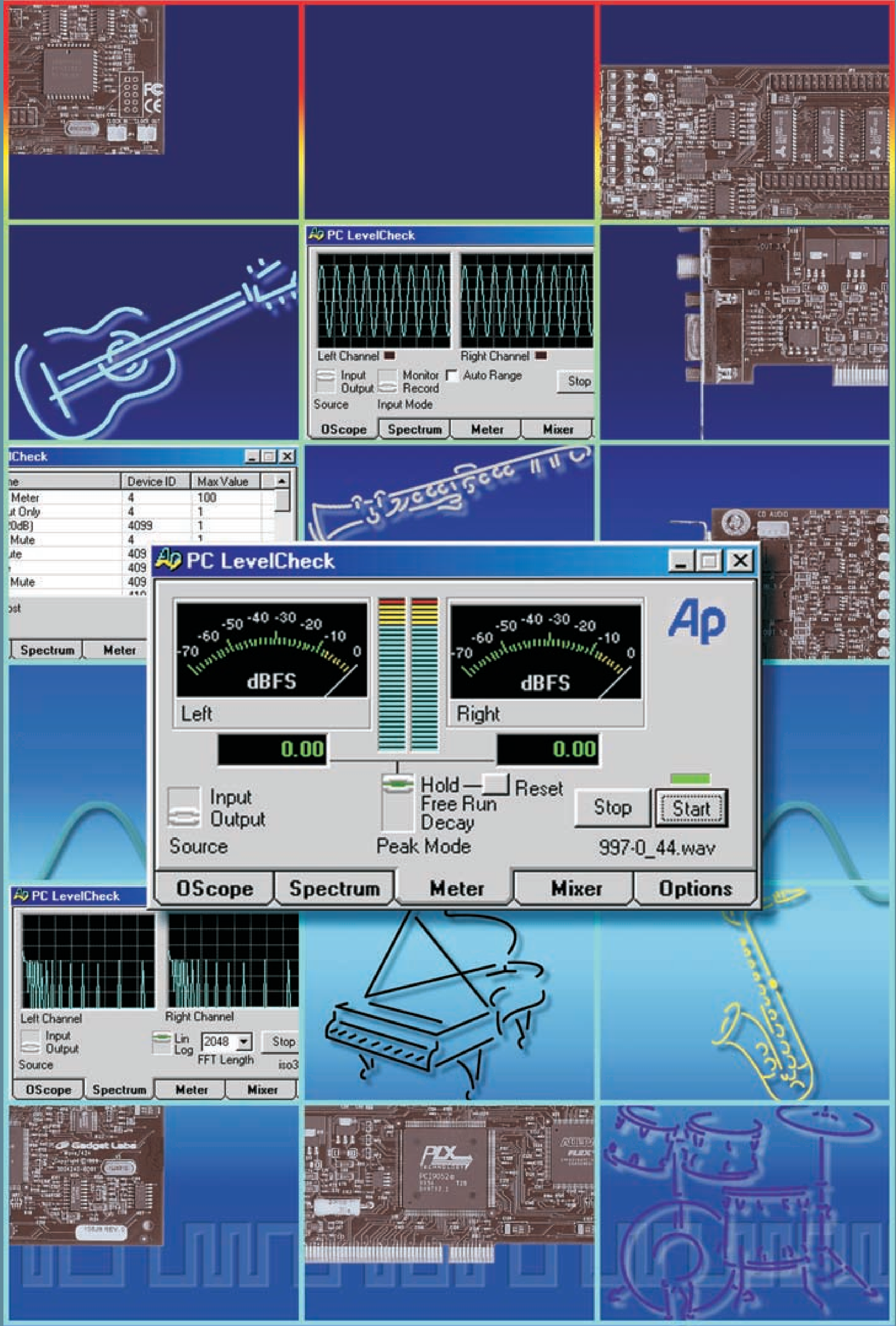
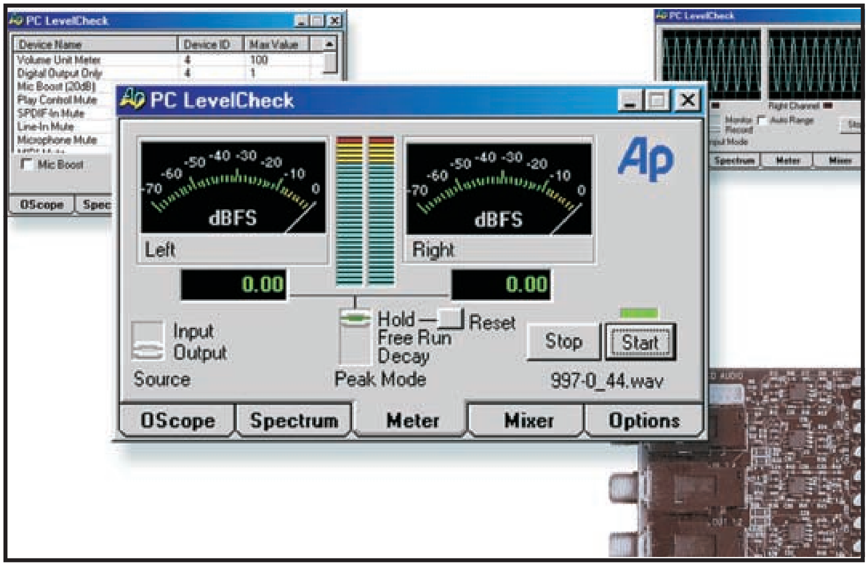


PC LevelCheck

Audio Precision PC .WAV File Level Meter

Audio precision®

PC LevelCheck



For PC LevelCheck Version 2.1
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Contents

Introduction

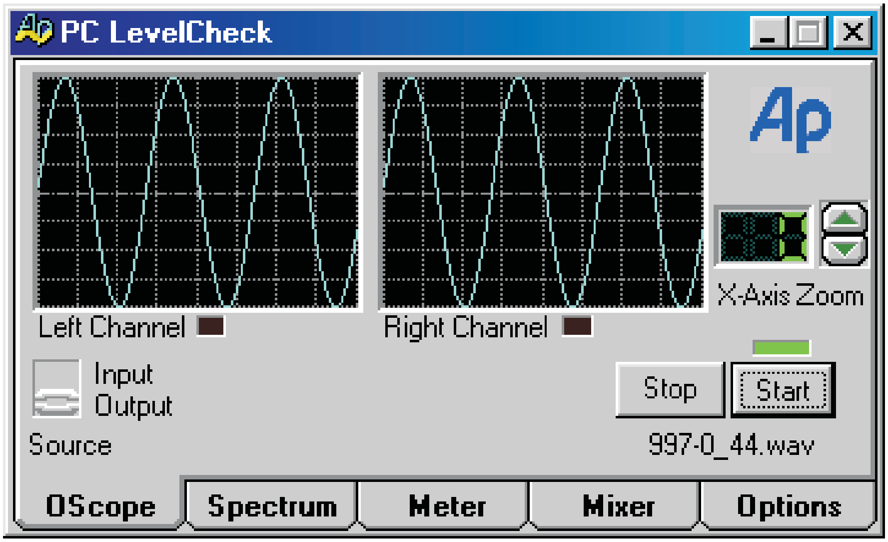


Figure 1 PC LevelCheck

PC LevelCheck (or *LevelCheck*) is a software PCM .wav file level meter and monitor. LevelCheck is an OLE automation server and can be controlled by APWIN Basic or run as a stand-alone application. See the **Glossary** (page 27) for definitions of PCM and OLE.

