voltage
- 2 Full Scale Output Voltage
- 3 Frequency Response (FR)
- 4 Total Harmonic Distortion plus Noise (THD+N)
- 5 Dynamic Range (DR)
- 6 Continuous Power
- 7 Crosstalk Between Signal Channels
- 8 Noise Level during System Activity
- 9 Sampling Frequency Accuracy
- 10 Input Impedance

For each specification, the following paragraphs describe the specification and discuss measurement techniques. Signal paths are defined in Appendix A (for example A-D-PC). These performance specifications are measured at the end-user accessible signal points; for example, the line in and line out jack sockets. Unless otherwise noted, the measurement bandwidth is 20 Hz to 20 kHz, the test signal frequency is 997 Hz, the system sample rate is 44.1 kHz or greater. The mixer settings are such that all volume controls are set to no attenuation and no gain (0 dB) in the signal path, with only the test channel unmuted.

It is common to filter the noise in the Dynamic Range (DR) measurement to compensate for the uneven frequency sensitivity of the ear, giving low and high frequencies less influence on the final measured value. Either CCIR-486 or A-weighting filters [1] can be used and are described in Appendix C. The definitions given below and the example plots are unweighted. If weighting is used, then the type should be specified, with the results, to allow the comparison of values. The measurement techniques are based on Audio Precision's Audio Measurement Handbook [1], the AES17-1991 measurement standard [2], and the EIAJ CP-307 CD measurement standard [3]. Most measurements are referenced to full scale. Therefore, full-scale definitions for the input and output are described first.

### **2.1 Full Scale Input Voltage (A-D-PC)**

Full-scale input voltage is the input voltage level that will cause the A/D converter output to be just equal to full scale, with no clipping on either positive or negative peaks. Any input offset voltage will cause the full-scale input voltage to be slightly reduced. In addition, analog circuitry could also limit the full scale value by saturating before a digital full-scale code is reached. The mixer volume should be set to 0 dB. Since all PC measurements are 16-bits 2's complement, full scale is  $\pm 32767$  if the THD+N is less than -40 dB. All measurements referring to the analog input are referenced to this full-scale value. This specification is usually measured for the line level input and the microphone level input.

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The test procedure is to apply a sine wave to the input and monitor the output of the A/D converter. This can be done in real time, or by recording the data to a file and analyzing it off-line. Increase the signal level until the output of the A/D converter reaches a full-scale digital code or until the THD+N reaches -40 dB, whichever occurs first. If the THD+N is reached first, reduce the signal level to just below the clipping point (within 0.5 dB of clipping point). Measure the input signal voltage with a root-mean-square (RMS) meter and set that voltage as full-scale (0 dB FS). If the full 16-bit number is not achieved due to clipping, then the actual number used for digital full scale should be clearly specified. If no value is given, the full 16-bit number is assumed ( $\pm 32767$ ).

On a few devices, the analog mixer (A-A) full-scale value might be different than the record path full-scale value. In these cases, full scale for the analog test (A-A) should use the mixer full scale, not the record full scale. All other measurements must use the record full scale (A-D-PC). If the analog mixer and record path full-scale values are different and only one value is used as a measurement reference, the value used must be the lower of the two.

## **2.2 Full Scale Output Voltage (PC-D-A)**

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 speakers. Outputs designed for line-level usually have minimal drive capability and the load used for testing should be set above 40 k $\Omega$ . For outputs designed to drive headphones and speakers, the load used for making performance measurements is very important and should reflect what the end user would experience. Headphone impedance's vary from 16  $\Omega$  to 92  $\Omega$  with 32  $\Omega$  being the most common. Speaker loading should be set at 8  $\Omega$ . For loads other than line-level, the load must be specified with the test results. Outputs designed for more than one function should be tested with each appropriate load. Some analyzers allow the output signal to be swept in amplitude (as shown in Figure 1), which shows the clipping point very well. Figure 1 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.

Output full scale is defined as the voltage produced at the output while playing (PC-D-A) a 997 Hz digital full-scale sine wave, assuming the THD+N is less than -40 dB relative to the signal level. If the THD+N is worse than -40 dB, then full scale is defined as the output signal level 0.5 dB below the level which induces -40 dB THD+N in the output data. If digital full-scale is not the full 16-bit number, then the actual full scale number should be clearly specified.

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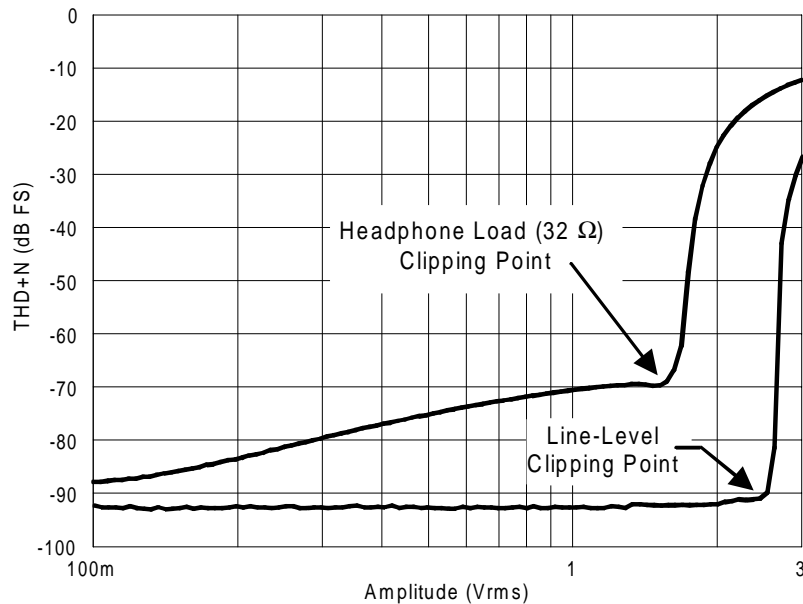


Figure 1. Headphone Clipping Point

## 2.3 Frequency Response (FR)

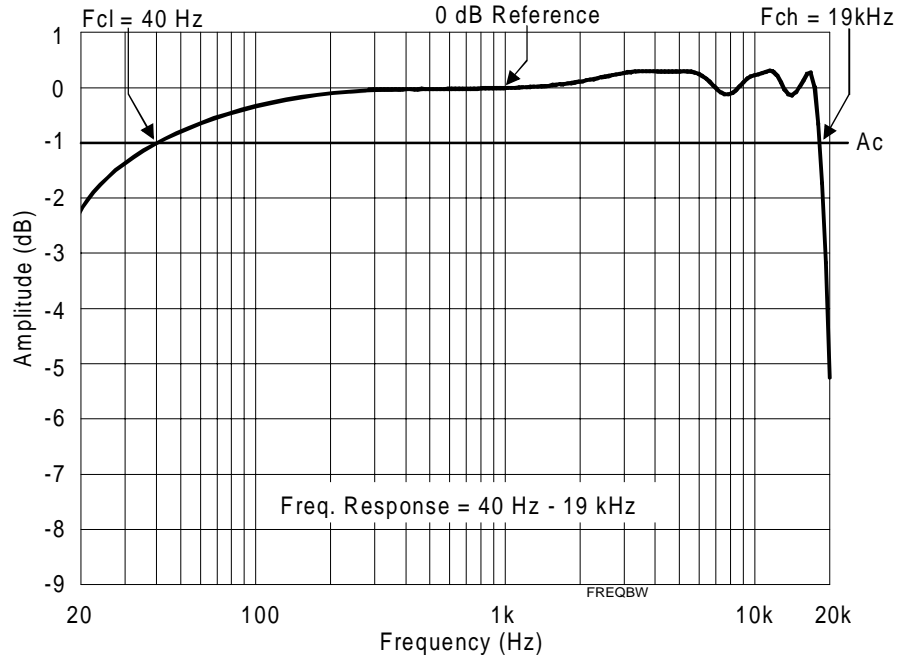
Frequency response is the variation in signal level as the frequency is varied. Amplitude limits or amplitude corners ( $A_c$ ) are required to define frequency response and are usually either  $\pm 1$  dB or  $\pm 3$  dB. FR is tested over the audio bandwidth of 20 Hz to 20 kHz. If spot frequencies are used for testing, choose frequencies at equal or less than octave intervals. Set the absolute magnitude to -20 dB FS. This level is chosen to minimize distortion components which would affect the FR measurement. Measure the level at 997 Hz and set that level as 0 dB (see Figure 2). Make all other frequency measurements relative to that 997 Hz level. Using Figure 2 as an example, the FR should be listed as

$$\text{FR } (A_c = 1 \text{ dB}) = 40 \text{ Hz to } 19 \text{ kHz.}$$

A more accurate representation of FR would be

$$\text{FR} = +0.5, -1 \text{ dB from } 40 \text{ Hz to } 19 \text{ kHz.}$$

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**Figure 2. Frequency Response Plot**

## 2.3.1 FR for Analog Mixer (A-A)

Using an audio analyzer, apply a swept sine wave (20 Hz to 20 kHz), at an amplitude of -20 dB FS, to an analog input. Measure the simultaneous analog output, and plot the deviation in signal level (from 997 Hz) versus frequency.

## 2.3.2 FR for Record (A-D-PC)

Apply a frequency swept sine wave (20 Hz to 20 kHz), at an amplitude of -20 dB FS, to an analog input. Record the data to memory or hard disk. Plot the deviation in signal level (from 997 Hz) versus frequency from the data.