As a stand-alone program, LevelCheck monitors the digital audio signal in your computer's sound card, displaying the level, waveform and spectra of the audio channels in the card. Additionally, PC LevelCheck has the ability to control the record and playback mixers, and to record and play .wav files. When used in conjunction with APWIN and PC Audio Device Performance Tests, it provides features useful for automated PC sound card audio analysis. For more information about APWIN, APWIN Basic and PC Audio Device Performance Tests, see **About Audio Precision** (page 25).

PC LevelCheck

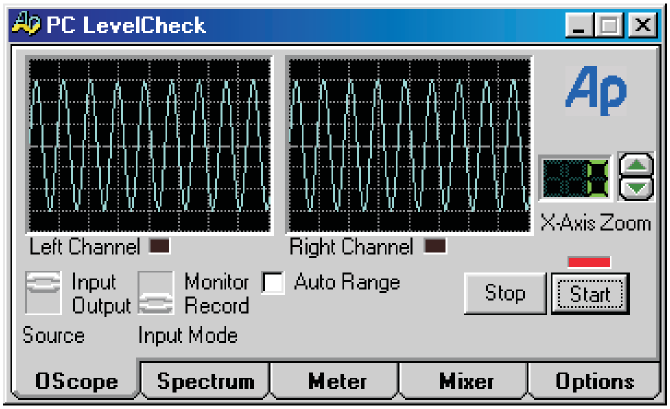
Common Controls

Source, Input Mode, Start and Stop are common controls in PC LevelCheck which appear on several of the display panels and affect the overall operation of the program.

Source: Input / Output

The **Source** switch box appears on the three signal display panels and offers two choices: **Input** and **Output**. Drag the switch up or down to change sources. With **Input** selected as the source, PC LevelCheck monitors the input signal to the sound card after it has been digitized in the card's analog-to-digital converter (ADC) and allows the option of recording the signal as a .wav file; **Output** selects the .wav file playback function of the sound card. When **Output** is selected, the name of the .wav file chosen for playback is displayed beneath the **Stop** and **Start** buttons.

Figure 2
*PC LevelCheck showing **Source**, **Input Mode** and **Auto Range** switches and **Stop** and **Start** buttons.*



Input Mode

With **Input** selected, an additional switch labeled **Input Mode** is visible on the three signal display panels. **Input Mode** allows you to select whether LevelCheck merely monitors the incoming signal or to direct the program to actually record a .wav file. Drag the switch up or down to select **Monitor** or **Record** modes.

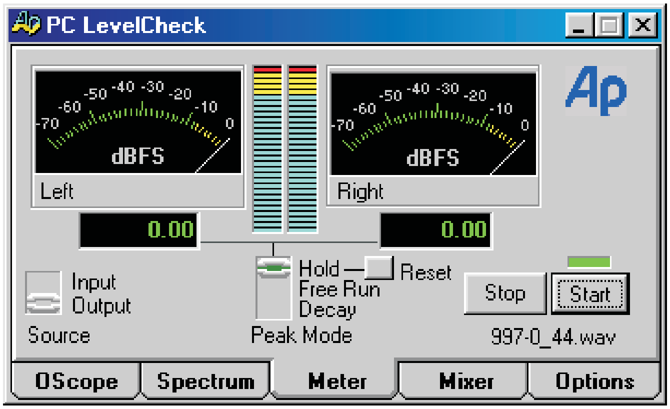
When **Input Mode** switch is set to **Record**, a Save Wave File dialog box appears so you can choose the name and the folder for the file you are about to record.

Stop and Start

The OScope, Spectrum and Meter panels each have a **Stop** and **Start** button.

On any of these three panels, when **Source** is set to **Input** and **Input Mode** is set to **Monitor,** click the **Start** button to acquire, process and display the audio embedded in the digital signal, moments after conversion from the analog inputs. During input signal monitoring the indicator bar above the **Start** button will be green. Click the **Stop** button to end acquisition.

Figure 3
PC LevelCheck Meter tab, showing Start and Stop buttons



With the **Source** switch is set to **Input** and the **Input Mode** switch is set to **Record**, the **Start** button now begins recording a .wav file. During recording the indicator bar above the **Start** button will be lighted red. **Stop** ends the recording.

When the **Source** switch is set to **Output**, clicking **Start** begins playback of the selected .wav file. (Playback files are selected on the **Options** panel.) During playback the indicator bar above the **Start** button will be green. Click **Stop** to end the file playback.

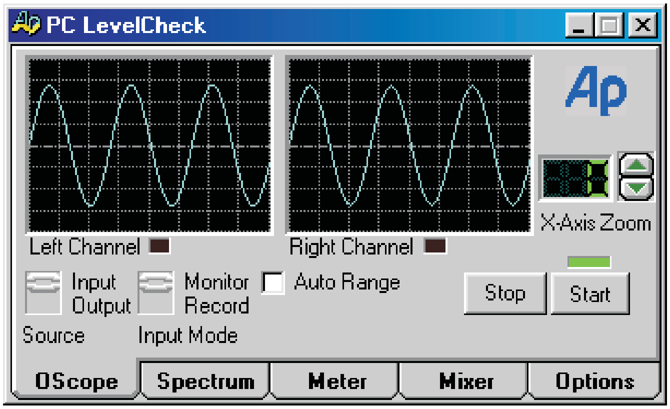
*Both **Source** and **Input Mode** switches are unavailable while PC LevelCheck is actually acquiring or playing a signal. If these switches appear dimmed and will not drag, click the **Stop** button and retry.*

Control Panels

PC LevelCheck has five tabbed control panels. The first three, labeled OScope, Spectrum, and Meter are primarily signal displays. Mixer offers mixer volume controls and mutes, and Options provides .wav file record and playback controls and other options.

OScope

Figure 4 PC LevelCheck OScope tab



The first panel simulates a dual-trace oscilloscope view of the signal, with a default horizontal axis of about 3 milliseconds (showing about 3 cycles of a 1 kHz waveform) and a vertical axis of 0 dB FS (dB FS means *decibels full scale*, one way of defining the maximum digital signal in a system. See the **Glossary**, page 27.)

X-Axis Zoom allows you to scroll through a number of uncalibrated horizontal axis settings to adjust the simulated oscilloscope view.