## 2.3.3 FR for Playback (PC-D-A)

Play a frequency swept sine wave (20 Hz to 20 kHz) from a test data file, at an amplitude of -20 dB FS, to an analog output. Measure the signal level and frequency and store. Plot the deviation in signal level (from 997 Hz) versus frequency.

## 2.3.4 Analog FR Record and Playback Path (A-D-PC-D-A)

Set the system to record and simultaneously playback the recorded data. Using an audio analyzer, apply a swept sine wave (20 Hz to 20 kHz), at an amplitude of -20 dB FS, to an analog input. Measure the simultaneous analog output, and plot the deviation in signal level versus frequency. If simultaneous record and playback is not possible, then this test may be achieved by recording the swept sine wave analog signal to disk, and then playing it back into the analog analyzer. This test yields a composite result of record and playback responses added together, which will mask details of the separate playback and record responses, but is useful to judge overall system performance. The data for Figure 2 was taken using this path.

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## **2.3.5 Digital FR Playback and Record Path (PC-D-A-D-PC)**

Set the system to playback and record simultaneously. Play a test file consisting of a swept sine wave (20 Hz to 20 kHz), at an amplitude of -20 dB FS, to an analog output. Connect the analog output to an analog input with a shielded patch cable. Record the analog input to memory or disk. Plot the deviation in signal level versus frequency. This test is useful for an automated self test, requiring only the appropriate software and a patch cable (assuming input and output levels are similar – see Appendix A.) This test requires a system capable of simultaneous record and playback and yields a composite result of playback and record responses added together.

## **2.4 Total Harmonic Distortion plus Noise (THD+N)**

Total Harmonic Distortion (THD) is the ratio of the amplitude of the signal harmonics to the test signal amplitude. More commonly used to indicate THD is a measurement known as Total Harmonic Distortion plus Noise (THD+N), which measures both the harmonics and the noise present in the output signal. 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. Also, the THD+N measurement is easier to perform than THD, since the only requirement is to filter out the test frequency and then perform an audio-band measurement (20 Hz to 20 kHz filter) of the residual, rather than perform a spectral analysis.

In a system with low distortion, the absolute THD+N value tends to be similar to the Dynamic Range (DR) value. In a system which exhibits significant distortion, the absolute THD+N value is worse than the DR value, indicating the presence of distortion. THD+N is often measured with a -3 dB FS test signal level, which measures large signal distortion. The ear frequency sensitivity to large signals is flat; therefore, when using large signals as test stimulus, weighting filters are not employed. Some analyzers can also measure THD+N continuously, while varying the signal level (Figure 3). The results should be specified relative to the full scale value, defined as dB FS. Many analyzers also produce the THD+N answer in dB, which can be converted to percent using the formula in Appendix B.

An example measurement result using the Line In record path could be

$$\text{THD+N (997 Hz, -3 dB FS)} = -85 \text{ dB FS.}$$

This example indicates that the test signal is a 997 Hz sine wave at -3 dB relative to the full-scale Line In. The THD+N result was -85 dB relative to a full-scale digital value.

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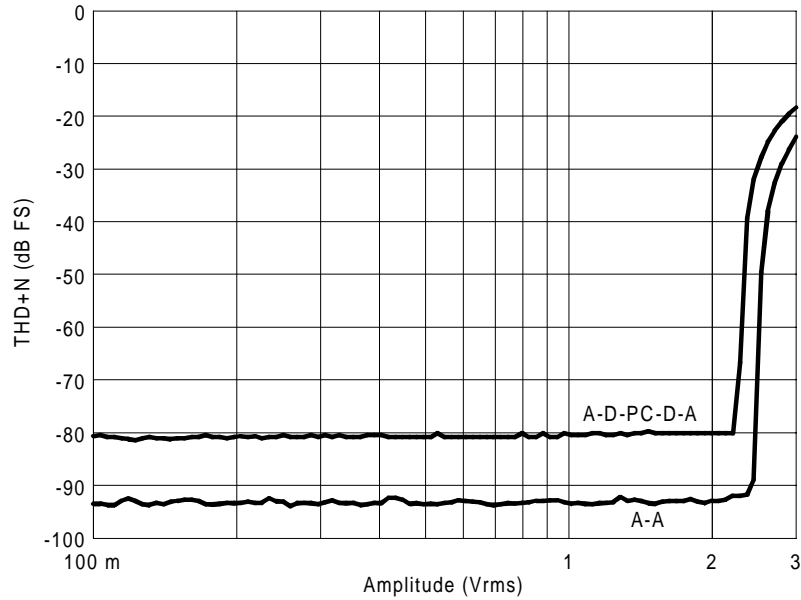
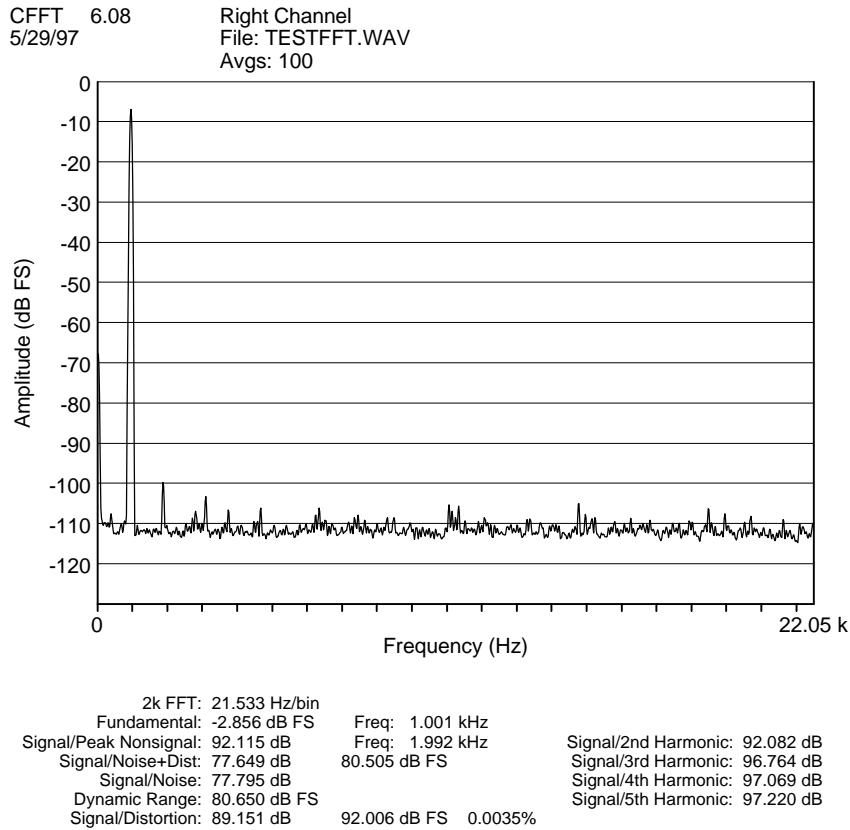


Figure 3. THD+N vs. Amplitude

## 2.4.1 THD+N for Record (A-D-PC)

Apply a sine wave at -3 dB FS to an analog input. Record a block of data to a file or memory. Perform an FFT analysis of the data. Ratio the RMS sum of the noise and distortion components (but not the test signal) to 0 dB FS. Some signal analysis packages can produce a spectral plot of the A/D output data (Figure 4).

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**Figure 4. Record Path Analysis Example**

## 2.4.2 THD+N for Playback (PC-D-A)

Playback a -3 dB FS sine wave to the analog output. Notch out the test frequency component from the output and measure the remaining signal. Ratio this to the previously measured 0 dB (full-scale) level.

## 2.4.3 THD+N for Analog Measurement of Record and Playback Path (A-D-PC-D-A)

Set the system to record and simultaneously playback the recorded data. Establish 0 dB FS level by increasing the signal input until the analog output signal clips. Set the level just below the clipping point. Reduce the input signal to -3 dB FS. Notch out the fundamental from the analog output and measure the residual level. Reference to the previously set 0 dB FS level. This test is useful if the only test equipment available is an analog audio analyzer. Some test equipment can perform a spectral analysis on the residual signal, yielding the frequency characteristics of the residual signal, rather than just an amplitude value (Figure 5). Having a spectral plot is useful for troubleshooting. As an example, the upper plot in Figure 8 clearly indicates that power-line harmonics (60 Hz and multiples) are affecting the measurement results. Without a spectral plot, determining that power-line harmonics were corrupting the measurement would be more difficult.