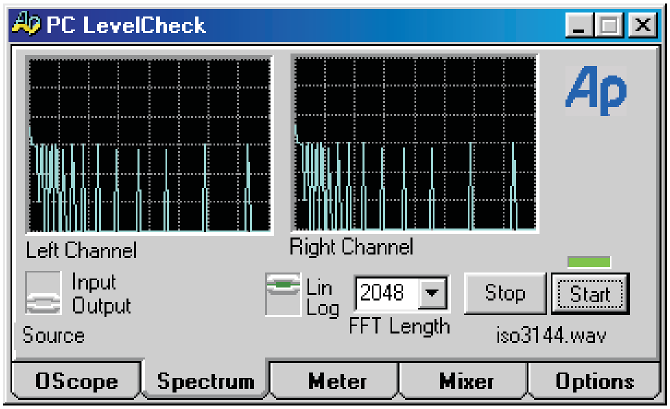
Click **Auto Range** to view small signals at full vertical scale. With **Auto Range** enabled, the actual range of the vertical axis is displayed below the check box. Even though the **Auto Range** check box is only shown on the OScope panel, its setting affects the readings displayed on the Spectrum panel as well.

Below each oscilloscope screen is a small peak level indicator, which lights red when the signal in that channel exceeds 0 dB FS.

*In **Auto Range** with no input signal (or a very small input signal) applied to the sound card, any offset of the OScope trace indicates a DC offset at the input to the ADC.*

Spectrum

Figure 5
PC LevelCheck
Spectrum tab



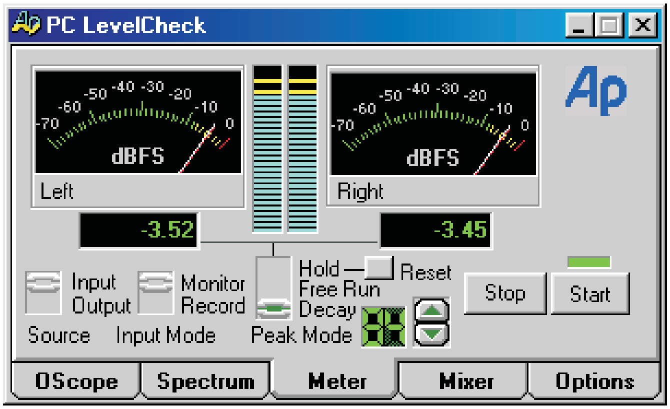
The second panel is a display of a dual channel spectrum analyzer. Drag the **Lin** / **Log** switch up or down to alternate horizontal axis display modes. In **Lin** mode the horizontal axis is set to a linear frequency scale with a range of 0 Hz to ½ the sampling frequency. In **Log** mode the horizontal axis is set to a logarithmic frequency scale with the same range, DC to Nyquist.

The graticule line below the top of the display represents the maximum level, 0 dB FS. When **Auto Range** is selected on the OScope panel, the vertical axis on the spectrum displayed is scaled by the same factor.

The spectrum analysis is performed digitally using fast Fourier transform (FFT) techniques. The length of the acquired FFT sample is selectable in the **FFT Length** list. Longer FFT lengths such as 1024 or 2048 give greater resolution, while shorter sample lengths offer faster measurement.

Meter

Figure 6
PC LevelCheck
Meter tab



The third panel represents a stereo peak level meter with both needle and bar graph displays. Full scale is 0 dB FS.

Display windows below the meters show the peak amplitude reached by the digital signal, referenced to 0 dB FS.

The Meter panel adds a **Peak Mode** switch to control the numerical readout in the display windows.

When **Peak Mode** is set to **Hold**, the windows display and hold the last maximum peak level acquired until the **Reset** button is clicked.

Free Run disables the **Hold** function, and the numerical displays will continually update the peak level values as they are acquired.

Decay is a compromise between these two. The numerical display windows are updated periodically, and you can select a wide range of periods between updates by scrolling the settings which appear in the adjacent window. Lower numbers result in more frequent updates; higher numbers set longer periods between updates.

Click **Reset** to clear the peak memory for these windows.

Mixer

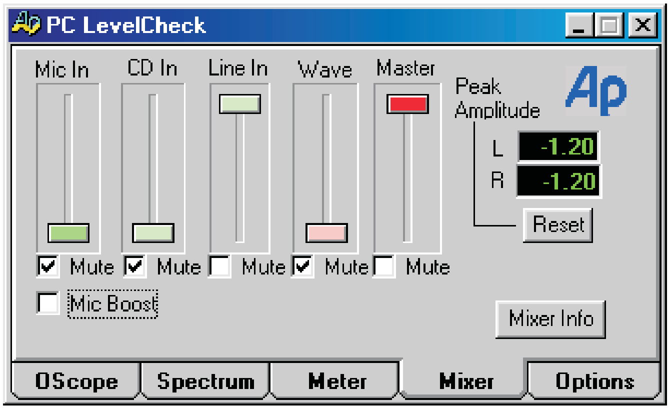


Figure 7
PC LevelCheck
Mixer tab

The Mixer panel gives easy access to the Windows Mixer Control application programming interface (API) providing a **Master** control and volume control sliders for **Mic In**, **CD In**, **Line In**, and **Wave**. Each channel can be muted by clicking the **Mute** check box below the slider. The controls on this mixer should parallel the operation of the native sound card mixer.

Click **Mic Boost** to add 30 dB of gain to the **Mic In** channel for microphone level inputs.

The information in the Meter panel **Peak Amplitude** boxes is repeated here, as is the **Reset** button.

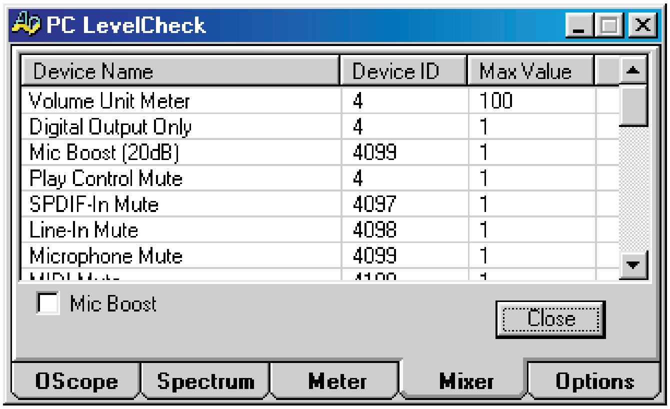
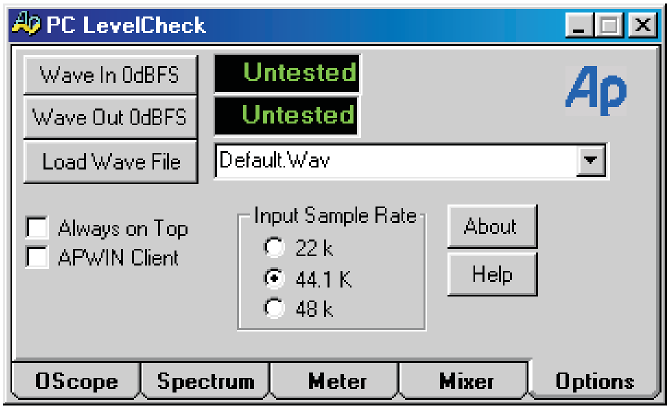


Figure 8
PC LevelCheck
Mixer Info window

All of the parameters of the mixer panel are accessible via OLE. Click **Mixer Info** to view a mixer property sheet, which shows the sound card driver's available control names, the associated Windows control I.D., and the maximum value for each control.

Options

Figure 9
PC LevelCheck
Options tab



The Options panel offers a number of miscellaneous functions.

The top two buttons in the upper left corner cause PC LevelCheck to calibrate the sound card's input and output analog levels to a 0 dB FS reference in recording and playing back .wav files. To do this, however, you must be running APWIN on the computer and be connected to System Two.

These calibration buttons are designed for use when PC LevelCheck is ***not*** *being used with the PC Audio Tests procedure. See* ***PC Audio Tests*** *(page 26).*

First select the **APWIN Client** box near the bottom of the panel. This establishes an OLE Automation relationship between the programs, with APWIN as the OLE server and PC LevelCheck as the client. Be sure that System Two is connected properly to your sound card, and that the System Two inputs and outputs are correctly configured.

If Audio Precision APWIN is not installed on your system, the ***APWIN Client*** *selection will not be available in PC LevelCheck.*

When all this is ready, click **Wave In 0 dB FS** to perform the input calibration. PC LevelCheck will measure the input level required to produce 0 dB FS in the sound card's ADC, and will enter that level both in the window next to the button and in the **References: dBr** box in the APWIN Analog Generator panel. See **Finding the 0 dB FS Input Level** (page 11).

Next, click **Wave Out 0 dB FS** to perform the output calibration. PC LevelCheck will measure the output levels produced by the sound card when a 0 dB FS .wav file is played, and will enter that level both in the window next to the button and in the **References: dBr A** and **dBr B** boxes in the APWIN Analog Analyzer panel. See **Finding the 0 dB FS Output Level** (page 13).

Wave In and **Wave Out** do not set stereo references within PC LevelCheck but use the left channel only.

When LevelCheck is launched the file Default.wav is set as the playback file. Click **Load Wave File** to choose an alternate .wav file for playback. The **Open Wave File** dialog box will appear.

Click **Always on Top** to keep the PC LevelCheck window in the foreground on your computer screen.

In the **Input Sample Rate** box, click **22 k**, **44.1 k** (the default) or **48 k** to set the sampling rate in the sound card ADC.

Using PC LevelCheck

As a stand-alone application, PC LevelCheck can monitor and measure the digital input and output levels within your sound card. As an OLE client it can interface with APWIN or other programs to bring new capabilities.

For example, through OLE and APWIN Basic, PC LevelCheck is an integral part of Audio Precision's PC Audio Device Performance Tests, a powerful group of procedures and tests which, with APWIN and System Two can provide automated testing of all important sound card parameters. For more information on Audio Precision, APWIN, System Two or PC Audio Tests, see **About Audio Precision** (page 25) or visit the Audio Precision Web site at www.audioprecision.com.

Confirming Signal Input

PC LevelCheck can easily verify your signal input connections and mixer settings. Turn on the System Two generator and set the outputs to a sine wave at about –20 dBV. (If you don't have System Two, use another tone generator or any audio source, set to a moderately low line level.) Then choose the **OScope** tab, select **Input** and click the **Start** button. You should see signal in both the left and right channel displays. If there are no signals in the OScope displays, go to the Mixer tab and be sure that the **Master** slider and the **Aux In** slider are both up and unmuted. Check your connections as well.

Finding the 0 dB FS Input Level

Sound cards vary greatly in their input signal characteristics. With PC LevelCheck you can easily determine the analog signal voltage which corresponds to the maximum digital signal level, 0 dB FS.

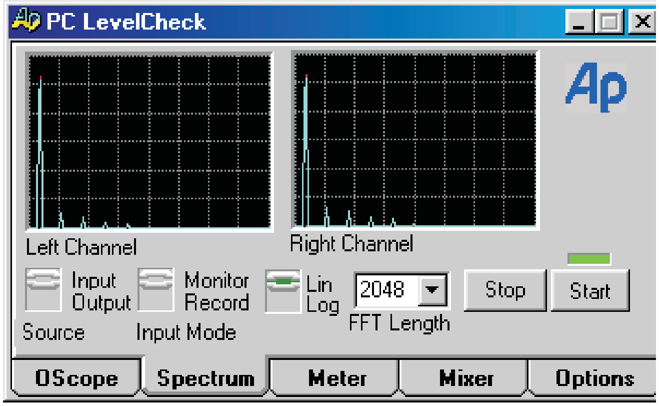
To determine the input voltage to produce a 0 dB FS recording, first you must be sure that the record volume control is properly set. The driver software included with some sound cards maps the record volume control to the **Aux In** slider. In this case, set the record volume by clicking the **Mixer** tab on LevelCheck and verifying that the **Aux In** slider is unmuted. Next drag the **Aux In** slider to the bottom of its range, and then move it up one step by pushing the up arrow key on your keyboard one time.

If your sound card is connected to System Two, click the **Options** tab and then the **APWIN Client** check box. When APWIN is connected, click the **Wave In 0dBFS** button. Using APWIN and System Two, PC LevelCheck will determine the input level which produces 0 dB FS, and will display that level in the top box on the Options panel, and will also enter the left and right channel values as dBr references in APWIN. When the APWIN generator units are set to dBr all amplitude settings are referenced to the sound card's full scale input.

If you are not using SystemTwo, connect an audio generator to the sound card's Line Inputs, with the frequency set near 1 kHz (the ideal mid-range frequency to test 16-bit PCM audio is 997 Hz) and the output voltage set to about 100 mV (–20 dBV).

Click **Spectrum.** Be sure **Source** is set to **Input** and **Input Mode** is set to **Monitor**. Click **Start**. You will see a spike in the spectrum corresponding to the tone you are feeding into the sound card.

Figure 10 Fundamental Tone and Harmonic Spikes



Now increase the level of the tone generator until you see harmonic spikes in the spectrum display. Reduce the level until it is just below the point where the harmonic spikes appear and note the signal level applied to the sound card inputs under these conditions. This is your input reference voltage.

The rms level which produces a 0 dB FS signal in the .wav file is the maximum input signal level of the sound card as a system. Good recording practice suggests choosing an operating level 10 to 20 dB below this level to provide adequate headroom.

Finding the 0 dB FS Output Level

Sound cards also vary greatly in their output signal characteristics. With PC LevelCheck you can easily determine the analog signal voltage which corresponds to the maximum digital signal level, 0 dB FS.

To determine the output voltage for 0 dB FS, first make sure the **Wave** and **Master** volume sliders on the Mixer panel are all the way up and unmuted. Then click the **Options** tab and load the file WaveOutCal.wav.

If your sound card is connected to System Two and you would like to set the analyzer level references, also click the **APWIN Client** check box. Then go to one of the display panels (OScope, Spectrum or Meter), set **Source** to **Output** and click the **Start** button. If PC LevelCheck is connected as an APWIN client, return to the Options tab and click **Wave Out 0dBFS** to measure the signal voltages and set the APWIN dBr references. When the APWIN analyzer units are set to dBr all amplitude measurements are referenced to the sound card's full scale output.

If you are not using APWIN or System Two, simply measure the rms AC voltage level at the outputs of the sound card.