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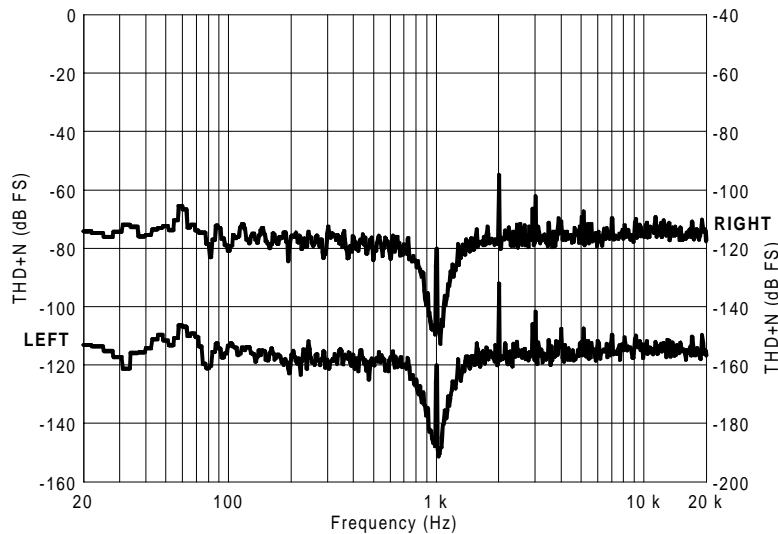


Figure 5. Playback FFT Results

## 2.4.4 THD+N for Digital Measurement of Playback and Record Path (PC-D-A-D-PC)

Set the system to playback and record simultaneously. Connect the analog output to an analog input with a shielded patch cable. Playback a digitally-generated sine wave at 0 dB FS. Adjust the gain of the record path until the A/D converter output is as large as possible without clipping. Set this level to 0 dB FS. Playback a digitally-generated sine wave at -3 dB FS. Perform an FFT on the digital data from the A/D converter and ratio the amplitude of the RMS sum of the non test signal frequencies to 0 dB FS.

## 2.5 Dynamic Range (DR)

Dynamic Range is the ratio of the full scale signal level to the RMS noise floor, in the presence of signal, expressed in dB FS. This specification is given as an absolute number and is sometimes referred to as *Signal-to-Noise Ratio (SNR) in the presence of a signal*. The label SNR should not be used due to industry confusion over the exact definition. DR can be measured using the THD+N measurement with a -60 dB FS signal. This low amplitude is small enough to minimize any large signal non-linearity, but large enough to ensure that the system under test is being exercised. Other test signal amplitudes may be used, provided that the signal level is such that no distortion components are generated. A playback path (PC-D-A) example measurement result could be listed as

$$DR = 85 \text{ dB FS A}$$

This example indicates that the dynamic range for the playback path is 85 dB relative to a Line Out full-scale value (FS). The “A” suffix indicates that an A-weighting filter was used in this measurement. See Appendix C for more information on weighting filters.

### 2.5.1 DR for Record (A-D-PC)

Establish an input 0 dB FS value as mentioned previously. Reduce the input signal to -60 dB FS. Perform an FFT on the digital data from the A/D and ratio the amplitude of the -60 dB signal to the RMS sum of the other frequencies. Reference the result to the 0 dB FS value.

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## 2.5.2 DR for Playback (PC-D-A)

Establish an output 0 dB FS value as previously mentioned. Play back a test digital sine wave file at -60 dB FS, notch out the fundamental and measure the residual level. Reference to the previously measured 0 dB FS.

## 2.5.3 DR for Analog Measurement of Record and Playback Path (A-D-PC-D-A)

Set the system to record and simultaneously playback the recorded data. Establish 0 dB FS level by increasing the signal input until the analog output signal clips. Set the level just below the clipping point. Reduce the input signal to -60 dB FS. Notch out the fundamental from the analog output and measure the residual level. Reference to the previously set 0 dB FS. This test is useful if the only test equipment available is an analog audio analyzer. This test yields a composite result of record and playback DR added together, which will mask details of the separate playback and record values, but does give a good indicator of overall quality.

## 2.5.4 DR for Digital Measurement of Playback and Record Path (PC-D-A-D-PC)

Set the system to playback and record simultaneously. Connect the analog output to an analog input with a shielded patch cable (assuming input and output levels are similar.) Playback the previously obtained 0 dB FS file. Adjust external gain in the record path until the A/D converter output is as large as possible without clipping. Set this level to 0 dB FS. Play back a test digital sine wave file at -60 dB FS (relative to the 0 dB FS point). Perform an FFT on the digital data from the A/D converter and ratio the amplitude of the -60 dB signal to the RMS sum of the other frequencies. Reference the result to the 0 dB FS measured value.

This test is useful for an automated self test, requiring only the appropriate software and a patch cable. The system must be capable of simultaneous record and playback.

## 2.6 Continuous Power

This specification is required for loads other than line level outputs (speaker and headphone) and gives the power under a specified load using a 997 Hz sine wave played at the listed full-scale voltage. This is also the maximum power attainable since it is full scale. The load is typically 8  $\Omega$  for speakers and 30  $\Omega$  for headphones and MUST be specified with the measurement data.

As an example, if the full-scale output under headphone loading conditions (30  $\Omega$  per channel) is 1.2 V<sub>RMS</sub>, the continuous power would be:

$$\text{Continuous Power} = (1.2)^2/30 = 48 \text{ mW}$$

This result should be listed as

$$\text{Headphone Continuous Power (30 } \Omega \text{)} = 48 \text{ mW/channel}$$

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## **2.7 Crosstalk Between Signal Channels**

Crosstalk is the leakage of information from one channel to another. This could occur in multiple scenarios such as: left to right or right to left on any stereo input pair or stereo output pair, input to output crosstalk, and output to input crosstalk. The measurement technique is the same for each of these measurements, with the appropriate signal path connections made.

A -20 dB FS test sine wave is applied to one input. The inputs to all other channels are either terminated to signal ground via a  $50\ \Omega$  resistor (for analog inputs), or driven with digital zero (for digital inputs). The output signal from each undriven channel is analyzed for the presence of the test signal frequency. The amplitude of the test signal frequency is measured, and expressed as a dB ratio to the test signal amplitude. This may be done by using a third octave filter to measure the undriven channels, or by FFT analysis. The test frequency should be varied in steps of less than an octave, and the measurements repeated. The result can then be plotted as a graph of crosstalk amplitude vs. frequency. If a single spot frequency is used, it should be 10 kHz.

## **2.8 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 of performing this 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, this can be verified by listening tests.

## **2.9 Frequency Accuracy**

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

### **2.9.1 Sample Frequency Accuracy on Playback (PC-D-A)**

Obtain a test data file containing a 997 Hz ( $\pm 25$  ppm) sine wave, recorded with a sample rate of exactly 44.1 kHz. ( $\pm 25$  ppm). Playback this data file using the PC under test. Measure the frequency of the 997 Hz sine wave with a frequency meter with an accuracy of better than  $\pm 10$  ppm. Express the deviation from 997 Hz as a percentage.

### **2.9.2 Sample Frequency Accuracy on Record (A-D-PC)**

Record a 10 second period of exactly 997 Hz ( $\pm 25$  ppm) sine wave, with a sample rate of 44.1 kHz. Playback this file using a system whose sample rate is known to be 44.1 kHz exactly ( $\pm 25$  ppm). Measure the frequency of the 997 Hz sine wave with a frequency meter with an accuracy of better than  $\pm 10$  ppm. Express the deviation from 997 Hz as a percentage.

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## 2.10 Input Impedance

Input impedance is important for interoperability among different audio system components. Interoperability minimizes aggravation from the end-users perspective and thereby promotes a good audio experience for the end user. Input impedance is germane to line-level inputs that may need to connect up to external audio equipment or internal PC connections such as the analog output from a CD-ROM drive. Input impedance is also critical to microphone inputs as impedance has a direct correlation to electret microphone sensitivity.

### 2.10.1 Line-Level Input Impedance

Line-level inputs should be designed with an impedance higher than  $10\text{ k}\Omega$ , and preferably around  $47\text{ k}\Omega$  (which is the typical load level for consumer audio equipment.) The impedance should be tested at 997 Hz.

### 2.10.2 Microphone Input Impedance

Microphone inputs vary considerably from sound card to sound card. Figure 6 illustrates a generic electret microphone and input circuit. The mic jack circuit illustrated is considered the most universal jack. This jack supports 2-pin dynamic mics as well as 3-pin electrets (as shown in the diagram.) There are two impedance's associated with the mic jack. The standard input impedance illustrated as  $Z_i$  and the bias impedance  $R_b$ . The input impedance  $Z_i$  should be tested similar to the line-level approach using a 997 Hz test signal. This input impedance (in parallel with the bias impedance  $R_b$ ) has a direct affect on the voltage sensitivity of electret mics and should be as high as possible, typically greater than  $20\text{ k}\Omega$  (an order of magnitude greater than  $R_b$ ) and no lower than  $10\text{ k}\Omega$ .

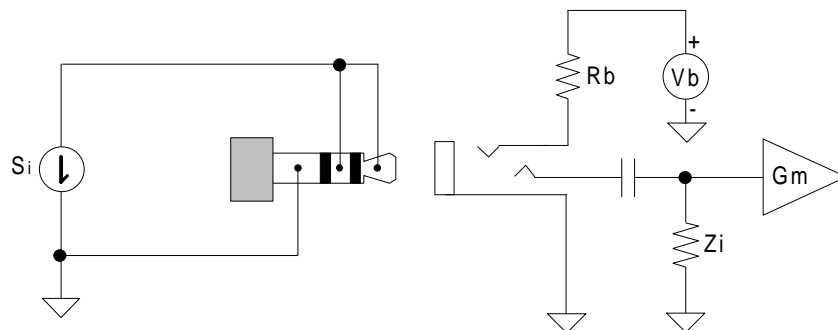


Figure 6. Generic Microphone Circuit

The bias impedance  $R_b$  has two components, an AC impedance and a DC impedance. If  $R_b$  is a single resistor connected to a bias voltage  $V_b$ , then the AC and DC impedance are identical. The DC impedance, along with the bias voltage will determine the operating voltage for the electret mic. The AC impedance has an affect on the mic sensitivity and should be no lower than  $2\text{ k}\Omega$  when tested at 997 Hz.