The rms level which corresponds to 0 dB FS is the maximum output signal level of the sound card as a system. You should expect normal signal levels to be referenced to an operating level 10 to 20 dB lower than this level.

OLE Methods and Properties

PC LevelCheck ActiveX components consist of two OLE objects, named LevelCheck.Monitor and LevelCheck.Mixer. The Monitor object contains methods and properties for controlling the level checking functions. The Mixer object contains the methods and properties for controlling the Windows Mixer API.

Table 1 below lists methods and properties for these objects.

Table 1 PC LevelCheck OLE Methods and Properties		
Monitor Properties		
AutoRange	Boolean	Sets Auto Range function in Source: Input
CurrentTab	Integer	Sets display mode: 1= OScope; 2 = Spectrum; 3 = Meters; 4 = Mixer; 5 = Options
LeftPeakValue	Double	Read only: value of left peak in dB FS
LeftPos	Integer	Sets application's left position on windows desktop
LoadError	Boolean	Read only; returns True if error occurred when loading
OnTop	Boolean	Sets Always On Top windows style (Z order)
Record	Boolean	Sets Input Mode to Record for recording wave files
RightPeakValue	Double	Read only: value of right peak in dB FS
TopPos	Integer	Sets application's top position on windows desktop
Visible	Boolean	Toggles visibility
WaveInFileName	String	File name for saving record wave file
WaveInSampleRate	Integer	Sample rate value
WaveOutFileName	String	File name for loading playback wave file
WaveSelect	Integer	Sets Source: 0 = Input; 1 = Output
Monitor Methods		
ResetPeakMeter	Void	Resets both left and right peak meters
WaveStart	Void	Starts playing, monitoring or recording wave data
WaveStop	Void	Stops playing, monitoring or recording wave data
Monitor Events		
Aborting	Void	Event fires when user closes PC LevelCheck application
PCMOverload	Integer	Event fires when PCM overload occurs: 0 = channel one; 1 = channel 2
Mixer Properties		
AuxMute	Boolean	Sets Aux In (Line In) Mute
CDMute	Boolean	Sets CD In Mute
MasterMute	Boolean	Sets Master Mute
MicBoost	Boolean	Sets Mic Boost switch
MicMute	Boolean	Sets Mic In Mute
WaveMute	Boolean	Sets Wave Mute
Mixer Methods		
GetVolume (ctrl,MaxValue)	Variant,Variant	Gets mixer volume ctrl [0 = Mic; 1 = CD; 2 = AUX; 3 = Wave; 4 = Master] MaxValue = 0–100
SetVolume (ctrl,MaxValue)	Variant,Variant	Sets mixer volume ctrl [0 = Mic; 1 = CD; 2 = AUX; 3 = Wave; 4 = Master] MaxValue = 0–100

Appendix A: Installation

PC LevelCheck is distributed from the Audio Precision Web site and also on CD-ROM accompanying other Audio Precision products.

To download the installation files from the Web, go to http://www.audioprecision.com, click the SOFTWARE button and scroll the list for the latest version of PC LevelCheck. Follow the instructions to download and expand the files.

If you are using Microsoft Internet Explorer, you will have the option of running InstallShield from the Web site or saving the installation file to disk. If you run the program, InstallShield will install PC LevelCheck on your computer over the net.

If you save the file to disk (the only option with Netscape Navigator), the installation file will be saved in the folder you designate. To launch InstallShield, go to that folder and double-click on Setup.exe

If you are installing from the CD-ROM, place the disk in your computer's drive. If your CD-ROM drive is set to automatically run, InstallShield will immediately launch. If not, go to My Computer and click on the icon for the CD-ROM drive. When the drive window opens, double-click on Setup.exe to launch InstallShield.

Simply follow the prompts in InstallShield. You can accept all the default choices, or you can designate your own folders and file locations.

When installation is complete, you are given the options of viewing the Readme file or immediately launching PC LevelCheck.

Appendix B: PC Sound Cards

A Typical PC Sound Card

A typical inexpensive PC sound card has, among other features, the capability to record and play .wav file format 16-bit PCM stereo digital audio at a 44.1 kHz sample rate. These cards also offers mono file capability, lower sampling rates and 8-bit word length as options. Analog connections are invariably unbalanced circuits with maximum levels of +6 dBV (2 V rms) or often much less, provided on 3.5 mm mini phone jacks.

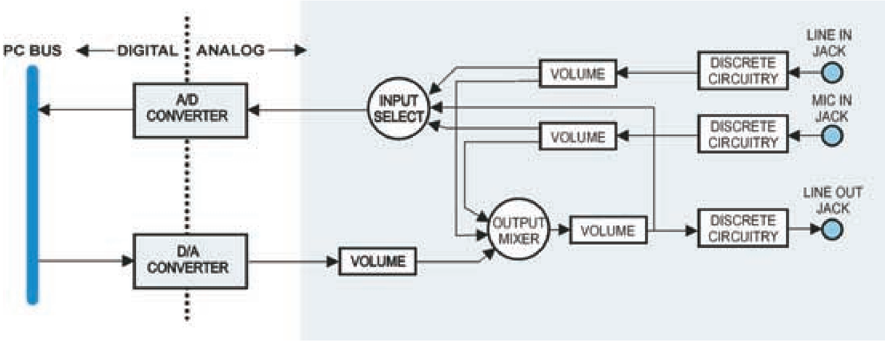


Figure 11 Typical Sound Card Block Diagram

Designed for the mass market, such a PC sound card is often referred to as a "consumer" sound card, as opposed to the "professional" or "semi-pro" sound cards designed for specialized markets.

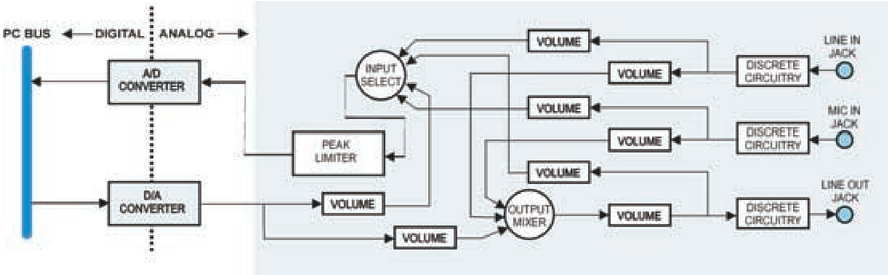


Figure 12 Typical Sound Card Block Diagram

Recording path

The recording signal path for a sound card includes a stereo input stage with relatively high input impedances in the 10 kW to 100 kW range, with a maximum input level sensitivity which can vary from as low as –25 dBV to perhaps +6 dBV. The sound card driver software provides a recording mixer to select and adjust analog inputs to the ADC. Volume control changes from the mixer are accomplished in the analog domain in discrete and often rather coarse steps. Gain, commonly 30 dB, can be added to the Mic In channel by selecting the "Mic Boost" option. The recording mixer typically will allow only one input channel to be selected at a time. Cards so configured cannot mix several input signals for recording.