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## 3 ACOUSTIC PERFORMANCE SPECIFICATIONS

The audio experience of the end-user is heavily influenced by the quality of the microphone and loudspeakers used by the PC manufacturer. Where such transducers are built-in or included with the system, there should be guidelines for their quality level.

The following is a list of acoustic performance specifications which are relevant to PC audio:

1. Frequency Response
2. Total Harmonic Distortion
3. Power Output of Loudspeaker
4. Microphone Sensitivity
5. Mechanical (Fan and Disk) Noise

For each one of these specifications, the following paragraphs describe the specification and discuss measurement techniques. Many of the measurements require the use of a calibrated, known frequency response and distortion, microphone, and a sound pressure level (SPL) meter. Bruel and Kjaer manufacture a range of reference microphones, and a reasonable SPL meter is available from Radio Shack. An anechoic chamber is not necessary for many of the measurements, although a quiet, relatively reflection free environment is useful.

One common test signal used to help overcome room reflections and standing waves is called Maximum Length Sequence (MLS). This is a short signal of frequency shaped noise. The output of the device under test is captured and an FFT analysis performed. The data is captured during the time period before any room reflections occur, thus removing room reflection effects. Alternatives to MLS include warble tones (where the test signal frequency is periodically varied about a mean value), which are particularly good for level setting.

### **3.1 Frequency Response**

#### **3.1.1 Frequency Response of Loudspeakers**

Place the reference microphone 0.5 m from the center axis of the speaker under test. Playback a pink noise test signal and use an SPL meter to set the level at 90 dB SPL at the test microphone. Playback a MLS test signal and capture the microphone output with an audio analyzer. Plot the frequency response result.

#### **3.1.2 Frequency Response of the PC Microphone**

Tape the reference microphone as close as possible to the microphone under test. Place a reasonably good quality loudspeaker 0.5 m on axis away from the microphone. Feed a pink noise test signal to the loudspeaker and use a SPL meter to set the level at 90 dB SPL at the test microphone. Adjust the gain of the microphone under test to yield a -20 dB FS digital level. Feed a MLS test signal to the loudspeaker and capture both microphone outputs with an audio analyzer. The reference microphone can be connected directly to the analyzer. The data from the microphone under test is normally captured on the PC and then transferred in digital form to the analyzer. Calculate the frequency response via the reference path and via the microphone under test. Adjust the reference microphone response for any deviations in the microphone response

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curve. Subtract the adjusted reference response from the microphone under test response and plot the result.

## **3.2 Total Harmonic Distortion**

### **3.2.1 THD of Loudspeakers**

Place the reference microphone 0.5 m from the center axis of the speaker under test. Playback a pink noise test signal and use a SPL meter to set the level at 90 dB SPL at the test microphone. Playback either a series of sine waves at test frequencies from 50 Hz to 10 kHz spaced less than an octave apart, or use a multi-tone test signal. Capture the reference microphone response, filter out the test signal and express the residual as a percentage of test signal amplitude. This method assumes that the noise floor of the system under test is significantly lower than the distortion harmonics. If this is not true, then the amplitude of each of the first five harmonics should be individually measured, summed in a RMS fashion and then ratioed to the test signal amplitude.

### **3.2.2 THD of Microphones**

It is possible to measure the THD of a microphone by comparing it to a known reference microphone. Since most PC microphones will be used for voice applications, and even cheap electret microphones have relatively low distortion, there is no requirement to measure the THD of PC microphones.

## **3.3 Power Output of Loudspeakers**

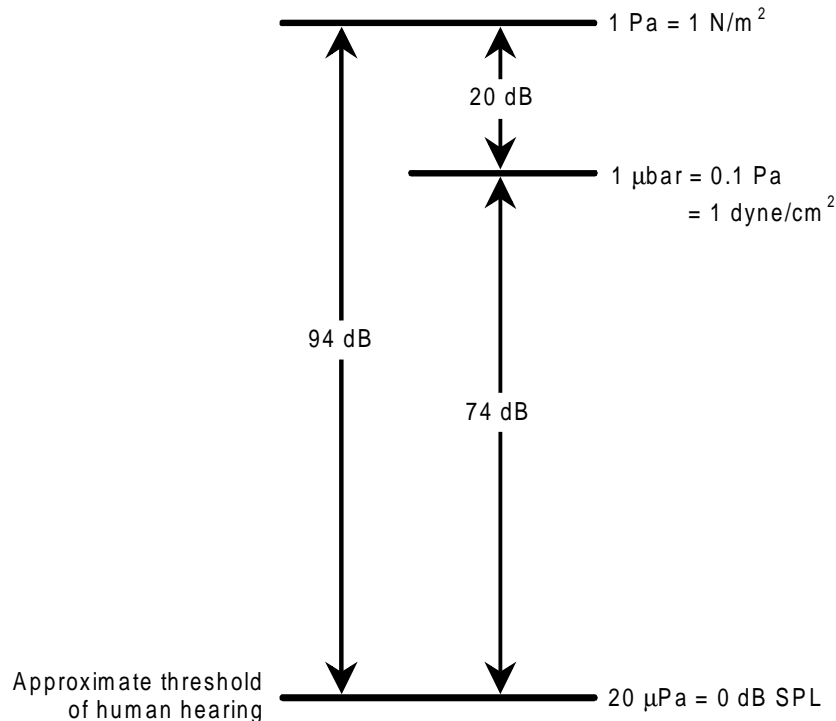
Loudspeakers are often rated at their electrical power maximum limits, which does not indicate the maximum acoustic output power available. A common definition of maximum acoustic output is when the distortion value has reached 10 % (20 dB). Thus the test method is similar to measuring loudspeaker distortion, except that the signal level is adjusted until the distortion is 10 %, and then the acoustic power level is measured with an SPL meter. This may be done at 997 Hz, or over a number of test frequencies spanning the audio range.

## **3.4 Microphone Sensitivity**

The sensitivity of the microphone has a great effect on usability. Mic sensitivity is generally specified as amount of voltage produced for a given air pressure (in dB), Sv. For dynamic mics (moving coil), Sv works well; however, electret mic actually produce a current that is proportional to a given air pressure, Si. When electret mics specify voltage sensitivity, Sv, a load resistor is assumed. This load resistor is not part of the mic, but part of the sound card and is equal to the equivalent input impedance of the sound card's mic input jack ( $R_b$  in parallel with  $Z_i$  in Figure 6). If voltage sensitivity (Sv) is specified for electret mics, then the bias resistor  $R_b$  should be 2 k $\Omega$  and the voltage at the mic should be 2.0 V. The actual sensitivity of the electret mic is dependant on the sound cards deviation from the load resistor used by the mic manufacturer to generate the Sv number.

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$S_v$  is specified in dB V per some reference. The preferred method is dB V/Pa, which is dB V relative to 1 Pascal. The test frequency to derive sensitivity should be 997 Hz. A typical value for electrets would be  $S_v = -45$  dB V/Pa and  $S_v = -56$  dB V/Pa for dynamic mics. Using the electret mic as an example, one Pascal of air pressure would produce  $-45$  dB V or  $5.62$  mV<sub>RMS</sub>. References other than one Pascal are also used although deviating from a common reference creates confusion among end-users. Another typical reference point is 1  $\mu$ bar. One  $\mu$ bar is 20 dB lower than one Pascal. The equivalent sensitivity in the above electret example would be  $-65$  dB V/ $\mu$ bar. The sensitivity is 20 dB lower because the reference pressure, 1  $\mu$ bar, is 20 dB lower than 1 Pascal. Figure 7 illustrates different acoustical reference points.



**Figure 7. Acoustic References**

A suggested test method is to place a loudspeaker on axis 0.5 m from the microphone. Drive the loudspeaker with a warbled 997 Hz test signal. Set the loudspeaker output to 94 dB SPL at 1 m. Set the PC to record, using an application which allows gain adjustment with a clipping indicator. Confirm that the software driven gain adjustment is capable of causing the recorded data to clip, and also capable of near to full scale operation with normal slider settings.

### **3.5 Mechanical Fan and Disk Noise**

Measure the fan noise using a sound pressure level meter at 0.5 m pointing at the fan outlet on axis. Measure the disk noise using a sound pressure level meter at 0.5 m pointing at the front of the PC system unit containing the hard disk while running a program which exercises the disk in a noisy manner. A similar measurement should be made with the CD-ROM drive being fully exercised.

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## 4 TEST EQUIPMENT

Test equipment must be substantially better in performance than the system under test. Here are some examples of known, high quality, audio test equipment.

### ***4.1 Audio Precision System One or Two***

This unit is an audio analyzer which uses a PC as an input and display device. It provides analog signal generation and measurement, and digital signal generation and measurement. There are many built-in routines for measuring A/D systems, D/A systems, D-D systems and A-A systems. All test setups and results may be saved on disk. Very flexible plotting routines are included. The unit is programmable to allow test automation. Better than 100 dB dynamic range specifications allows measurement of professional audio equipment. Cost is about \$28,000.

### ***4.2 Audio Precision Portable One Plus***

This unit is an audio analyzer which allows A-A and A-D-PC-D-A testing only. It is self-contained with built-in controls and display. A sweepable generator and notch filter for THD+N, DR, and FR measurements are provided. Plotting is only available using a dot-matrix printer. Cost is about \$7,000.