In some sound cards the analog signal is applied to the input of the ADC linearly, allowing extreme signal excursions the possibility of exceeding the maximum input level of the ADC. In other cases, the signal is routed through a peak limiter to prevent digital clipping. While protecting the unwary user from severe distortion, this can complicate measurement by making it difficult to determine the precise input level necessary to produce 0 dB FS.

When the presence of a peak limiter makes direct measurement of the onset of digital clipping impossible, PC LevelCheck uses another method to closely estimate the input level required for 0 dB FS, as follows:

A low-level (-25 dBV) 997 Hz sine wave is applied to the input of the sound card and PC LevelCheck reads the digital code created by the ADC in response. This digital data corresponds to an embedded audio level which can be expressed in units of dB FS. With this baseline, the generator level required to produce 0 dB FS can be estimated and applied to the input of the sound card.

Nonlinearities in the sound card can throw off this estimate, so PC LevelCheck next increments the estimated input level up and down in an attempt to locate the threshold of 1% distortion at the output of the ADC. The largest applied input signal which produces just less than 1% distortion is then defined as 0 dBr.

In the first case, measuring a sound card with no limiter, 0 dBr = 0 dB FS. In measuring a card with a limiter, 0 dBr = 1 % THD+N, which will be close to but not exactly 0 dB FS, depending upon the behavior of the limiter.

Playback path

The playback signal path for a sound card includes a stereo output stage with relatively low output impedances in the 100 W range. Outputs which are designed to drive headphones have even lower impedances. Analog output levels for a 0 dB FS signal are typically between –1 dBV and +6 dBV; for outputs which are also designed to power speakers, the circuits will have higher current capabilities and levels as high as +12 dBV. The sound card driver software provides a playback mixer to adjust and mute analog outputs from the DAC. Volume control changes from the mixer are accomplished in the analog domain in discrete, coarse steps.

Sound Card Driver Software

With the exception of occasional configuration switches or jumpers, the functions of PC sound cards are entirely controlled by driver software.

Although setting levels, selecting sources and recording and playing .wav files through a PC sound card can be done in many different applications, for test and measurement purposes it is often simplest to use the software provided with Microsoft Windows: Windows Volume Control, Windows Sound Recorder and Windows Media Player.

Sound card configuration options can be set from a number of software locations via a number of paths, including:

- Start Settings Control Panel Multimedia Audio
- Start Settings Control Panel System Device Manager Sound, Video and Game Controllers your sound card here
- Start Programs Accessories Entertainment Sound recorder Edit Audio Properties
- Start Programs Accessories Entertainment Sound recorder Save As Change…
- Start Programs Accessories Entertainment Media Player View Options
- Start Programs Accessories Entertainment Volume Control Options Properties
- Start Programs Accessories Entertainment Volume Control Options
- Advanced Controls Advanced…

Audio Connections

The consumer market sound card usually has three audio jacks on the back panel: Line In, Mic In and Line Out; some also have Aux In, Speaker Out or Surround jacks, and MIDI and game port connections. There are also usually internal connections on Molex headers to the PC Compact Disc drive, the PC speaker circuit and sometimes the PC telephone audio device (TAD).

PC LevelCheck uses only the Line Out connector and the Line In connectors. On a typical sound card, these are invariably 3.5 mm stereo jacks, and should be connected to System Two as shown:

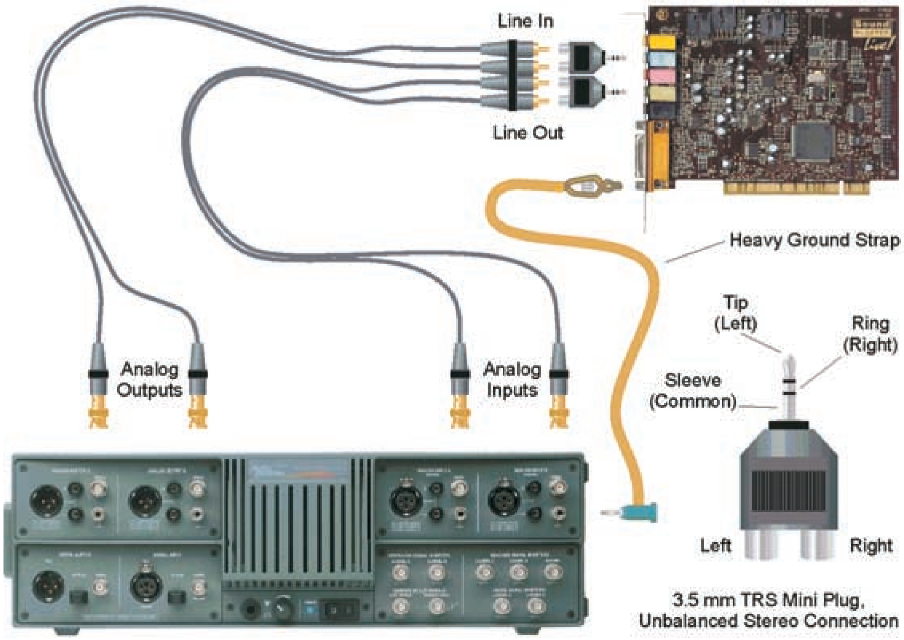


Figure 13 Connecting a typical PC sound card to System Two for testing

"Professional" PC Sound Cards

A smaller percentage of sound cards are designed for specialized markets: specifically, the professional audio and broadcast industries, and semi-professional "project studio" environments. Although these cards offer more features, superior performance and better interfaces, in general the testing methods using PC LevelCheck are identical, with the interface connections being the primary difference.

As always, there is great variance in features and implementation, but pro sound cards may offer:

- Full duplex operation (simultaneous file record and playback capability)
- Sampling rate of 48 kHz
- Greater resolution and signal to noise ratio, with digital word length of 20 or 24 bits.
- Pro-type analog interfaces, with high-level, balanced connections

- Superior analog circuitry, ADCs and DACs.

All of the above features are fully compatible with PC LevelCheck. Other added features need a little more discussion:

- Multitrack capability (more than the standard left and right stereo tracks).
- Higher sampling rates, such as 88.2 kHz or 96 kHz.
- Direct digital interfaces, such as S/PDIF, ADAT optical or AES3

Multitrack capability

Pro audio cards may be able to record and play on more than just the two standard (left and right) stereo channels, either by using two or more physical cards (linked for synchronization) or by installing multiple sound devices on a single physical sound card.

Multitrack sound cards come with proprietary software which allows simultaneous use of more than two channels. PC LevelCheck, Windows Sound Recorder, Windows Media Player and, for that matter, System Two are all two-channel devices. But Microsoft Windows does support use of more than one sound card, and using a multitrack card in Windows you will find channels 3–4, 5–6, etc. as additional audio devices at:

Start Settings Control Panel System Device Manager Sound, Video and Game Controllers your sound card here

To use the higher-numbered channels, select the device driver for the additional pair. Then move the interconnection cables between System Two and the sound card to the correct jacks. To measure all the channels on a multitrack sound card with PC LevelCheck, you must move through the channels two at a time by reconfiguring your software drivers and repatching your analog connections.