### ***4.3 Automatic Self-test***

Some categories of test lend themselves to being automated. Manufacturers of sound systems and silicon could provide test suites which allow testing of standard audio configurations by running the tests on the PC under test. Such test software could be made available on company web sites for general distribution, and could allow WHQL audio quality pre-screening by OEMs.

## 5 DETAILED TEST SETUP

This section defines the test procedure for testing an audio device in a multimedia PC system. This setup is an attempt to provide a consistent test methodology across different audio systems with different capabilities. Setting up the audio part is critical to getting consistent, fair and repeatable readings.

### ***5.1 Validate the Test System***

The audio system under test, defined as equipment under test (EUT), will only measure as good as the test system; therefore, the limitations of the test hardware must 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 will 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 will not be accurate. For example, if the test system is 10 dB better than the EUT, then the EUT will measure 0.5 dB worse than it



# Personal Computer Audio Quality Measurements

actually is. If the test system has the same performance as the EUT, then the EUT measurements will measure 3 dB worse than it actually is.

Once the performance of the test system is known, the ground of the test system must be connected to the ground of the EUT. Ground loops cause many problems in audio testing. If the performance gets worse when the grounds are connected, grounding problems exist and must be resolved before true EUT testing can begin. Figure 8 illustrates the results of a 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 3.5 dB. 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/power strip as the system under test. The “bad grounding” in the Figure below was generated by using two isolated wall sockets: one for the test equipment and one for the EUT.

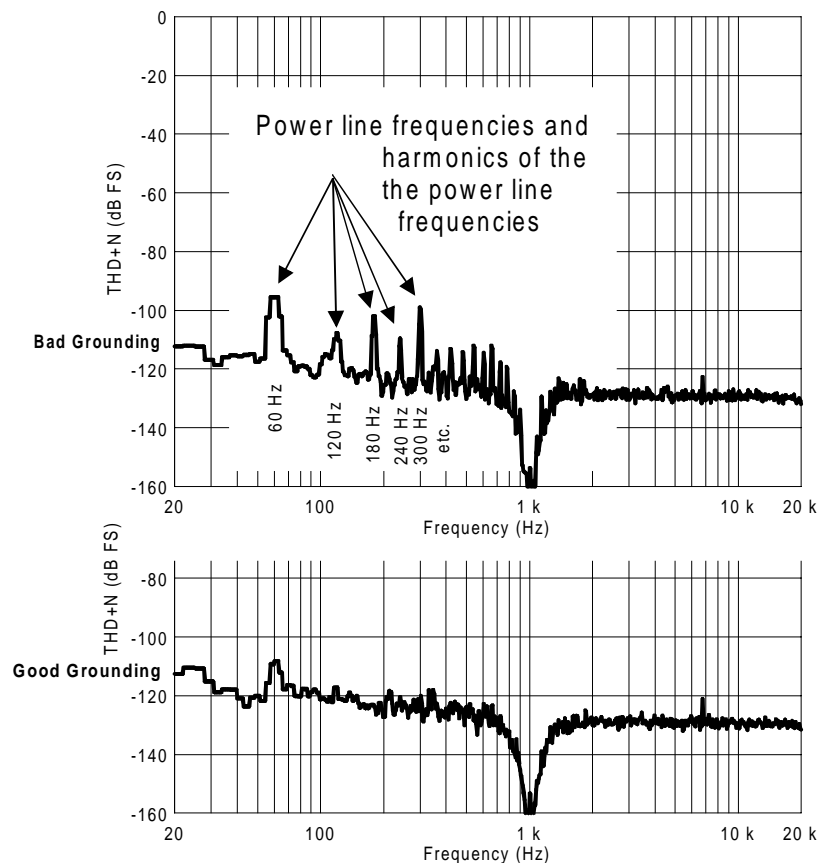


Figure 8. Test System Grounding Example

## 5.2 System Under Test Setup

The EUT must be setup properly for the tests described below. Systems from different manufacturers have varying capabilities. There needs to be a fair way to test the audio system without restricting the feature set supported.

# Personal Computer Audio Quality Measurements

## 5.2.1 Mute all channels except the one under test

The basic setup is to mute all audio sources except the path under test. This provides fair testing regardless of the number of audio paths supported. Audio systems from different manufacturers vary considerably in architecture. Some systems have a multiplexer going to the ADCs while others support a mixer arrangement. The only common point is one channel enabled (always the case in the multiplexer arrangement) with the other channels muted (the mixer arrangement).

## 5.2.2 Set volume for channel under test to 0 dB.

The second step is to set the volume in the part to 0 dB - no attenuation and no gain. This is the only fair reference point across different manufacturers. Gain tends to make performance worse and attenuation tends to make performance look better. Manufacturers support different volume ranges for a given input.

Using the Line-In analog mixer for example, the Creative Labs Sound Blaster Pro supports a volume range of 0 to -46 dB, whereas the WSS Line-In analog mixer supports a volume range of +12 dB to -34 dB. Even on a single device, different audio paths have different gain ranges. For example, the WSS analog mixer supports +12 to -34 dB; whereas, the DAC volume control is defined as 0 to -94.5 dB. The only common volume level across all parts is 0 dB - no gain and no attenuation. Using this value also allows manufacturers to choose their own volume ranges for differentiation or personal preference.

## 5.3 TESTS

The following tests were performed using external equipment and software on the EUT. The following assumes that the unmuted path volume is set to 0 dB and that all paths not being tested are muted. Since all audio performance measurements are relative to full scale, full scale must be the first measurement made. This provides fair comparison points without restricting the absolute value of full scale. The units used for the distortion and noise measurements would then be “dB FS” indicating “dB relative to full scale.”

Full scale (FS) is defined differently for the three different measurement points: inputs, outputs, and digital measurement points. Digital full-scale maximum is constant once the number of bits is specified. In the PC all measurements should be 16-bits; therefore, maximum full-scale is  $\pm 32767$ . If analog circuitry clips before the digital full-scale code is reached, then digital full scale is lower than  $\pm 32767$  and must be specified.

# Personal Computer Audio Quality Measurements

## 5.3.1 Finding Full Scale

The test sequence for inputs is:

- Unmute the input path to the ADC (and set volume of path to 0 dB)
- Record the data from the ADC in 16-bit 2's complement format
- Set the generator to a 997 Hz sine wave
- Increase the amplitude into the ADC until the recorded numbers reach near  $\pm 32767$  (assuming the THD+N is less than  $-40$  dB)
- Set the generator voltage as the 0 dB FS point for that input
- If the THD+N is worse than  $-40$  dB (relative to the magnitude of the incoming signal), lower the amplitude until the THD+N is  $-40$  dB and set the resultant input voltage to 0 dB FS. Also note in the measurements that digital full scale input (record full scale) is lower than  $\pm 32767$ .

The test sequence for outputs is:

- Unmute the path from the DACs to the output under test (volume set to 0 dB)
- Play a digital full-scale 997 Hz sine wave
- Load the output appropriately
- Set the analyzer voltage as 0 dB FS for that output (assuming the THD+N is less than  $-40$  dB)
- If THD+N worse than  $-40$  dB (relative to the magnitude of the output signal), lower the amplitude of the digital sine wave until THD+N is  $-40$  dB and set the resultant analyzer voltage to 0 dB FS for that output. In addition, note in the measurements that digital full scale output (playback full scale) is lower than  $\pm 32767$ .

## 5.3.2 Frequency Response

The test sequence is:

- Unmute the path under test
- Set generator to 997 Hz,  $-20$  dB FS (relative to the input under test)
- (Make sure all filters are off on the analyzer - widest bandwidth possible)
- Set measured value from the analyzer to 0 dB
- Sweep frequency from 20 Hz to 20 kHz, recording amplitude values
- Find the lower and upper frequency corner where amplitude crosses the  $A_c$  point

### 5.3.2.1 Analog Mixer (A-A)

- Unmute the analog mixer path under test
- Run test as listed above

A mixer path example, using the CS4237B, Line In jack to Output jack would be to set registers I2 and I3 in the WSS space to 0x68, which would unmute the path to the output mixer and set the volume to 0 dB.

# Personal Computer Audio Quality Measurements

## 5.3.2.2 Full Duplex Digital Loopback (A-D-PC-D-A)

- Unmute the input path to the ADCs
- Unmute the playback path through the DACs
- Run a full-duplex loopback on the PC that takes data from the ADCs and sends it back to the DACs
- Run test as listed above

The mixer path example, using the CS4237B, Line In jack to Output jack would be to set:

for recording:

I2, I3 = 0x68 (unmute path to ADC and set volume to 0 dB)

for playback:

I6, I7 = 0x00 (unmute digital playback and set volume to 0 dB)

X14, X15 = 0x00 (unmute digital mixer and set volume to 0 dB)

then run full-duplex software.

## 5.3.3 Dynamic Range

The test sequence for the analog mixer (A-A) and full-duplex (A-D-PC-D-A) paths mentioned above is:

- Unmute the path under test
- Set the generator to 997 Hz, -60 dB FS (relative to the test input full-scale voltage)
- Set analyzer bandwidth to 20 Hz - 20 kHz
- If weighting filters are used, enable weighting filter, i.e. A-weighting
- Set the analyzer to measure THD+N relative to the full-scale output voltage  
Notch out the fundamental and measure the residual energy

The test sequence for the record path alone (A-D-PC) is:

- Unmute the path under test
- Set generator to 997 Hz, -60 dB FS
- Record data on EUT
- Use FFT software to calculate Dynamic Range on the recorded data

The test sequence for the playback path alone (PC-D-A) is:

- Unmute the path under test
- Play a dithered digitally-generated 997 Hz, -60 dB FS sine wave.
- Set analyzer bandwidth to 20 Hz - 20 kHz
- If weighting filters are used, enable weighting filter, i.e. A-weighting

# Personal Computer Audio Quality Measurements

- Set the analyzer to measure THD+N relative to the full-scale output voltage  
Notch out the fundamental and measure the residual energy

## 5.3.4 Total Harmonic Distortion and Noise

This test sequence is the same as the Dynamic Range sequence, except the test signal level is now -3 dB FS. In addition, weighting filters should be turned off (flat response) for THD+N measurements.

The test sequence for the analog mixer (A-A) and full-duplex (A-D-PC-D-A) paths is:

- Unmute the path under test
- Set the generator to 997 Hz, -3 dB FS (relative to the tested input full-scale voltage)
- Set analyzer bandwidth to 20 Hz - 20 kHz
- Set the analyzer to measure THD+N relative to the full-scale output voltage  
Notch out the fundamental and measure the residual energy

The test sequence for the record path alone (A-D-PC) is:

- Unmute the path under test
- Set generator to 997 Hz, -3 dB FS
- Record data on EUT
- Use FFT software on the PC to calculate THD+N

The test sequence for the playback path alone (PC-D-A) is:

- Unmute the path under test
- Play a dithered digitally-generated 997 Hz, -3 dB FS sine wave.
- Set analyzer bandwidth to 20 Hz - 20 kHz
- Set the analyzer to measure THD+N relative to the full-scale output voltage  
Notch out the fundamental and measure the residual energy

The tests on the record path (A-D-PC) can be combined into one sequence that calculates THD+N and Dynamic Range at the same time by isolating the distortion components in the Dynamic Range measurement.

## 5.3.5 Continuous Power

If an output is designed for headphone or speaker loading:

- Connect the proper load to the output
- Set the mixer to analog loopback (A-A)
- Set the generator for 997 Hz, 0 dB FS
- If the output THD+N is less than or equal to -40 dB
  - Measure the  $V_{RMS}$  output and set as full scale
- If the output THD+N is worse than -40 dB
  - Reduce the output full scale value until THD+N equal to -40 dB
  - Measure the  $V_{RMS}$  output and set as full scale
- Calculate the continuous power using the load and full scale voltage.

# Personal Computer Audio Quality Measurements

## 6 REFERENCE DOCUMENTS

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- [2] AES17-1991 (ANSI S4.51-1991), "AES standard method for digital audio engineering - Measurement of digital audio equipment," NY: Audio Engineering Society, Inc. (1991).
- [3] CP-307, "Methods of Measurement for CD Players," Standards of Electronic Industries Association of Japan (1985).
- [4] H. Tremaine, *Audio Cyclopedia*, IN: Howard W. Sams & Co., Inc., pp. 1663-1668 (1969).
- [5] G. Ballou, *Handbook for Sound Engineers*, Carmel, IN: SAMS, a Division of Macmillan Computer Publishing, pp. 1388-1390 and p. 1435 (1991).
- [6] "Acoustics - Normal Equal-Loudness Level Contours," ISO 226 (1987).
- [7] CCIR Recommendation 468-4, "Measurement of Audio Frequency Noise in Broadcasting, Sound Recording Systems and on Sound Programme Circuits," *Broadcasting Service (Sound)*, Dubrovnik: International Telecommunication Union, Volume X - Part 1, pp. 285-291 (1986).

## APPENDIX A. MEASUREMENT PATHS

Most PC sound systems have multiple measurement paths. This section names and defines the alternative paths. For chips that include a PC bus interface (i.e. ISA or PCI) the tests other than analog loopback (A-A) must cross the bus interface which will show any analog degradation due to bus interference. There are three different types of paths: single direction record and playback (Figure 9), analog external loopbacks (Figure 10), and digital PC loopbacks (Figure 11).

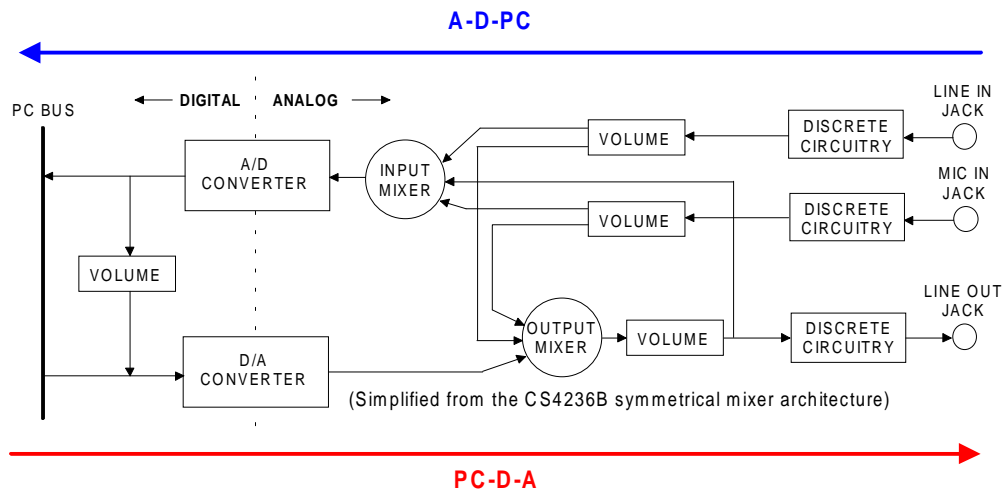


Figure 9. Record and Playback Paths

# Personal Computer Audio Quality Measurements

## A.1 Analog to Digital Path (A-D-PC)

Apply an analog signal to an analog input, set the mixer to route the signal to the A/D converters. Route the digital data into memory or disk in the PC. Analyze the digital data to get the record path performance numbers.

## A.2 Digital to Analog Path (PC-D-A)

Playback digitally-generated ideal data from memory or disk to the D/A converter. Route the analog out from the D/A converter to an analog output. Using external equipment, measure the performance of the playback path.

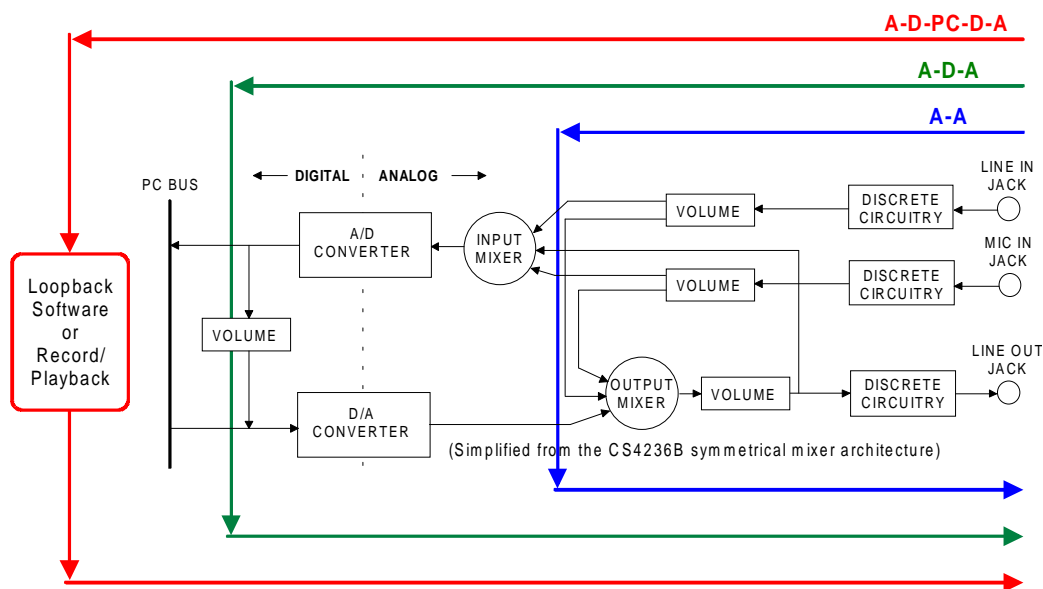


Figure 10. Analog External Loopback Paths

## A.3 Analog Signal Path (A-A)

Apply an analog signal to an analog input, set the mixer to route the analog signal to an analog output connector. Using external equipment, measure the performance of the analog mixer path. The signal does not go through the A/D converter or D/A converter whose outputs should be muted. This mode is often used for listening to music CD's using the CD-ROM drive analog output.

## A.4 Analog to Digital to Analog Signal Path (A-D-PC-D-A)

Apply an analog signal to an analog input, set the mixer to route the signal to the A/D converters. Route the digital data into memory or disk in the PC. Playback the data from the memory or disk to the D/A converters (may be simultaneous with recording or later). Route the analog out from the D/A converters to an analog output. Using external equipment, measure the performance of the full-duplex loopback path.

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This path yields a composite result of record and playback measurements added together, which will mask details of the separate playback and record values, but does give a good indicator of overall quality. For example, if the record measures 80 dB FS, and the playback measures 80 dB FS, then the composite measurement will be 77 dB FS. If the specifications for record and playback respectively are 80 dB FS, then a composite measurement of 80 dB FS guarantees that both record and playback must be operating at greater than or equal to 80 dB FS. However, any composite measurement between 77 dB FS and 80 dB FS indicates that record and playback MAY be inside specification. A composite measurement of less than 77 dB FS indicates that one (or both) of the record or playback measurements is less than 80 dB FS, and therefore is below the record/playback specification.