Connections for Professional Sound Cards

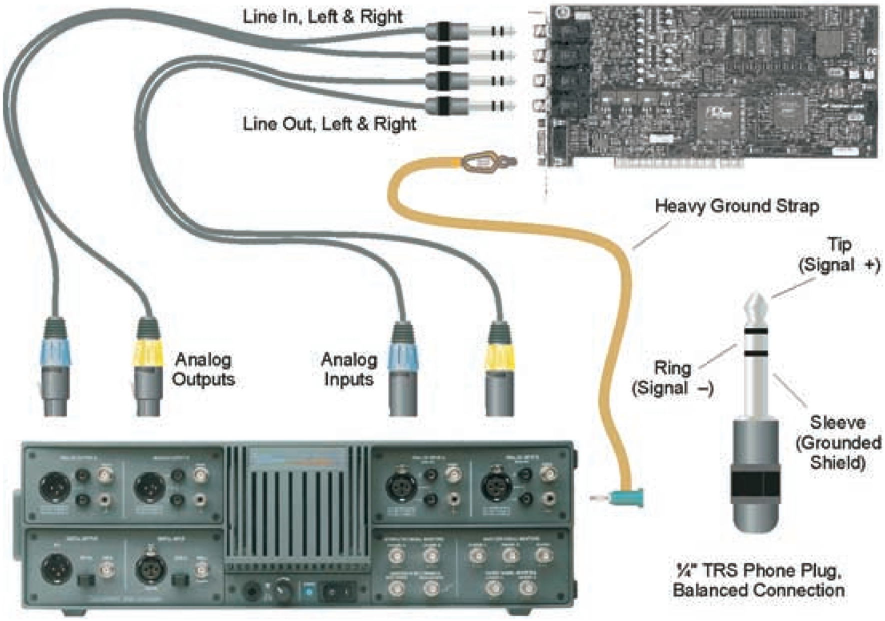


Figure 14 Connecting a typical "pro" PC sound card to System Two for testing

Appendix C: About Audio Precision

Figure 15
Audio Precision World Headquarters in Beaverton, Oregon, USA

Audio Precision is the world's largest company dedicated purely to audio test and measurement. We develop and manufacture a wide range of innovative audio test instruments of comprehensive capabilities which offer performance at industry-leading levels.

More than 10,000 AP instruments are in use around the world, in the hands of development engineers, manufacturing and quality control engineers and technicians, broadcast and recording engineers, university professors and students, consultants, and magazine reviewers.

Our products are used to evaluate mixing consoles, digital and analog recorders, power amplifiers, transmitters, microphones and loudspeakers, computer sound cards, set-top converters, hearing aids, aircraft entertainment and intercommunications systems, professional video cameras and recorders, routing switchers, A-to-D and D-to-A converters, cell telephones and other telecom equipment, and almost any other device that transmits, stores, processes, or reproduces audio signals.

Call us at 1-800-231-7350 or visit our Web site at http://www.audioprecision.com

System Two

Figure 16
Audio Precision System Two and APWIN

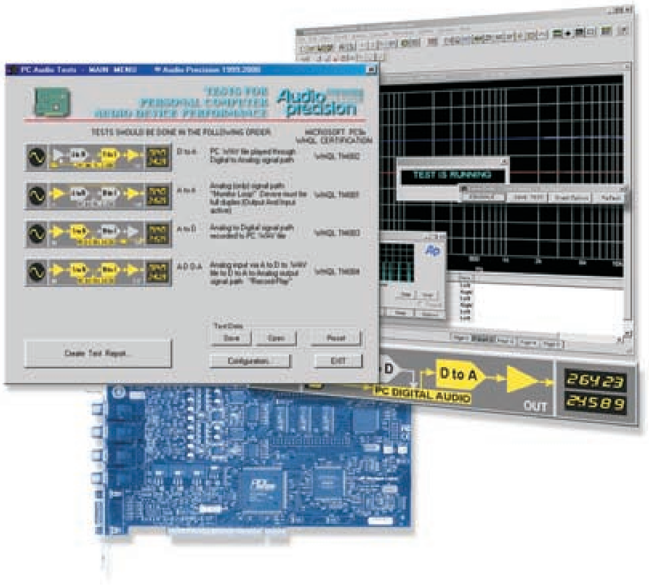
System Two is a PC-controlled audio measurement system (using APWIN software) that includes both stimulus and measurement capability. It can measure a variety of audio parameters including gain, frequency response, distortion, noise, frequency, phase, crosstalk, and more. System Two also has sophisticated digital audio measurement capabilities. The Dual Domain model can characterize digital audio jitter, measure rise & fall time, and display *eye* patterns.

Audio Precision also manufactures instruments with lesser and greater capabilities (System One and System Two Cascade), portable instruments and ancillary equipment.

PC Audio Tests

Audio Precision's *PC Audio Device Performance Tests* (or *PC Audio Tests*) is a complete set of tests and procedures for measuring the performance of the digital recording and playback functions of a PC sound card or other computer sound device. The sound card must be connected to an Audio Precision System Two and installed in the computer running APWIN.

Figure 17
PC Audio Tests



Audio Precision's System Two and APWIN software provide carefully generated test stimuli and powerful analysis, with direct read and write access to the host computer's data files. To these existing capabilities PC Audio Tests brings new tests, utilities and .wav files especially created for sound card evaluation, all run from within a user-friendly APWIN Basic procedure. The companion program PC LevelCheck further extends APWIN's reach to include direct real-time access to the digital audio signal in the computer's memory.

PC Audio Tests measures sound card levels, distortion and noise, dynamic range and frequency response in both the analog and digital domains. The tests in PC Audio Tests are designed to produce specific measurements required either to satisfy a standard, such as Microsoft WHQL pc99, AES6, AES17, or a custom specification.

Glossary

ADC—see **analog-to-digital converter**.

AES3 interface—a digital interface standard for professional audio equipment interconnection, defined in the AES3 standard. Formerly known as the *AES/EBU interface*. The digital signal carried in the AES3 interface is partially compatible with the digital signal carried on the S/PDIF interface.

AES—the Audio Engineering Society, with headquarters in New York City.

analog audio, analog signal—a representation of an audio signal as a continuously variable quantity. An analog audio signal is usually an electrical voltage varying in analogy to the sound waves it represents.

analog-to-digital converter—a device for converting an analog input signal into a series of digital values representing the instantaneous amplitude of the signal at regular sampling intervals. Abbreviated ADC. See **digital recording or processing**.

analog-to-digital—abbreviated A-to-D, A/D, A-D and so on. See **digital recording or processing** and **analog-to-digital converter.**

API—Application Programming Interface, the software interface used between application programs and the Windows operating system.

balanced—a term referring to an audio transmission line in which the signal is applied differentially between two conductors, each of which has equal impedances to a common reference or ground.
A balanced line is usually constructed with three conductors: an internal twisted pair of wires, one carrying signal "+" or HIGH, the other carrying signal "–" or LOW. These are surrounded by a third conductor in the form of a braided or foil shield, which is connected to the common or ground terminal at one or both ends of the cable.
Used with properly engineered equipment, balanced lines are superior

in performance to unbalanced connections, yielding better rejection of common-mode interference caused by electrostatic and electromagnetic fields. Additionally, since the audio in a balanced circuit is isolated from the ground conductor, ground-current-induced noise is much more easily dealt with. Also called a *symmetrical line*. See **ground**, **unbalanced**.

bandpass filter—a filter which passes a specific frequency band (called the *passband*) essentially without attenuation while attenuating frequencies both below and above the specified band.

bit depth—see **word length**.

bits of resolution—the number of bits of the binary word by which signals are represented in a digital recording or transmission system. Each bit adds approximately 6 dB to the theoretical dynamic range available. Thus, a 16-bit digital system is capable of approximately 96 dB dynamic range, etc.

bus—in electricity and electronics, a conductor common to three or more circuits. In computers, the data bus is a set of conductors which carry common data between three or more subunits of the computer.

clipping—the action of a system in flattening and squaring off signal peaks when driven with a signal whose peak amplitude is beyond its linear signal-handling capability.

crest factor—the ratio of a signal's peak amplitude to its rms amplitude.