Most codecs provide an internal loopback path from the A/D digital output to the D/A digital input, bypassing the PC bus. (A-D-A). This is only useful for codec characterization and debugging, and is not useful for system measurements, since the data flowing over the PC bus can degrade analog performance for audio codecs with a parallel bus interface. Using A-D-PC-D-A is a more realistic and severe test in this case.

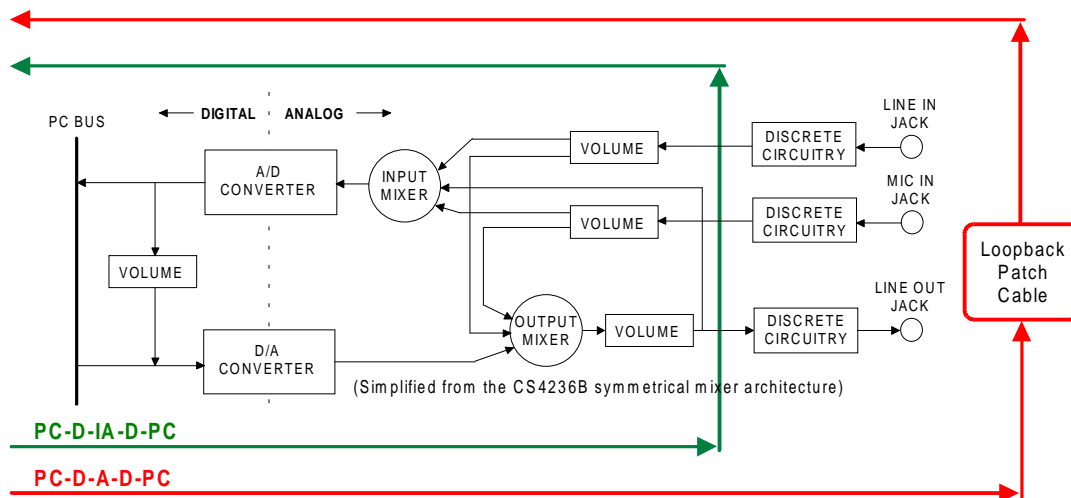


Figure 11. Digital PC Loopback Paths

## A.5 Digital to Analog to Digital Path (PC-D-A-D-PC)

Playback digitally-generated ideal audio data from memory or disk to the D/A converter. Route the analog out from the D/A converters to an analog output connector. Route the analog signal to an analog input via an external shielded cable, usually fitted with 3.5 mm jack plugs. Set the input mixer to route the analog signal to the A/D converter. Input the data from the A/D converter into the PC memory or disk. Analyze the digital data for full duplex path performance data.

This path is useful for an automated self test, requiring only the appropriate software and a patch cable. The system must be capable of simultaneous record and playback. This test assumes that the nominal full-scale analog output signal level is approximately equal to the nominal full-scale analog input signal level. If this is not true, then the test results do not provide accurate



# Personal Computer Audio Quality Measurements

performance numbers. For systems that have different output and input full-scale values, the full-scale output voltage should be adjusted to equal the full-scale input voltage by using a special patch cord that has gain or attenuation built-in.

As with the analog full-duplex path (A-D-PC-D-A), this path yields a composite result of record and playback added together, which will mask details of the separate playback and record values, but does give a good indicator of overall quality. For example, if a record measurement is 80 dB FS, and a playback measurement is 80 dB FS, then the composite will measure 77 dB FS. If the specifications for record and playback respectively are 80 dB FS, then a composite measurement of 80 dB FS guarantees that both record and playback must be operating at greater than or equal to 80 dB FS. However, any composite measurement between 77 dB FS and 80 dB FS means that record and playback MAY be inside specification. A composite measurement of less than 77 dB FS indicates that one (or both) of the record and playback measurements is less than 80 dB FS, and therefore is below the record/playback specification.

Most codecs provide an internal loopback path from the D/A analog output to the A/D analog input, bypassing the external components (PC-D-IA-D-PC). This is only useful for codec characterization and debugging, and is not useful for system measurements, since any external components can degrade analog performance.

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## A.6 Serial Codecs

Serial codecs are defined as codecs that interface on the digital side to a serial bus as opposed to codecs that reside on a parallel bus such as ISA or PCI. These serial codecs could be AC '97 codecs that connect to a digital controller, or codecs that reside on a USB or IEEE 1394 serial bus. The above paths are still valid testing points for serial codecs; however, there is one other path that is valid and is shown in Figure 12.

This path is listed as A-D-S-D-A and is valid as long as the power for the serial codec system is from the same source as a typical user would use (no lab power supplies). This path goes from analog to digital and then across the serial bus and then gets looped back to the codec's DACs and back out through the analog interface. Since the serial codec does not have a built-in parallel interface, bus interference is not expected.

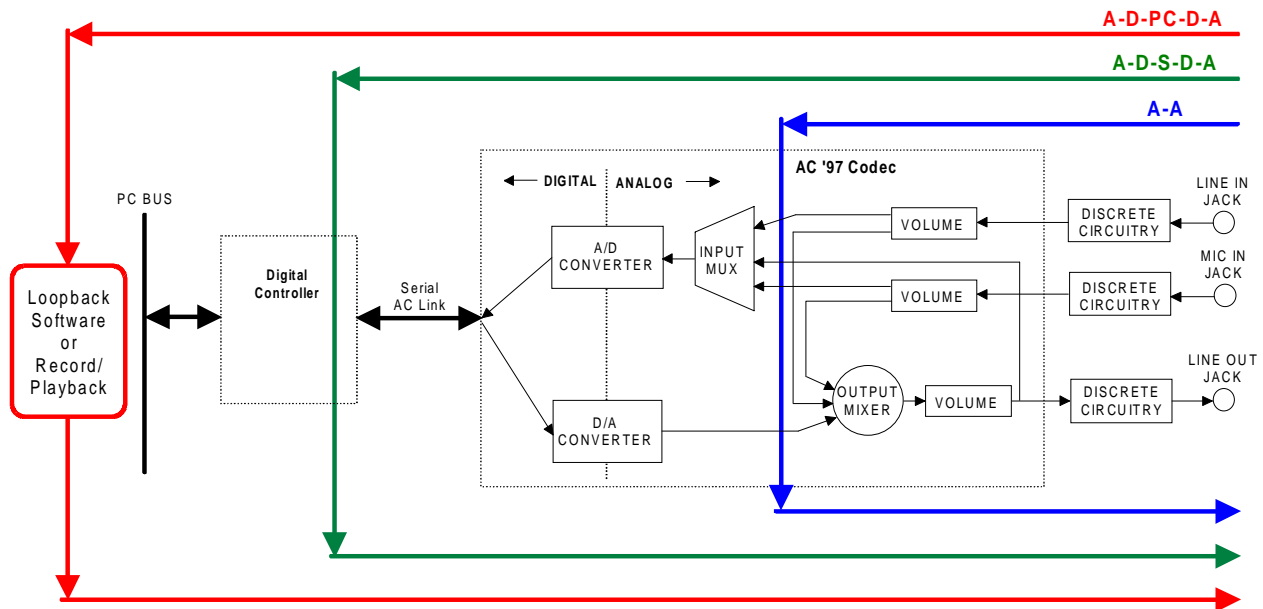


Figure 12. Serial Codec Analog Loopback Paths

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## APPENDIX B. DECIBELS

“Decibels” were initially used for equating power levels in audio and radio transmission [4]. The ‘bel’, named after Alexander Graham Bell, is the logarithmic ratio of two powers and is expressed as:

$$\text{bel} = \log \frac{P_1}{P_2}$$

Due to the large size of the bel, the decibel, or one tenth of a bel, has become the standard and is abbreviated ‘dB’. The formula being:

$$\text{dB} = 10 \log \frac{P_1}{P_2}$$

Although the decibel is a measurement of power ratios, it can also be used to measure voltage, current, and air pressure. Voltage and current are proportional to power as the square of their values. Likewise air pressure is proportional to the square of acoustic power. A voltage example would be:

$$\text{dB} = 10 \log \frac{(V_1)^2}{(V_2)^2} \quad \text{or} \quad 20 \log \frac{V_1}{V_2}$$

Decibels and percent (%) are similar quantities since they are both ratios. Percent is still used in some audio measurements such as THD. The equation to convert between them is:

$$? \text{ dB} = 20 \log \frac{? \%}{100}$$

Both are relative measurements and are meaningless without knowledge of the reference. In general, % and dB are relative to the level of the incoming signal; however, this definition is not universal. Volume control levels are always relative to the incoming signal. Using a volume control example:

<u>Percent</u>		<u>Decibels</u>
400 %	$20 \log (400/100)$	+12 dB
200 %	$20 \log (200/100)$	+6 dB
100 %	$20 \log (1)$	0 dB
50 %	$20 \log (50/100)$	-6 dB
25 %	$20 \log (25/100)$	-12 dB
10 %	$20 \log (10/100)$	-20 dB
1 %	$20 \log (1/100)$	-40 dB
0.1 %	$20 \log (0.001)$	-60 dB
0.01 %	$20 \log (0.0001)$	-80 dB

If the volume control is set to 0 dB, then the output from the volume control is 100 % of the input signal.

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Decibels are used in audio performance measurements for two reasons. First, the ear discerns sounds in a somewhat logarithmic fashion. Second, the total dynamic range of the ear is very large and logarithms are more convenient on large number ranges.

When specifying performance numbers such as DR or THD+N, the reference must be specified. For example, if a -3 dB FS input signal produces a THD+N of 0.01 % and a -60 dB FS input signal produces a THD+N of 5.6 %, without knowing the reference level for the two measurements, the 5.6 % appears worse than the 0.01 % even though it is actually better. Measurements using percent are “generally” referenced to the incoming signal level. Since the -60 dB incoming signal is much lower than the -3 dB signal, the percent THD+N relative to the incoming signal is larger. When using ‘%’, the reference is inferred and not clearly stated. If the measurement is referenced to 100 times full scale, the THD+N would be 0.0001 % which looks great. However, the user could never achieve that reference value since 100 times full scale is impractical.

The standard reference for audio measurements is full scale. Using the same example data as above, but with dB FS units, yields the following analysis. The -3 dB FS input signal produces a THD+N of -83 dB FS and the -60 dB FS input signal produces a THD+N of -85 dB FS. It is clearly apparent that the -60 dB THD+N is better than the -3 dB THD+N since they both reference the same point: full scale. Full scale is a valid reference since it is the largest signal that the system can handle.

There are many decibel suffixes that link dB values to various reference points. The following is a short and incomplete list.

## **B.1 dB FS**

dB FS is the preferred unit of measurement for audio and stands for “dB relative to full scale.” For analog inputs and outputs, full scale must first be measured. Then all measurements relating to that input/output are relative to the measured full-scale value. For digital measurements, full scale is limited by the number of bits specified. In the PC, all measurements should be 16-bits; therefore, full-scale is limited to  $\pm 32767$  since this is the largest value produced by a 16-bit number. Note that this reference point only holds true when the THD+N is better than -40 dB. If the THD+N is worse than -40 dB, then the signal level must be lowered to the point where -40 dB is achieved, and that point is defined as full scale. If  $\pm 32767$  is achieved with better than -40 dB THD+N, then  $\pm 32767$  is the reference point for dB FS measurements. Assuming this reference point is achieved, the formula is:

$$\text{dB FS} = 20 \log \left( \frac{\text{max} - \text{min}}{65536} \right)$$

The term “dB FS” refers to different values for different measurement points. Using the record path (A-D-PC) and Line In for example, the statement

$$\text{THD+N} (-3 \text{ dB FS}, 997 \text{ Hz}) = -83 \text{ dB FS}$$

# Personal Computer Audio Quality Measurements

indicates that the test signal is a 997 Hz tone at -3 dB relative to the Line-In full scale (FS), and the THD+N value is -83 dB relative to a digital full-scale (FS) value. Note that “FS” on either side of the equation refer to different values. Assuming the volume through the path is 0 dB, the magnitude of the digital codes should be  $\pm 23197$ , which is -3 dB FS, as shown in the equation below.

$$20 \log \left( \frac{23197 - -23197}{65536} \right) = 20 \log \left( \frac{46394}{65536} \right) = 20 \log (0.7079) = -3 \text{ dB FS}$$

## **B.2 dBr**

dBr stands for “dB relative to an absolute reference.” This unit is used by the Audio Precision test equipment to indicate the measurement is relative to a specified value. The specified value can be entered on a System One or Two. This value is normally set to the measured full-scale value making dBr equivalent to dB FS.

## **B.3 dB V**

dB V stands for “dB relative to 1 Volt.” Since the divisor is 1, the formula simplifies as follows:

$$? \text{ dBv} = 20 \log \frac{? \text{ V}}{1 \text{ V}} \quad \text{or} \quad 20 \log (? \text{ V})$$

## **B.4 dBm**

This is a power rating used in the professional and broadcast industries and stands for “dB relative to 1 mW.” When using this value for audio measurements, the resistance must be known since audio performance measurements are based on voltage, not power. Old professional audio vacuum-tube systems had 600  $\Omega$  output and input impedance’s providing a true transfer of power. However, today’s systems have low output impedance and high input impedance transferring very little power between components. Most equipment listing measurements in dBm today, are actually measuring dBu (see section B.5). Professional audio uses 600  $\Omega$ ; however, sometimes 300  $\Omega$  or 150  $\Omega$  is used. For audio measurements and 600  $\Omega$ , the voltage would be:

$$P = \frac{V^2}{R} ; \text{ therefore, } V = \sqrt{1 \text{ mW} \times 600 \Omega} = 0.7746 \text{ V}$$

## **B.5 dBu**

dBu eliminates the confusion with dBm by setting the voltage reference to 0.7746 V. Therefore, dBu is equivalent to dBm using 600  $\Omega$ , but since dBu is referenced to a fixed voltage, the resistance is irrelevant.

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## B.6 dB SPL

dB SPL is a decibel acoustic sound pressure level (SPL) measurement, used in acoustic transducer measurements (i.e. speakers and microphones). The reference for SPL is  $20 \mu\text{Pa}$  [5] which is the typical minimum threshold for hearing. Therefore, the formula is:

$$\text{dB SPL} = 20 \log \frac{P}{20 \mu\text{Pa}}$$

## APPENDIX C. WEIGHTING FILTERS

Human hearing is not equally sensitive at all audio frequencies [6]. The ear is most sensitive to frequencies in the 1 kHz to 5 kHz range. Weighting filters try and give readings that roughly correspond to the way humans respond to sound. Sound pressure level measurements use weighting filters based on three curves: A, B, and C. The curve selected was based on the loudness of the sound. The C-weighting curve is generally flat over most of the audio band (FR = +0, -3 dB from 30 Hz to 8 kHz) and used for loud signals. The B-weighting curve is used for intermediate level sounds and has the same upper corner as the C-weighting, but the lower amplitude corner is 120 Hz. The A-weighting curve is used for measuring low level signals such as background noise. This curve, shown in Figure 13, has become a standard for acoustic noise measurements of all signal levels for the Occupational Safety and Health Administration (OSHA).

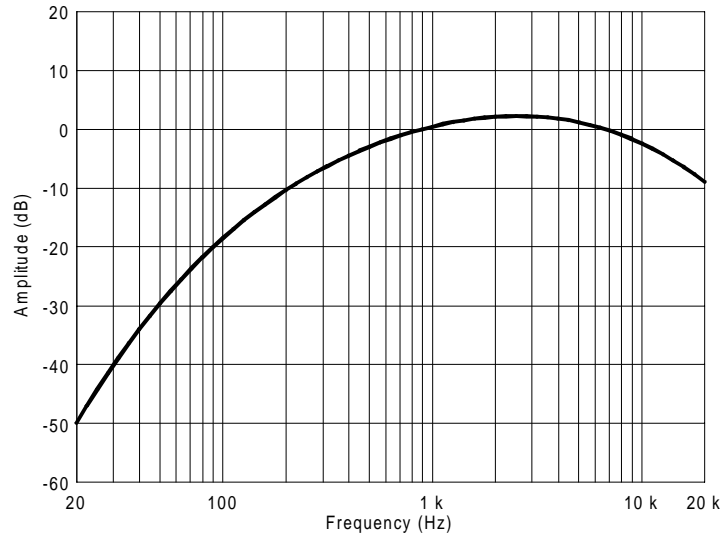
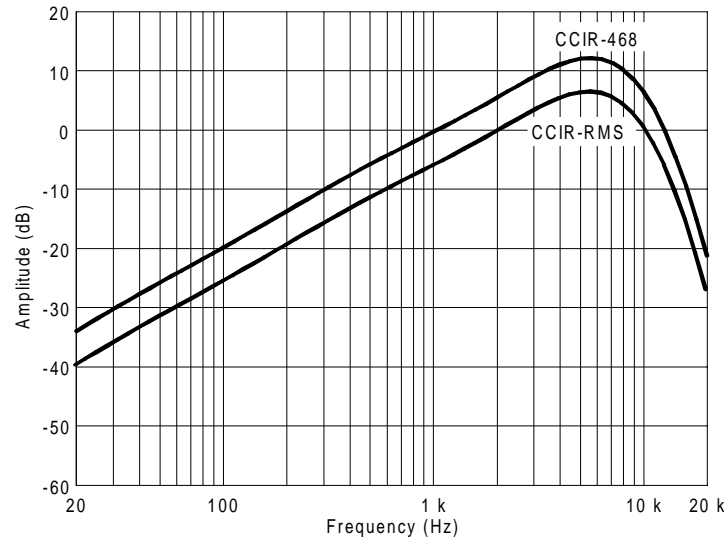


Figure 13. A-Weighting Curve

Other filters are used for audio performance measurements. The sound broadcast industry uses a CCIR-468 weighting filter [7] for noise measurements with a quasi-peak method of measurement. This filter is illustrated in Figure 14. The Audio Engineering Society [2] and Dolby use an attenuated version of this filter (CCIR-RMS) where the 0 dB point is 2 kHz instead of 1 kHz, with noise measured on an RMS basis.

# Personal Computer Audio Quality Measurements



**Figure 14. CCIR Weighting Curves**

When making an audio performance measurement, the result must 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) might be listed as

Line Level Analog Mixer DR = 90 dB FS A.

The “A” suffix indicates that an A-weighting filter was used to make the measurement. If no weighting filter is used, the suffix after the “dB FS” would be omitted. If the Dynamic Range measurement was measured using the AES filter, the measurement would be listed as

Line Level Analog Mixer DR = 90 dB FS CCIR-RMS.