DAC—see **digital-to-analog converter**.

dB—abbreviation for decibel, a ratio unit for expressing signal amplitudes. If the amplitudes are expressed in voltage, dB = 20 log10 (V1/V2). If the amplitudes are expressed in power, dB = 10 log10 (P1/P2). Two important points to remember:
1. The dB, mimicking our hearing, is not a linear unit but a measurement on a logarithmic scale.
2. decibels are relative units of measure, having no meaning in an absolute sense. A dB must always be referenced to something to have meaning. You can speak of a "4 dB change" or a "60 dB signal-to-noise ratio" because these are both relative statements, but

describe a signal level as 39 dB means nothing. All absolute measurements expressed as decibels must have an indication of reference, such as dBu, dBV, dBr and so on. See **dBu**, **dBV**, **dBr**.

dB FS—decibels referenced to digital Full Scale (FS), where 0 dB FS is the rms value of a sine wave whose positive peak just reaches positive full scale.
For everyday use, 0 dB FS can be considered to be the maximum digital amplitude available within a system. However, it should be noted that because 0 dB FS has been defined for a sine wave, waveforms with lower crest factors can exceed 0 dB FS by as much as 3.01 dB before incurring digital clipping. See **crest factor** and **digital clipping**.

dBm—decibels relative to a reference value of 1 milliwatt. dBm is a power unit and requires knowledge of power levels (voltage and current, or voltage and impedance, or current and impedance) rather than merely voltage. This term has historically been associated with a measurement of audio levels in "professional" use, but it is rarely correctly applied and should not be used except in certain very specific circumstances. As a unit of power, the voltage value of a dBm will vary with the circuit impedance.
Use the term *dBu* instead. See **dBu**.

dBr—decibels relative to an arbitrary reference value, **r**. The reference value must be stated for this to be a meaningful unit. The dBr can be a handy shortcut during tests, allowing you, for example, to measure a specific voltage and note it as your reference, 0 dBr; and then to express further measurements in dBr relative to this level.

dBu—decibels relative to a signal level of 0.7746 V rms. dBu now is the common term for analog audio amplitude in "professional" audio interfaces and circuits. 0 dBu (0.7746 V rms) equals 0 dBm, but *only* in a 600 ohm load impedance. Unless you are clearly interested in measuring power in a circuit of known impedance, use dBu, not dBm. See **dBm**.

dBV— decibels relative to a signal level of 1 volt rms. dBV now is the common term for analog audio levels in "consumer" audio interfaces and circuits. 0 dBV equals +2.218 dBu.

digital overflow—*see* **digital clipping**.

digital recording or processing—a technique in which the original signal is periodically sampled and the amplitude value at each sampling instant is converted into a number represented by a binary word.

digital-to-analog— Abbreviated D-to-A, D/A, D-A and so on. See **digital recording or processing** and **digital-to-analog converter.**

digital-to-analog converter—a device which converts a stream of digital numbers, each representing the amplitude of a signal at a particular sampling time, into a corresponding analog signal. Abbreviated DAC.

DSP—digital signal processing.

dynamic range—the difference, usually expressed in dB, between the highest and lowest amplitude portions of a signal, or between the highest amplitude signal which a device can linearly handle and the noise level of the device.

EUT—a common abbreviation in the test and measurement field for "equipment under test."

FFT—fast Fourier transform, a technique to compute the amplitude versus frequency and phase versus frequency information from a set of amplitude versus time samples of a signal.

ground loop—an inadvertent signal path formed when interconnecting the chassis of two or more pieces of equipment, each possessing a safety ground. Ground loops can cause hum-related interference.

ISO—International Organization for Standards, the largest of the many international groups for technical and industrial cooperation. The ISO is based in Geneva, Switzerland.

line level—a relatively high amplitude range suitable for transmission of audio signals. Line level is typically in the 0 dBu to +8 dBu range.

OLE—Object Linking and Embedding, the Microsoft technology which empowers the exchange of data and control between different software applications; the forerunner of ActiveX automation.

one-third octave—a bandwidth of 1/3 octave, or a frequency ratio of 1.2599:1. Three successive frequency changes by this ratio result in a total frequency change of 2:1 (one octave). Moderately narrow bandpass filters are often set at a bandwidth of 1/3 octave; frequency response measurement techniques which use spot frequencies (such as real-time spectrum analysis or multitone tests) often use frequencies centered at 1/3 octave distances.

PCM—see **pulse code modulation**.

pulse code modulation—a form of data transmission in which amplitude samples of an analog signal are represented by digital numbers. Abbreviated PCM. Almost all digital audio schemes use PCM.

resolution—the smallest change in a measured parameter to which a measurement instrument can respond.

rms—see **root mean square**.

root mean square—the preferred form of ac signal detection which measures amplitude in terms of its equivalent power content, regardless of signal waveshape. Abbreviated rms.

sample frequency, sample rate—the frequency at which the signal is sampled in a digital system. The sample rate must exceed twice the highest analog frequency to be converted. Commonly used sample rates are 48 kHz, 44.1 kHz, and 32 kHz.

signal-to-noise ratio—the difference in level between a reference output signal (typically at the normal or maximum operating level of the device) and the device output with no signal applied. Signal-to-noise ratio is normally stated in dB. The device input conditions for the noise measurement must be specified, such as “input short circuited” or with a specific value of resistance connected at the device input instead of a signal.

S/PDIF—Sony / Philips Digital Interface; a digital interface for consumer audio equipment. Sometimes also referred to as the EIAJ interface. The S/PDIF is similar to the professional AES3 interface, but is normally an unbalanced coaxial signal of lower amplitude. Most of the status byte definitions are different between S/PDIF and AES3.

THD+N—total harmonic distortion plus noise. Measured by attenuating the fundamental signal with a narrow-band notch filter, then measuring the remaining signal which consists of harmonics of various order, wide-band noise, and possibly interfering signals. This is the common harmonic distortion method implemented in most analyzers.

third octave—*see* **one-third octave**.

unbalanced—an audio connection in which the desired signal is present as a voltage with respect to ground or common, rather than as a differential signal across a pair of balanced conductors.

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